Administrator’s Guide for Avaya Communication Manager

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About this book

Overview

Avaya Communication Manager is the centerpiece of Avaya applications. Running on a variety of Avaya Media Servers and DEFINITY Servers, and providing control to Avaya Media Gateways and Avaya communications devices, Communication Manager can be designed to operate in either a distributed or networked call processing environment.

Communication Manager carries forward all of a customer's current DEFINITY capabilities, plus offers all the enhancements that enable them to take advantage of new distributed technologies, increased scalability, and redundancy. Communication Manager evolved from DEFINITY software and delivers no-compromise enterprise IP solutions.

Communication Manager is an open, scalable, highly reliable and secure telephony application. The software provides user and system management functionality, intelligent call routing, application integration and extensibility, and enterprise communications networking.

Purpose of this book

This book describes the procedures and screens used in administering the most recent release of Communication Manager running on any of the following:

- Avaya media servers
  - DEFINITY servers
  - S8100, S8300, S8500, or S8700 Media Servers
- Avaya media servers configured as a Local Survivable Processor (LSP).
- Avaya media gateways
  - MCC1 or SCC1 Media Gateways
  - G350, G600, G650, or G700 Media Gateways

Newer releases of Communication Manager contain all the features of prior releases.

Intended audience

This document is intended for system administrators and managers, for users interested in information about specific features, and Avaya personnel responsible for planning, designing, configuring, selling, and supporting the system.

Contents

This document includes the following chapters:

- Chapter 1: System Basics tells you how to log in and log off, set permissions for others who use the administration terminal, set daylight savings rules, set the system date and time, post messages, and back up the information you administer.
- Chapter 2: Planning the System provides information on system-wide functions. It explains how to read, use, and make simple changes to your dial plan, and how to assign feature access codes.
- Chapter 3: Managing Phones explains how to add, swap, and remove phones, and how to customize a telephone, for Communication Manager administrators.
- Chapter 4: Managing Telephone Features explains how to administer feature buttons for your users' phones.
- Chapter 5: Managing Attendant Consoles explains attendant console feature buttons, and tells you how to change, move, or add attendant consoles.
- Chapter 6: Managing Displays provides information on the messages that appear on the read-out screen on display phones.
- Chapter 7: Handling Incoming Calls shows you how to set up call coverage for incoming calls to be sure that incoming calls are answered when the called party is unavailable.
- Chapter 8: Routing Outgoing Calls explains how Avaya Communication Manager handles outgoing calls and tells you how to modify call restrictions and your routing plan.
- Chapter 9: Managing MultiMedia Calling describes the Multimedia Applications Server Interface (MASI), and provides instructions on administration, monitoring, and troubleshooting. This section also provides information on Multimedia Call Handling (MMCH), which enables users to control voice, video, and data transmissions using a telephone and PC.
- Chapter 10: Setting Up Telecommuting provides information on system-wide settings and individual user administration for telecommuting.
- Chapter 11: Enhancing System Security provides information on analyzing and setting up basic system security, preventing toll fraud, using logins and permissions and passwords, and dealing with security violations.
- Chapter 12: Managing Trunks contains procedures for working with analog and digital trunks. Specialized trunks such as APLT, tandem, release-link, DMI-BOS and ISDN trunk groups are not covered in this manual.
- Chapter 13: Managing Announcements tells you how to record, save, copy, restore and delete announcements.

- Chapter 14: Managing Group Communications shows you how to administer your system so users can page other users or use their phones as intercoms. You can also give specific users permission to monitor other users' calls or to interrupt active calls with important messages.
- Chapter 15: Managing Data Calls describes the system features available to enable data communications.
- Chapter 16: Administering Media Servers describes how to administer an Avaya S8700 or S8710 Media Server and an Avaya G700 Media Gateway and the Avaya S8300 Media Server.
- Chapter 17: Collecting Billing Information provides information on account codes, and on tracking and collecting information about calls.
- Chapter 18: Command Reference Use the commands in these tables to access each administration screen.
- Chapter 19: Telephone Reference describes many of the telephones that you can connect to Communication Manager. It also describes the unique features and buttons for each telephone series to help you administer your user phones.
- Chapter 20: Screen Reference provides a brief description and a graphic representation of the Communication Manager screens used for administration. It also lists the valid values for fields on the screens, and describes when and why to use each value.

Conventions used

Become familiar with the following terms and conventions. They help you use this book with Communication Manager.

- A "screen" is the display of fields and prompts that appear on a terminal monitor.

  See the Screen Reference chapter for an example of a screen and how it is shown in this book.
- We use the term "telephone" in this book. Other Avaya books might refer to telephones as voice terminals, stations, or endpoints.
- Keys and buttons are printed in a bold font: **Key**.
- Titles of screens are printed in a bold font: **Screen Name**.
- Names of fields are printed in a bold font: **Field Name**.
- Text (other than commands) that you need to type into a field are printed in a bold font: **text**.
- Commands are printed in a bold constant width font: **`command`**.
- Variables are printed in a bold constant width italic font: ***`variable`***.

- We show complete commands in this book, but you can always use an abbreviated version of the command. For example, instead of typing **`list configuration station`**, you can type **`list config sta`**.
- If you need help constructing a command or completing a field, remember to use **Help**.
  - When you press **Help** at any point on the command line, the system displays a list of available commands.
  - When you press **Help** with your cursor in a field on a screen, the system displays a list of valid entries for that field.
- Messages that the system displays are printed in a constant width font: **system message**.
- To move to a certain field on a screen, you can use the **Tab** key, directional arrows, or the **Enter** key on your keyboard.
- If you use terminal emulation software, you need to determine what keys correspond to **Enter**, **Return**, **Cancel**, **Help**, and **Next Page** keys.
- We show commands and screens from the newest release of Communication Manager and see the most current books. Substitute the appropriate commands for your system and see the manuals you have available.
- The status line or message line can be found near the bottom of your monitor. This is where the system displays messages for you. Check the message line to see how the system responds to your input. Write down the message if you need to call the helpline.
- When a procedure requires you to press **Enter** to save your changes, the screen clears. The cursor returns to the command prompt. The message line shows "**command successfully completed**" to indicate that the system accepted your changes.

Admonishments

Admonishments that might appear in this book have the following meanings:

Note:

Draws attention to information that you must heed.

Tip:

Draws attention to information that you might find helpful.

CAUTION:

Denotes possible harm to software, possible loss of data, or possible service interruptions.

WARNING:

Denotes possible harm to hardware or equipment.

DANGER:
Denotes possible harm or injury to your body.

SECURITY ALERT:
Denotes when system administration might leave your system open to toll fraud.

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Related Sources

The following documents provide additional information.

Administration for Network Connectivity for Avaya Communication Manager, 555-233-504

ATM Installation, Upgrades, and Administration using Avaya Communication Manager, 555-233-124

Avaya Application Solutions -- IP Telephony Deployment Guide, 555-245-600

Avaya Communication Manager Advanced Administration Quick Reference, 03-300364

Avaya Communication Manager Basic Administration Quick Reference, 03-300363

Avaya Communication Manager Basic Diagnostics Quick Reference, 03-300365

Avaya Licensing Guide, 03-300157

Avaya Toll Fraud and Security Handbook, 555-025-600

Converged Communications Server Installation and Administration, 555-245-705

DEFINITY Communications Systems Generic 2.2 and Generic 3 Version 2 DS1/CEPT1/ISDN PRI Reference, 555-025-107

DEFINITY Enterprise Communications Server Release 1.1 Getting Started with the Avaya R300 Remote Office Communicator, 555-233-769

Feature Description and Implementation for Avaya Communication Manager, 555-245-205

Hardware Guide for Avaya Communication Manager, 555-245-207

Highlights of Avaya Communication Manager, 555-245-704

Installation, Upgrades and Additions for Avaya CMC1 Media Gateways, 555-233-118

Maintenance Alarms for Avaya Communication Manager 2.1, Media Gateways and Servers, 03-300190

Maintenance for Avaya DEFINITY Server R, 555-233-117

Maintenance for Avaya MultiVantage and DEFINITY Server CSI, 555-233-119

Maintenance for Avaya DEFINITY Server SI, 555-233-123

Maintenance Commands for Avaya Communication Manager 2.1, Media Gateways and Servers, 03-300191

Maintenance Procedures for Avaya Communication Manager 2.1, Media Gateways and Servers, 03-300192

Overview for Avaya Communication Manager, 555-233-767

System Capacities Table for Avaya Communication Manager on Avaya Media Servers, 555-245-601

Upgrades and Additions for Avaya DEFINITY Server R, 555-233-115

Reports for Avaya Communication Manager, 555-233-505

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Mention this document's name and number, *Administrator's Guide for Avaya Communication Manager,* 555-233-506.

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How to get help

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If you need additional help, the following resources are available. You may need to purchase an extended service agreement to use some of these resources. See your Avaya representative for more information.

Go to the Avaya Web site at http://www.avaya.com/support:

- If you are within the United States, click the **Escalation Management** link. Then click the appropriate link for the type of support you need.
- If you are outside the United States, click the **Escalation Management** link. Then click **International Services**, which includes telephone numbers for the international Centers of Excellence. Or contact your local Avaya authorized dealer for any additional help and questions.

Chapter 1: System Basics

Logging into the System

You must log in before you can administer your system. If you are performing remote administration, you must establish a remote administration link and possibly assign the remote administration extension to a hunt group before you log in. The members of this hunt group are the extensions of the data modules available to connect to the system administration terminal. For information about setting up remote administration, contact your Avaya technical support representative.

Note:

Avaya recommends that you change your password frequently to prevent unauthorized access to your system. For instructions on how to change your password, see Changing Passwords on page 409.

When not using the system, log off for security purposes.

Instructions

Logging into the system

This procedure provides instructions for logging in from the system terminal not a remote terminal.

To log into the system:

1. Enter your login name and press **Enter**.
2. Enter your password and press **Enter**.

   For security, your password does not display as you type it.
3. Enter the kind of terminal you have or the type your system emulates and press **Enter**.

   The Command prompt displays.

   Note:

   If you enter the wrong terminal type, it can lock up your system. If the system is not responding to keyboard commands, type **`newterm`** and press **Enter**. Enter the correct terminal type on the new screen and press **Enter**. If this does not work, turn the power off only on the terminal and then turn it back on. The terminal reboots and you can log in again.

Logging in for remote administration

To log in for remote administration:

1. Dial the Uniform Call Distribution (UCD) group extension number.

   Note:

   The UCD group extension number is assigned when you set up remote administration.

   - If you are off-premises, use the Direct Inward Dialing (DID) number, a Listed Directory Number (LDN) (you must use a telephone), or the trunk number dedicated to remote administration.
   - If you are on-premises, use an extension number.

     If you dialed a DID number, dedicated trunk number, or extension, you receive data tone or visually receive answer confirmation.

     If an LDN was dialed, the attendant will answer.

     a. Ask to be transferred to the UCD group extension number.

        You receive data tone or visually receive answer confirmation.

     b. Transfer the voice call to your data terminal.

   The Login prompt displays.

2. Complete the steps for Logging into the System on page 21.

   For information about setting up remote administration, contact your Avaya technical support representative.

Failed login attempts

SAT login failure

If you are using the System Access Terminal (SAT), you can use the Security Violations Notification feature to disable a user's login if a threshold of failed login attempts is reached within a specified time period. Administer these parameters on the **Security-Related System Parameters** screen. For instance, if you set the **Login Threshold** to 3 and the **Time Interval** to 10 minutes, and there are 3 failed login attempts in 10 minutes or less, that login is disabled. To re-enable a login ID that has been disabled by a security violation, type **`enable login`** ***nnnnn***, where ***nnnnn*** is the login you want to re-enable. You can view SVN-disabled logins on the "monitor security-violations login" report, or by typing **`list logins`**.

For more information on setting up SVN for SAT login lockout, see the "Screen Reference" chapter, Login Administration and Security-Related System Parameters sections. See also Chapter 11: Enhancing System Security. For a complete description of the Security Violation Notification feature, see "Security Violation Notification " in *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205.

Server login failure

Communication Manager has an automatic lockout mechanism that is activated when there are 4 login failures in 10 minutes or less. The login will be locked out for 5 minutes. An account lockout event is logged to the security log. This type of lockout applies for all shell and Web access logins, including SAT logins.

Accessing the S8700 or S8710 Media Server

To administer the S8700 or S8710 Media Server, you must be able to access it. Personal computers and services laptop computers equipped with a network PCMCIA card, Avaya Site Administration (ASA), and a Web browser are the primary support access for system initialization, aftermarket additions, and continuing maintenance.

You can access the S8700 or S8710 Media Server in one of three ways:

- directly
- remotely over the customer's local area network (LAN)
- over a modem

A direct connection and over the customer's LAN are the preferred methods. Remote access over a modem is for Avaya maintenance access only.

Accessing the S8700 or S8710 Media Server Directly

You can access the S8700 or S8710 Media Server directly by plugging a computer into the services port which defaults to port 2 (Eth1) on the back of the media server. You must use a crossover cable with an RJ45 connector on each end. Plug the other end into the network connector (NIC card) on the your computer. You may need a NIC card adapter.

Once connected, you can administer the media server using three tools:

- Media server web interface for server-specific administration.
- ASA for various features of Avaya Communication Manager
- A command line interface in a Telnet window for features of Avaya Communication Manager.

Media Server Web Interface

You can access the media server web interface either by connecting directly to the services port on the media server or connecting over the customer's network.

Connected to the services port

To use the media server web interface:

1. Open either the Netscape or MS Internet Explorer browser.
2. In the **Location/Address** field, type **192.11.13.6** and press **Enter**.
3. When prompted, log in to administer the S8700 or S8710 Media Server and the features of Avaya Communication Manager.

Connected to the customer network

To use the media server web interface:

1. Open either the Netscape or MS Internet Explorer browser.
2. In the **Location/Address** field, type the active server name or IP address and press **Enter**.
3. When prompted, log in to administer the S8700 or S8710 Media Server and the features of Avaya Communication Manager.

   You may also connect directly to an individual server using its name or IP address.

Avaya Site Administration

To use ASA:

1. You must have a PC or laptop with Avaya Site Administration installed on it.
2. You must be connected over a LAN to either:
   - a dedicated LAN port for service technicians, or
   - a customer LAN port

Instructions

1. Start up ASA by double-clicking the ASA icon, or click **Start>Programs>Avaya Site Administration**.
2. In the **Target System** field, use the pull-down menu to select the desired system.

   If you have not set up the desired system, click the **New VOICE SYSTEM** icon (on the left of the toolbar row). Complete the **Add Voice System** wizard to build the system record. When completed, the record will be available in the **Target System** pull-down list.
3. Click **Start GEDI**.

   You now are connected to the desired system.

Installing Avaya Site Administration

If you do not have ASA on your computer, make sure your personal computer (PC) or laptop first meets the following minimum requirements:

Operating systems Windows 2000 Windows XP 2003 (Standard and Enterprise)	**Processor/RAM** Pentium-class 300 MHz/64 MB Pentium-class 300 MHz/64 MB Pentium-class 300 MHz/64 MB
Graphics adapter	SVGA with minimum screen resolution of 800 x 600
Floppy disk drive	3-1/2 in. 1.44-MB floppy disk drive
CD-ROM	CD-ROM drive (required to install ASA from CD)
Available hard disk space	A minimum of 100-MB free hard disk space is required. The requirement for disk space depends on the size and number of configuration data sets.
Printer port	Standard PC or network printer port is required for connecting to a printer to produce station button labels.
Network adapter	Required for network access to the S8700 or S8710 Media Server, AUDIX, and other network-connected systems.
Free serial ports	One free serial port capable of 9600-bps operation is required for a connection to each serial device (UPS). Avaya recommends that PCs have at least a 16550A UART or 16550A UART simulator (capable of 56 kbps DTE-speed connections). USB and internal modems should emulate this hardware. A second serial port is required for simultaneous connection to AUDIX through a serial connection.

Install ASA on your computer using the Avaya Site Administration CD. Place the ASA CD in the CD-ROM drive and follow the installation instructions in the install wizard.

ASA supports a terminal emulation mode, which is directly equivalent to using SAT commands on a dumb terminal or through a Telnet session. ASA also supports a whole range of other features, including the graphically enhanced interface (GEDI) and Data Import. For more information see the Help, Guided Tour, and Show Me accessed from the ASA Help menu.

Configuring Avaya Site Administration

When Avaya Site Administration is initially installed on a client machine, it needs to be configured to communicate with Communication Manager on the Media Server.

When it runs initially, after downloading, you need to create a new entry for the switch connection. To create new entries for the switch, use the following procedure.

Adding a Switch Administration Item

1. Click **File > New > Voice System**.

   The system displays the **Add Voice System** screen .
2. Enter a name in the Voice System Name: field.
3. Click **Next**. The connection type dialog box displays.
4. Click the **Network connection** radio button.
5. Click **Next**. The Network Connection dialog box displays.
6. Enter the IP address used to connect to the Media Server.
7. Click **Next**. The Network Connection/Port Number dialog box displays.
8. TCP/IP Port Number: For the port number, enter port **5023**.
9. Click **Next**. The Network Connection/Timeout Parameters dialog box displays. Leave the default values for the timeout parameters.
10. Click **Next**. The login type dialog box displays.
11. Click the **"I want to login manually each time"** radio button.
12. Click **Next**. The switch summary dialog box displays.
13. Check the information, use the **Back** button to make corrections if necessary, and click the **Test** button to test the connection.
14. When the connection is successfully tested, click **Next** and then **Finish**.

Accessing the S8700 or S8710 Media Server Remotely over the network

You can access the S8700 or S8710 Media Server from any computer connected through the LAN. To access either server, use the IP address assigned to the server you want to access. You can also use the active server address to connect automatically to the server that is active. Once connected, you can administer the media server using three tools:

- Media server web interface for server-specific administration and call processing features
- Avaya Site Administration for Communication Manager (Only available on the active Communication Manager server)
- A command line interface via a Telnet window for Linux shell commands.

Logging in with Access Security Gateway

Access Security Gateway (ASG) is an authentication interface used to protect the system administration and maintenance ports and logins associated with Avaya Communication Manager. ASG uses a challenge and response protocol to validate the user and reduce unauthorized access.

You can administer ASG authentication on either a port type or login ID. If you set ASG authentication for a specific port, it restricts access to that port for all logins. If you set ASG authentication for a specific login ID, it restricts access to that login, even when the port is not administered to support ASG.

Authentication is successful only when Avaya Communication Manager and the ASG communicate with a compatible key. You must maintain consistency between the Access Security Gateway Key and the secret key assigned to the Communication Manager login. For more information about ASG, see Using Access Security Gateway on page 403.

Before you start

Before you can log into the system with ASG authentication, you need an Access Security Gateway Key, and you need to know your personal identification number (ASG). The Access Security Gateway Key must be pre-programmed with the same secret key (such as, ASG Key, ASG Passkey, or ASG Mobile) assigned to the Avaya Communication Manager login.

Verify that the **Access Security Gateway (ASG)** field on the **System-Parameters Customer Options** screen is set to **y**. If not, contact your Avaya representative.

Instructions

To log into the system with ASG:

1. Enter your login ID and press **Enter**.

   The system displays the challenge number (for example, 555-1234) and system Product ID number (for example, 1000000000). The Product ID provides Avaya Services with the specific identifier of your Avaya MultiVantage communications application.

2. Press **ON** to turn on your Access Security Gateway Key.
3. Type your PIN and press **ON**.

   The Access Security Gateway Key displays a challenge prompt.

4. At the challenge prompt on the Access Security Gateway Key, type the challenge number without the "-" character (for example, 5551234) from your screen and press **ON**.

   The Access Security Gateway Key displays a response number (for example, 999-1234).

5. At the response prompt on your terminal, type the ASG response number without the "-" character (for example, 9991234) and press **Enter**.

   The Command prompt displays.

   Note:

   If you make 3 invalid login attempts, the system terminates the session. For more information, see the appropriate maintenance book for your system.

Fixing problems

Note:

When logging in failures occur, if you are a super-user, you can use the **`list asg-history`** command to determine the cause. The ASG history log contains the last 100 or 250 records depending on your system. This log contains the date and time, the port mnemonic, the login ID entered (correct or incorrect), and the status for each session attempt. For specific information about the ASG history log, see *Reports for Avaya Communication Manager*.

Logging off the System

For security, log off any time you leave your terminal. If you use terminal emulation software to administer Communication Manager, log off the system and exit the emulation application before switching to another software package.

Instructions

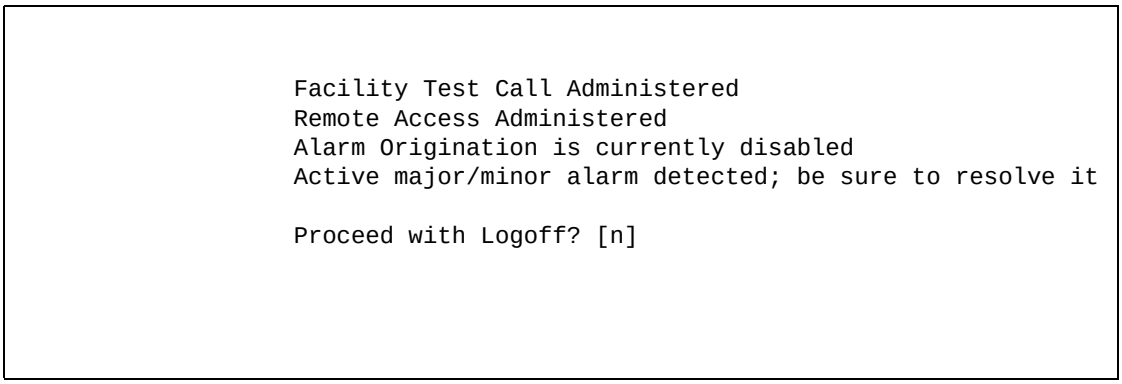
To log off:

1. Type **`logoff`** and press **Enter**.

   If the Facility Test Call or Remote Access features are administered, Alarm origination is disabled, or if you have busied out resources or active minor or major alarms, a security screen displays. You may want to take appropriate action (for example, disable these features or address any alarms) before you log off.

   If none of the above special circumstances exist, the system logs you off.

Figure 1: Logoff screen



2. At the **Proceed with Logoff** prompt, type **y** to log off.

   If you log off with alarm origination disabled and the system generates an alarm, Avaya support services will not receive any notification of the alarm. For more information about alarms, see the maintenance book for your system.

Setting Command Permissions

Avaya Communication Manager allows you to modify the permissions associated with a login. The system maintains default permissions for each level of login, but you may want to further restrict the login, or at least make sure the defaults are appropriate for the user. The default values for these fields vary based on the login type.

When set to **y**, the permissions on the Command Permission Categories screen apply for any object that is not restricted. The second and third pages of the Command Permission Categories screen allow you to restrict the user from any access to specified objects. If you want to limit a user's permissions beyond those on page one, enter the objects in this list. For example, if you want a user to be able to add and change stations, but not VDNs, you can enter **y** in the **Administer Stations** field and the **Additional Restrictions** field. Then on page 2 or 3, enter **vdn** as a restricted object.

Instructions

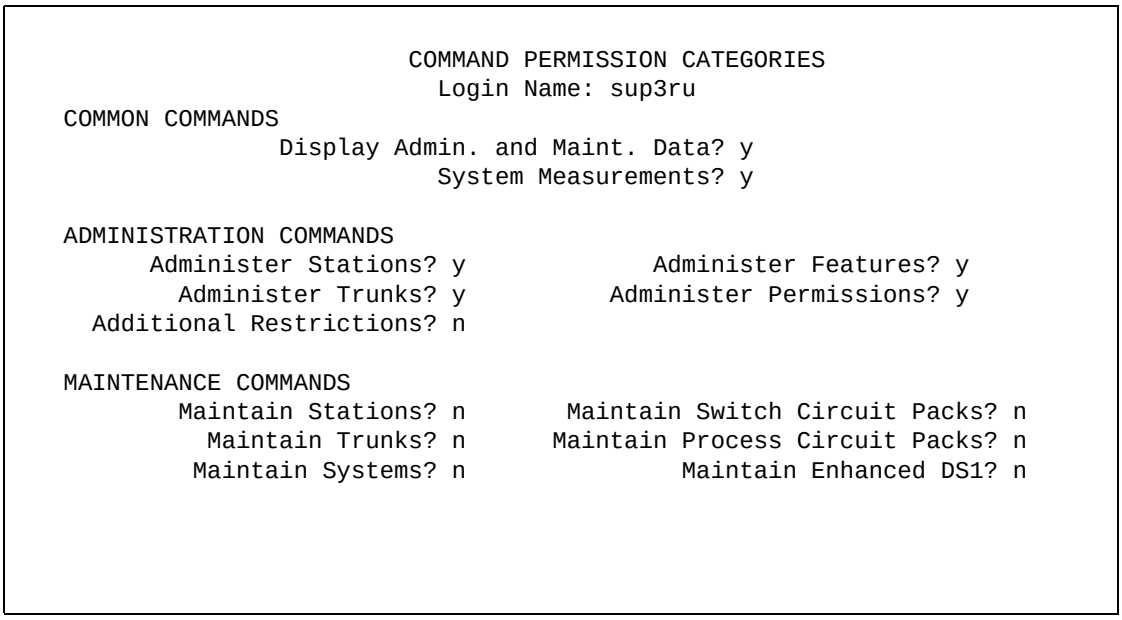
In our example, we set the permissions necessary to allow the user to administer daylight savings time rules.

To change command permissions:

1. Type **change permissions sup3ru** and press **Enter**.

   The **Command Permission Categories** screen displays.

Figure 2: Command Permission Categories screen



```
                        COMMAND PERMISSION CATEGORIES
                          Login Name: sup3ru
COMMON COMMANDS
               Display Admin. and Maint. Data? y
                          System Measurements? y

ADMINISTRATION COMMANDS
      Administer Stations? y             Administer Features? y
        Administer Trunks? y           Administer Permissions? y
  Additional Restrictions? n

MAINTENANCE COMMANDS
        Maintain Stations? n       Maintain Switch Circuit Packs? n
          Maintain Trunks? n      Maintain Process Circuit Packs? n
         Maintain Systems? n               Maintain Enhanced DS1? n
```

2. Type **y** in the **Display Admin and Maint Data** field.
3. Type **y** in the **Administer Features** field.
4. Press **Enter** to save your work.

More information

There are 2 types of users — superuser and non-superuser.

- A superuser provides access to the add, change, display, list, and remove commands for all customer logins and passwords. The superuser can administer any mix of superuser/non-superuser logins. The superuser can administer between 10 and 19 logins depending on your system.

  Logins must be 3 to 6 alphabetic/numeric characters, or a combination of both.

  Note:

  If several users are logging in and out at the same time, a user may see the message: **Transient command conflict detected; please try later**. After the users have completed logging in or out, the terminal is available for use.

Note:

Non-superusers may change their password with permission set by the superuser. However, once a password has been changed, a non-superuser must wait 24 hours before changing the password again.

Establishing Daylight Savings Rules

Avaya Communication Manager allow you to set the daylight savings time rules so that features, such as time-of-day routing and call detail recording (CDR), adjust automatically to daylight savings time. The correct date and time ensure that CDR records are correct. You can set daylight savings time rules to transition to and from daylight savings time outside of normal business hours, so the number of affected CDR records is small.

You can set up 15 customized daylight savings time rules. This allows Communication Manager administrators with servers in several different time zones to set up a rule for each. A daylight savings time rule specifies the exact time when you want to transition to and from daylight savings time. It also specifies the increment at which to transition (for example, 1 hour).

Instructions

Establishing daylight savings rules

In our example, we set daylight savings time rules.

To modify a daylight savings rule:

1. Type **`change daylight-savings-rules`** and press **Enter**.

   The **Daylight Savings Rules** screen appears.

Figure 3: Daylight Savings Rules screen

```
DAYLIGHT SAVINGS RULES
Rule          Change Day                  Month___Date Time____Increment

0:    No Daylight Savings

1:    Start: first Sunday___ on or after April___ 1  at _2:00  01:00
      Stop:  first Sunday___ on or after October_25  at _2:00
2:    Start: first _________ on or after ________ __ at __:__
      Stop:  first _________ on or after ________ __ at __:__
3:    Start: first _________ on or after ________ __ at __:__
      Stop:  first _________ on or after ________ __ at __:__
4:    Start: first _________ on or after ________ __ at __:__
      Stop:  first _________ on or after ________ __ at __:__
5:    Start: first _________ on or after ________ __ at __:__
      Stop:  first _________ on or after ________ __ at __:__
6:    Start: first _________ on or after ________ __ at __:__
      Stop:  first _________ on or after ________ __ at __:__
7:    Start: first _________ on or after ________ __ at __:__
      Stop:  first _________ on or after ________ __ at __:__
```

2. Complete the **Start** fields for rule 1.

   Rule 1 applies to all time zones in the U.S. and begins on the first Sunday on or after 1 April at 2:00 a.m. with a 01:00 increment. Daylight Savings Time stops on the first Sunday on or after 25 October at 2:00 a.m. also with a 01:00 increment (used as a decrement when switching back to Standard time.

   The increment is added to standard time at the specified start time and the clock time shifts by that increment (for example, for 01:59:00 to 01:59:59 the clock time shows 01:59 and at 02:00 the clock shows 03:00).

   On the stop date, the increment is subtracted from the specified stop time (for example, for 01:59:00 to 01:59:59 the clock time shows 01:59 and at 02:00 the clock shows 01:00).

   a. Type **Sunday** in the **Change Day** field.

   b. Type **April** in the **Month** field.

   c. Type **1** in the **Date** field.

   d. Type **2:00** in the **Time** field.

   e. Type **1:00** in the **Increment** field.

      This information specifies the day, month, date, and time and increment at which you want the system clock to transition to daylight saving time.

Note:

You cannot delete a daylight savings rule if it is in use on either the **Locations** or **Date and Time** screens. However, you can change any rule except rule 0 (zero).

3. Complete the **Stop** fields for rule 1.
   a. Type **Sunday** in the **Change Day** field.
   b. Type **October** in the **Month** field.
   c. Type **25** in the **Date** field.
   d. Type **2:00** in the **Time** field.

   This information specifies the day, month, date, and time you want the system clock to transition back to standard time.
4. Press **Enter** to save your changes.

Displaying daylight savings time rules

To display daylight savings time rules:

1. Type **`display daylight-savings-rules`** and press **Enter**.

   The **Daylight Savings Rules** screen appears. Verify the information you entered is correct.

Setting the system date and time

Update the date and time for events such as a leap year, the change to or from daylight savings time, or a system restart after a power failure. The correct date and time assure that CDR records are correct. CDR does not work until the date and time have been entered.

Note:

Changing the date and time may modify CDR data by 9 hours and 59 minutes. Therefore, you should change the date and time after normal business hours. After you change the date and time, review the time settings for any adjunct (other than AUDIX) linked to your system that uses the system time.

Before you start

Before you can set the date and time, you need to know whether it is currently daylight savings or standard time and know which daylight savings rule number you want to use. Daylight savings rule numbers are located on the **Daylight Savings Rules** screen.

Setting Time of Day Clock Synchronization

Time of Day Clock Synchronization enables a server to synchronize its internal clock to UTC time provided by Internet time servers. Avaya uses the LINUX platform system clock connected to an Internet time server to provide time synchronization. The interface for these systems is web-based.

LINUX is used in:

- S8700 or S8710 IP-Connect
- S8700 or S8710 Multi-Connect
- S8700 or S8710 Media Server

For additional information, see *Avaya Communication Manager Contact Center Guide to ACD Contact Centers*.

The following systems use Avaya Site Administration (ASA) for time synchronization.

- DEFINITY Server R
- DEFINITY Server CSI
- DEFINITY Server SI

Before you start

- A standard TCP/IP LAN connection is required to connect to the Internet time servers. If a LAN connection is not available, time sync will be done by setting the platform clock manually through the command line or web interface.
- On the target server running Communication Manager, verify if Daylight Savings Time is on.

Instructions

Note:

These instructions are for Avaya DEFINITY R, DEFINITY CSI, and DEFINITY SI servers. For additional information, see *Avaya Communication Manager Contact Center Guide to ACD Contact Centers*.

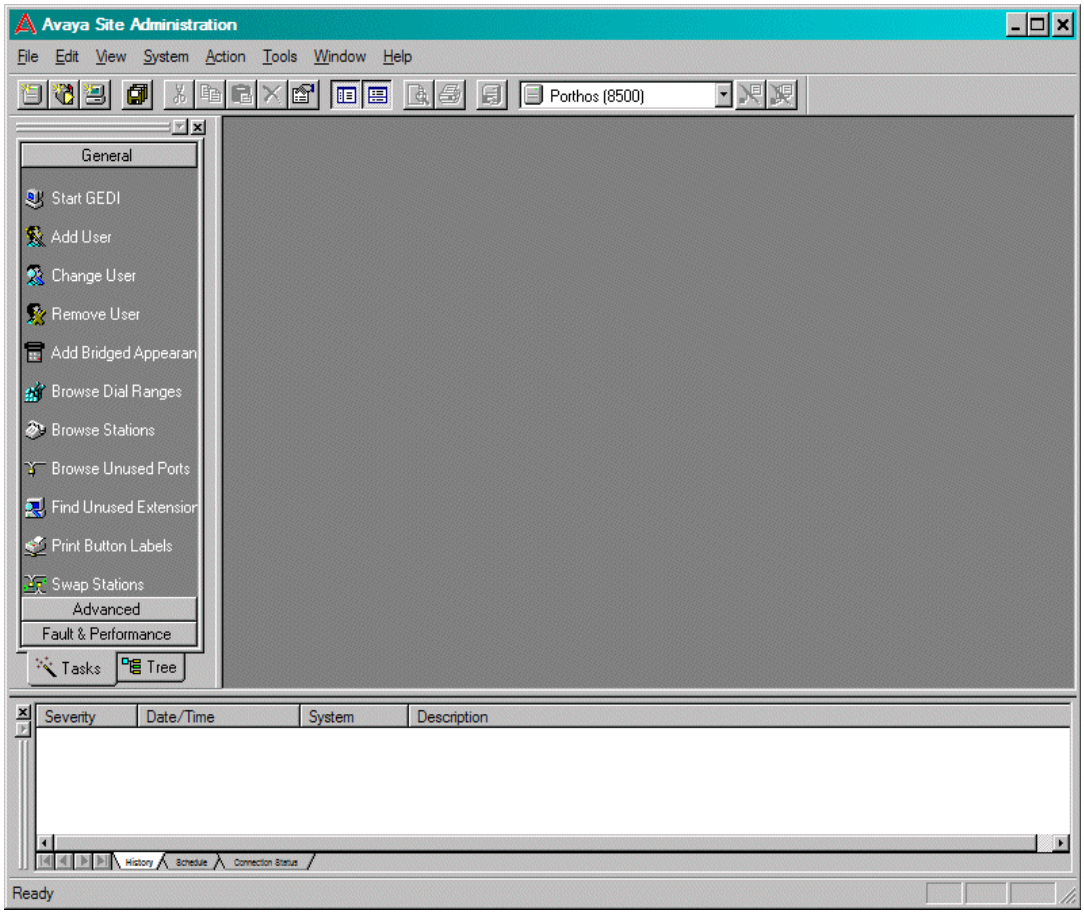
To set Time of Day Clock Synchronization:

1. Activate the **Avaya Site Administration** screen.

2. Click the **Fault & Performance** tab.

   The **Fault & Performance** icons display.

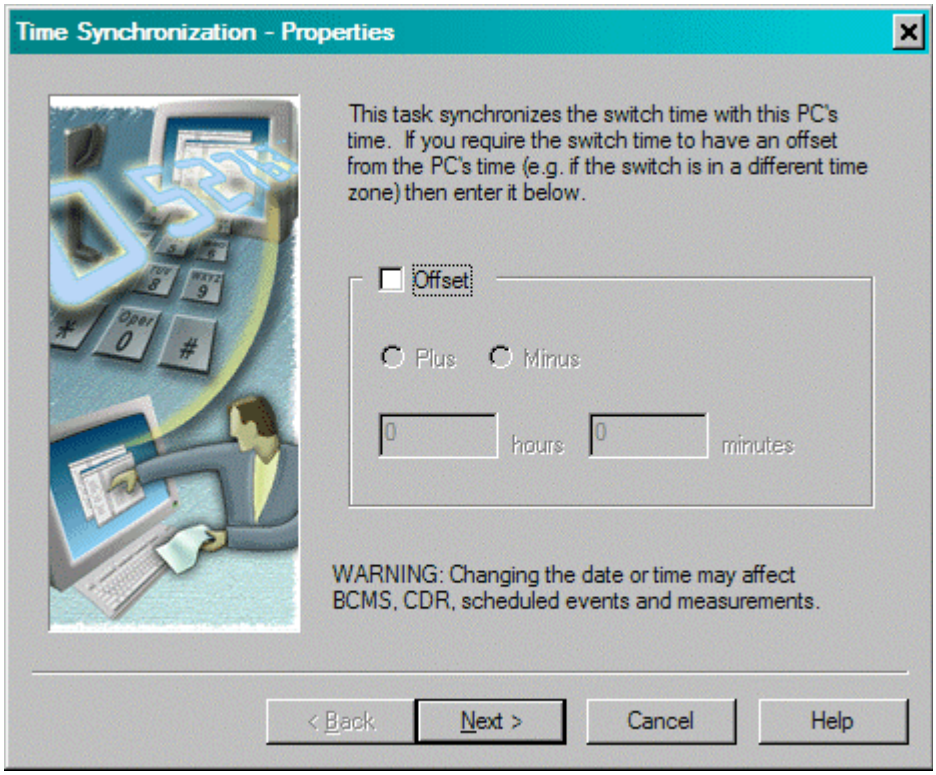
Figure 4: Avaya Site Administration menu



3. Click **Time Synchronization**.

   The **Time Synchronization - Properties** screen displays.

Figure 5: Time Synchronization - Properties screen



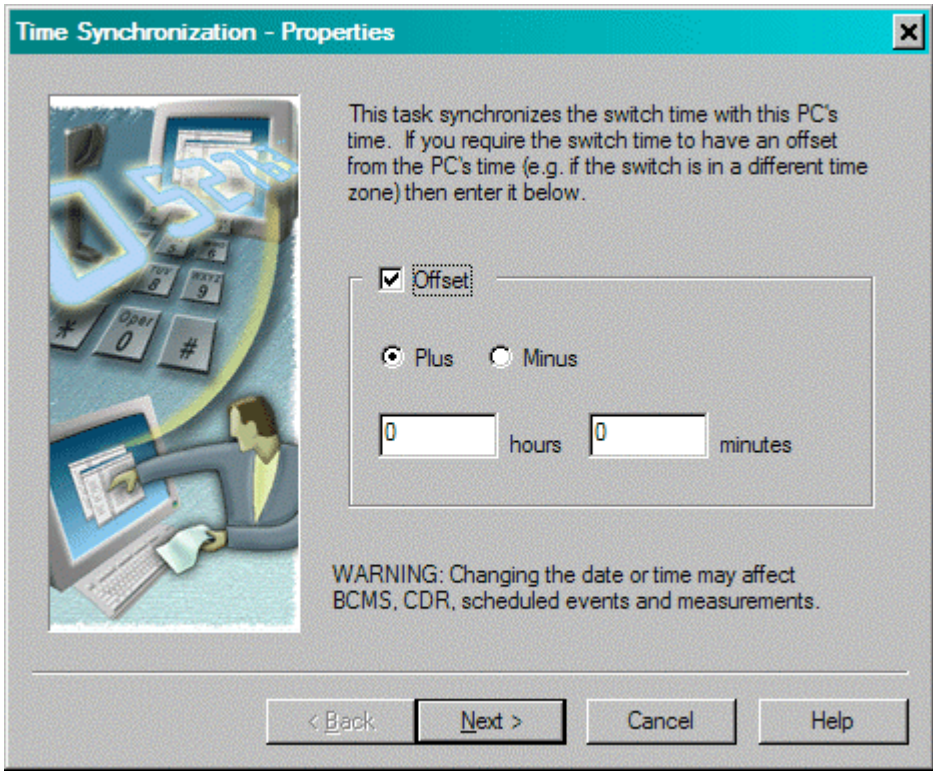
4. Click in the **Offset** box.

   The **Plus** and **Minus** radio buttons and the **Hours and Minutes** fields display.

5. Click **Next**.

   The **Time Synchronization - Properties** screen displays.

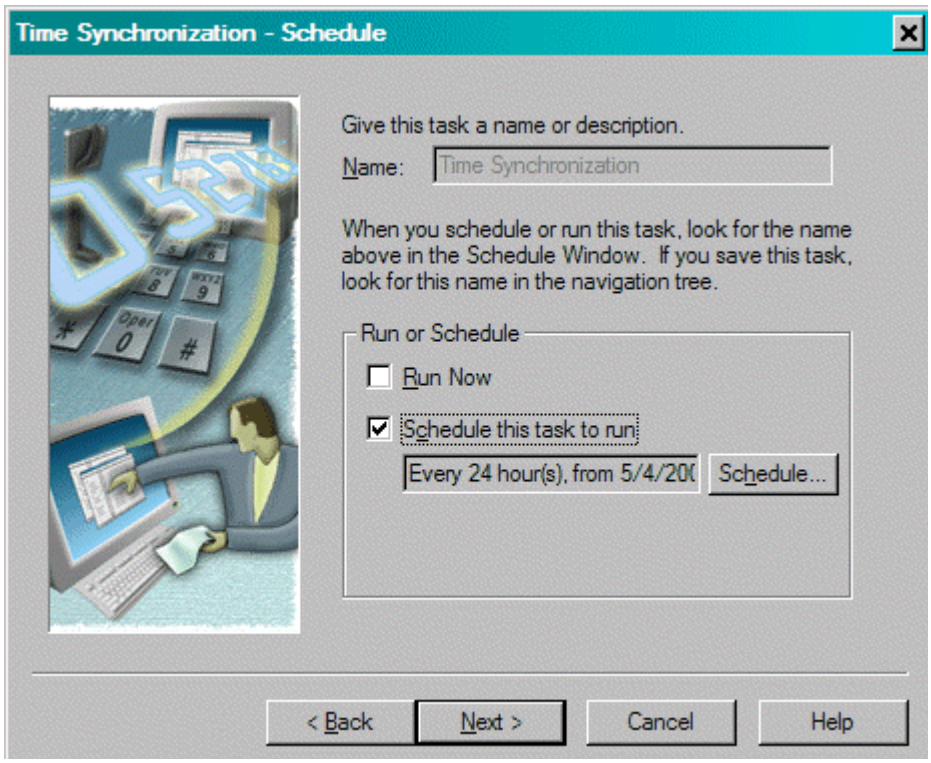
Figure 6: Time Synchronization - Properties screen



6. Click **Plus** to add hours to the remote station (located to the west of the system time) or click Minus to subtract hours to the remote station (located to the east of the system time).
7. In the **hours** field, enter the number of hours to be added or subtracted to synchronize with the remote site.
8. Click **Next**.

   The **Time Synchronization - Schedule** displays.

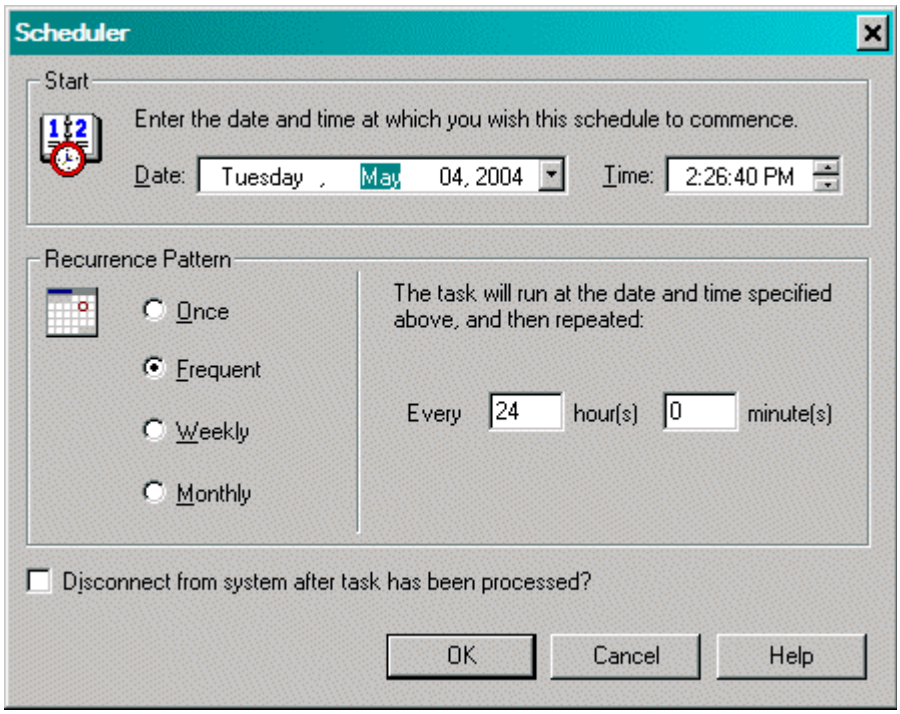
Figure 7: Time Synchronization - Schedule screen



9. Select either:
   - **Run Now** to run this program immediately and click **Next**.
   - **Schedule this task to run** and check the field below to determine if the default setting is satisfactory. If this setting is not satisfactory, click **Schedule**.

   The **Scheduler** screen displays.

Figure 8: Scheduler screen

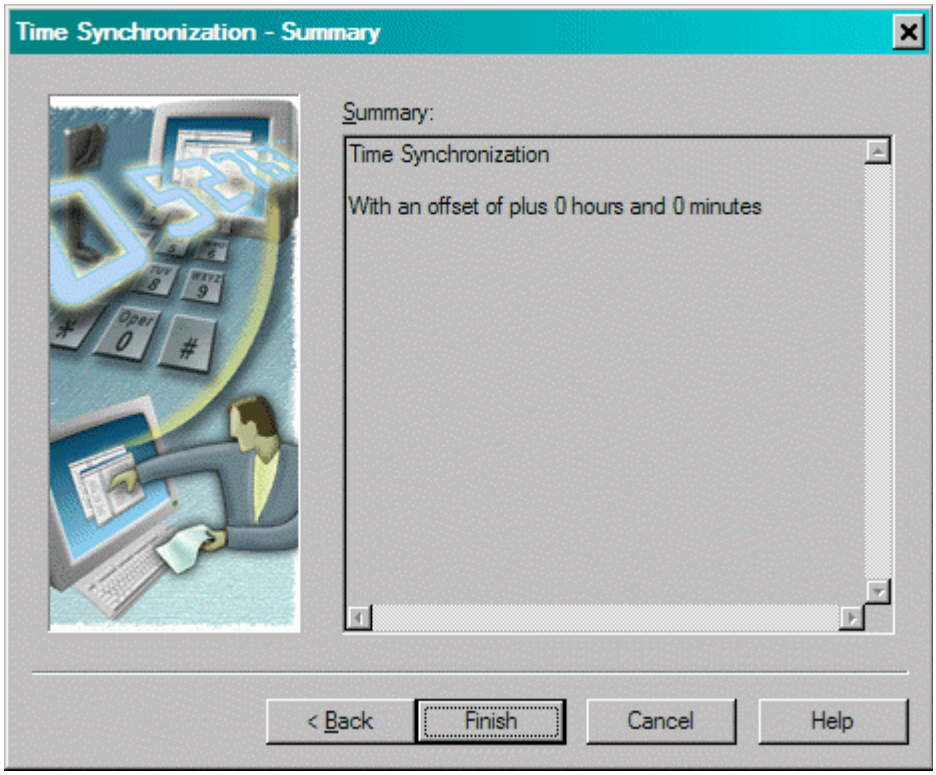


10. In the **Date** field, highlight each object and use the pull-down menu to select the desired setting.
11. In the **Time** field, highlight each item of time and use the pull-down menu to select the desired time.
12. In the **Recurrent Pattern** area, select one radio button for the desired frequency. Notice that each radio button is explained in the area to the immediate right.
13. If there are times to be included, set to the desired frequency.
14. Click **OK**.

    The **Time Synchronization - Schedule** displays.
15. Click **Next**.

    The **Time Synchronization - Summary** screen displays.

Figure 9: Time Synchronization - Summary screen



16. If the time synchronization description is satisfactory, click **Finish**.

    If the time synchronization is not satisfactory, click **Back** and revise the necessary information.

Related topics

See *Avaya Communication Manager Contact Center Guide to ACD Contact Centers,* for more information about using time of day clock synchronization.

Setting the System Date and Time

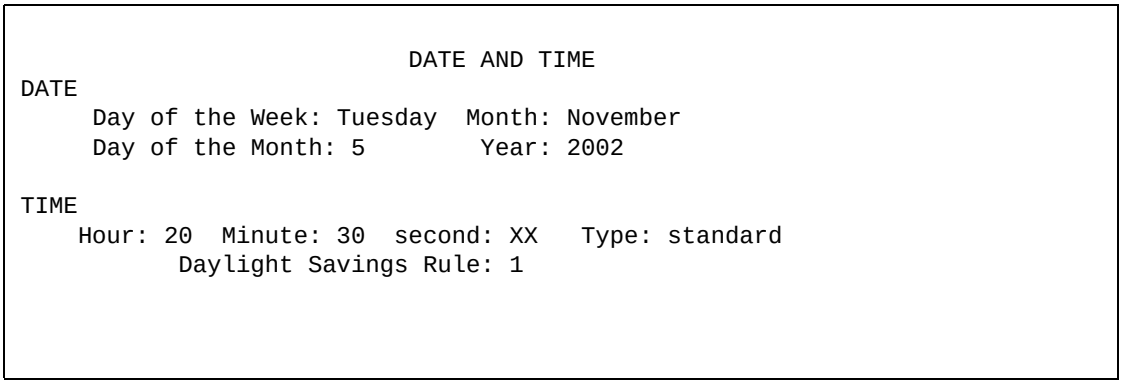
In our example, we set the date and time to Tuesday, November 5 at 8:30 p.m. standard time.

To set the system date and time:

1. Type **set time** and press **Enter**.

   The **Date and Time** screen displays.

Figure 10: Date and Time screen



2. Complete the **Date** fields.
   a. Type **Tuesday** in the **Day of the Week** field.
   b. Type **November** in the **Month** field.
   c. Type **5** in the **Day of the Month** field.
   d. Type **2002** in the **Year** field.
3. Complete the **Time** fields.

   Use the 24-hour clock to set the hour, so if the current time is 2:00 p.m., you enter **14:00**. You cannot update **Second** — it automatically resets to 0 when you save your changes.
   a. Type **20** in the **Hour** field.
   b. Type **30** in the **Minute** field (8:30 p.m.).
   c. Type **standard** in the **Type** field.
   d. Type **1** in the **Daylight Savings Rule** field.
4. Press **Enter** to save your changes.

   Note:

   When you change the date or time, some display phones may not automatically refresh the display. If this occurs, have each user press the date/time button on their telephone to update the display.

Displaying the system date and time

To display the system date and time:

1. Type **`display time`** and press **Enter**.

   The **Date and Time** screen displays. Verify the information you entered is correct.

Related topics

See Establishing Daylight Savings Rules on page 31 for more information about setting system time.

For additional information, see *Avaya Communication Manager Contact Center Guide to ACD Contact Centers,* 555-230-716.

Using the Bulletin Board

Avaya Communication Manager allows you to post information to a bulletin board. You can also display and print messages from other Avaya server administrators and Avaya personnel using the bulletin board. Anyone with the appropriate permissions can use the bulletin board for messages. Only one user can post or change a message at a time.

Whenever you log in, the system alerts you if you have any messages on the bulletin board and the date of the latest message. Also, if Avaya personnel post high-priority messages while you are logged in, you receive notification the next time you enter a command. This notification disappears after you enter another command and reoccurs at login until deleted by Avaya personnel.

You maintain the bulletin board by deleting messages you have already read. You cannot delete high-priority messages. If the bulletin board is at 80% or more capacity, a message appears at login indicating how much of its capacity is currently used (for example, 84%). If the bulletin board reaches maximum capacity, new messages overwrite the oldest messages.

Note:

The bulletin board does not lose information during a system reset at level 1 or level 2. If you save translations, the information can be restored if a system reset occurs at levels 3, 4, or 5.

Instructions

Displaying messages

To display the bulletin board:

1. Type **display bulletin-board** and press **Enter**.

   The **Bulletin Board** screen displays.

Figure 11: Bulletin Board screen

```
Message (* indicates high-priority)                  Date
*Avaya is in the process of                         03/02/98
*investigating your trunk lockup problem.           03/02/98
*The Bulletin Board will be updated as              03/02/98
*we find information.                               03/02/98
*
*
*
*
*
*
We recently added a new trunk group (14)            03/02/98
and have had many of the members getting            03/02/98
locked up.                                          03/02/98
```

Posting a message

In our example, we post a message to the bulletin board about a problem with a new trunk group, and an Avaya representative replies to our message.

To post a message to the bulletin board:

1. Type **change bulletin-board** and press **Enter**.

   The **Bulletin Board** screen displays. There are three pages of message space within the bulletin board. The first page has 19 lines, but you can only enter text on lines 11-19. The first 10 lines on page 1 are for high-priority messages from Avaya personnel and are noted with an asterisk (*). The second and third pages each have 20 lines, and you can enter text on any line. The system automatically enters the date the message was posted or last changed to the right of each message line.

2. Type your message.

   You can enter up to 40 characters of text per line. You also can enter one blank line. If you enter more than one blank line, the system consolidates them and displays only one. The system also deletes any blank line if it is line one of any page. You cannot indent text on the bulletin board. The **Tab** key moves the cursor to the next line.

3. Press **Enter** to save your changes.

Deleting messages

To delete a message from the bulletin board:

1. Type **change bulletin-board** and press **Enter**.

   The **Bulletin Board** screen appears.

2. Enter a space as the first character on each line of the message you want to delete and press **Enter**.

3. Press **Enter** to save your changes.

Saving Translations

Communication Manager retains all translation data in memory while the system is operating. If it goes down, you lose all this translation data. You must save in-memory translation data to the memory card (flash ROM), disk, or tape. Saving translation data to memory card or tape is the same as backing up your system.

Note:

Save translations on a daily basis. You may want to save translations after business hours to prevent dial tone delays or during slow business hours if your business is open 24 hours.

The save translation command writes two time-stamped identical copies of the translation data to the selected memory card, disk, or tape. The save writes one complete copy first, then writes the second copy in a different area of the device — both with the same time-stamp. Failure during a save, including a system crash, usually affects only one copy. The affected copy is marked "bad" and should not be used for backup.

You can set save translation to be completed automatically as part of regularly scheduled maintenance or manually, as needed. For more information about saving translations automatically, see the maintenance book for your system.

Tip:

To determine if your system saves translations automatically, type **display system-parameters maintenance** to see if you have scheduled maintenance.

Translation copy protection assigns a number to a specific telephone system and to the flash card or set of flash cards that belong to that system. For Avaya Communication Manager on a DEFINITY Server CSI or Communication Manager on a DEFINITY Server SI, this number is the same on both the translation storage device (flash card) and the Flash PROM (Programmable Read Only Memory) of the processor circuit pack. In a duplicated system, the Flash PROM of each processor circuit pack has a translation ID and both ID's are stored on the memory card.

An attempt to initialize (boot) the system with translations that do not contain the same identification number as stored in the processor circuit pack raises a major alarm and disables access to the save translations command for all non-Avaya logins. You also receive a warning message on the copyright screen notifying you of the mismatch. Contact your Avaya representative to correct this mismatch and reset the save translations command. You must correct the mismatch before the end of the specified grace period, otherwise you cannot access system management commands (such as: add, change, remove, and duplicate) that modify translation data. Avaya specifies the grace period during a system installation or following an upgrade.

Before you start

If you are saving translations to a memory card or tape, you must verify the memory card or tape is in place and clear any active alarms from the alarms panel.

If you have Avaya Communication Manager on a DEFINITY Server CSI or Communication Manager on a DEFINITY Server SI, verify the memory card translation ID matches the translation ID of your server's Flash PROM.

Instructions

In our example, we save translations to the tapes on both processor A and B.

To save translations manually:

1. Type **save translation both tape** and press **Enter**.

   The save process can take up to 10 minutes. You cannot administer your system while the save is in process. The **Save Translation** screen displays.

Figure 12: Save Translation screen

```
SAVE TRANSLATION


Processor          Command Completion Status              Error Code
SPE_A                                  Success            0
SPE_B                                  Success            0
```

2. If there is an error message in the **Command Completion Status** field and an error code in the **Error Code** field, clear the error and repeat the save process.

More information

When mass storage system (MSS) devices on both processors in a duplex system are specified, translation data is saved from the active processor to the active and standby MSS devices at the same time. If the save to one device fails or one device is out of service, the other save continues. You receive the status of each save separately.

Note:

If you have a duplex system and you save translation data to both MSS devices one at a time, translation data inconsistencies between the two devices can occur.

Fixing problems

Note:

You cannot see whether the translation ID on the flash card corresponds to the number on the Processor circuit packs. However, if the numbers do not match, the system issues an error message when you attempt a save translation operation.

When failures occur, the system responds with the following error codes.

Problem	Possible causes	Solution
1	Save translation cannot write to the active drive.	Repeat the save translation process for the active drive.
2	Save translation cannot write to the standby drive.	Repeat the save translation process for the standby drive.

For more information about error messages and how to correct them, see the maintenance book for your system.

Related topics

See your maintenance book for information about backing up or restoring your system.

See Saving Announcements on page 463 for information about backing up announcements for your system.

See Restoring Announcements on page 466 for information about restoring announcements to your system.

Performing Backups

Information on performing backups to your system can be found in the *Maintenance Procedures Guide*, 03-300192.

Chapter 2: Planning the System

Understanding Your Configuration

At a very basic level, Avaya Communication Manager consists of hardware to perform call processing, and the software to make it run. You use the administration interface to let the system know what hardware you have, where it is located, and what you want the software to do with it.

You can find out which circuit packs are in the system and which ports are available by entering the command **`list configuration all`**. There are variations on this command that display different types of configuration information. Use the help function to experiment, and see which command works for you.

To view a list of port boards on your system:

1. Type **list configuration port-network** and press **Enter**.

   The **System Configuration** screen appears.

Figure 13: System Configuration screen

```
 SYSTEM CONFIGURATION
Board                                                Assigned Ports
Number     Board Type Code     Vintage     u=unassigned t=tti p=psa

01A05      DIGITAL LINE          TN754B 000002      01 u  03 u  05 u  07 08
01A06      ANALOG LINE           TN742  000010      01 02 03 04 u  u  u  u
01B05      ANALOG LINE           TN746B 000008      u  u  u  u  u  u  u  u
                                                    u  u  u  u  u  u  u  u
01C04      ANALOG LINE           TN746B 000008      u  u  u  u  u  u  u  u
                                                    u  u  u  u  u  u  u  u
01C05      DIGITAL LINE          TN2224 000004      01 u  u  04 u  u  07 08
                                                    u  u  u  u  u  u  u  u
                                                    u  u  u  u  u  u  u  u
01C06      HYBRID LINE           TN762B 000004      01 02 u  u  u  u  u  u
01C10      DIGITAL LINE          TN754  000004      u  u  u  u  u  u  u  u
```

The **System Configuration** screen shows all the boards on your system that are available for connecting phones, trunks, data modules and other equipment. You can see the board number, board type, circuit-pack type, and status of each board's ports. The **u** entries on this screen indicate unused ports that are available for you to administer. These may also appear as **p** or **t**, depending on settings in your system.

You will find many places in the administration interface where you are asked to enter a port or slot. The port or slot is actually an address that describes the physical location of the equipment you are using.

A port address is made up of four parts:

- cabinet — the main housing for all the server equipment. Cabinets are numbered starting with 01.
- carrier — the rack within the cabinet that holds a row of circuit packs. Each carrier within a cabinet has a letter, A to E.
- slot — the space in the carrier that holds an individual circuit pack. Slots are numbered 01-16.
- port — the wire that is connected to an individual piece of equipment (such as a telephone or data module). The number of ports on a circuit pack varies depending on the type.

So, if you have a single-carrier cabinet, the circuit pack in slot 06 would have the address 01A06. If you want to attach a telephone to the 3rd port on this board, the port address is 01A0603 (01=cabinet, A=carrier, 06=slot, 03=port).

Understanding the Dial Plan

Your dial plan tells your system how to interpret dialed digits. For example, if you dial 9 on your system to access an outside line, it is actually the dial plan that tells the system to find an external trunk when a dialed string begins with a 9.

The dial plan also tells the system how many digits to expect for certain calls. For example, the dial plan may indicate that all internal extensions are 4-digit numbers that start with 1 or 2.

Let us take a look at an example dial plan so you'll know how to read your system's dial plan. The following figure shows an example of a simple dial plan.

Figure 14: Dial Plan Analysis Table screen

```
change dialplan analysis                                Page   1 of 1
                          DIAL PLAN ANALYSIS TABLE
                                                          Percent Full:    9

   Dialed  Total  Call                          Dialed  Total  Call
   String  Length Type                          String  Length Type
     0       1    attd                           __      _    ____
     1       3    dac                            __      _    ____
     3       5    ext                            __      _    ____
     4       6    ext                            __      _    ____
     5       7    ext                            __      _    ____
     *       3    fac                            __      _    ____
     #       3    fac                            __      _    ____
```

The **Dial Plan Analysis Table** defines the dialing plan for your system.

The **Call Type** column in the **Dial Plan Analysis Table** indicates what the system does when a user dials the digit or digits indicated in the **Dialed String** column. The **Total Length** column indicates how long the dialed string will be for each type of call. For example, this dial plan shows that when users dial a 5-digit number that starts with 3, they are dialing an extension.

The **Dial Plan Analysis Table** in our example contains the following call types:

- Attendant (attd) — Defines how users call an attendant. Attendant access numbers can be any number from 0 to 9 and contain 1 or 2 digits.

  In our example figure, the system calls an attendant when users dial 0.

- Dial access code (dac) — Allows you to use trunk access codes (TAC) and feature access codes (FAC) in the same range. For example, you could define the group 100 to 199, which would allow both FAC and TAC in that range. Dial access codes can start with any number from 1 to 9, * and #, and contain up to 4 digits.

  In our example figure, dial access codes begin with 1 and must be 3 digits long.

  Note:

  The **Dial Plan Analysis Table** does not allow you to enter a range specifically for trunk access codes. However, the **Trunk Group** screen still allows you to assign a TAC to a trunk group. The TAC you enter on the **Trunk Group** screen must match the format you have administered for a DAC on the **Dial Plan Analysis Table**.

- Extensions (ext) — Defines extension ranges that can be used on your system. In our figure, extensions must be in the ranges 30000 to 39999, 40000 to 49999 and 50000 to 59999.

- Feature access codes (fac) only — FAC can be any number from 1 to 9 and contain up to 4 digits. You can use * or #, but only as a first digit. In our example, feature access codes can begin with * or # and are 3-digits long.

The **Dial Plan Analysis Table** works with the **Dial Plan Parameters Table** for fully defining your dial plan. The **Dial Plan Parameters Table** allows you to set system-wide parameters for your dial plan.

Displaying your dial plan

You might want to take this opportunity to look at and interpret your own dial plan. To display your system's dial plan:

1. Type **`display dialplan analysis`** and press **Enter**.

Modifying your dial plan

It is easy to make changes to your dial plan. For example, we will add a new range of dial access codes to the dial plan. We want to be able to assign both FAC and TAC in the 700 to 799 range.

1. Type **`change dialplan analysis`** and press **Enter**.

   The **Dial Plan Analysis** appears.
2. Move the cursor to an empty row.
3. Type **7** in the **Dialed String** column. Press **Tab** to move to the next field.
4. Type **3** in the **Total Length** column. Press **Tab** to move to the next field.
5. Type **dac** in the **Call Type** column.
6. Press **Enter** to save your changes.

Adding extension ranges

You may find that as your needs grow you want a new set of extensions. Before you can assign a station to an extension, the extension must belong to a range that is defined in the dial plan. We will add a new set of extensions that start with 3 and are 4 digits long (3000 to 3999).

To add this set of extensions to the dial plan:

1. Type **`change dialplan analysis`** and press **Enter**.

   The **Dial Plan Analysis Table** screen appears.
2. Move the cursor to an empty row.
3. Type **3** in the **Dialed String** column. Press **Tab** to move to the next field.
4. Type **4** in the **Total Length** column. Press **Tab** to move to the next field.
5. Type **ext** in the **Call Type** column.
6. Press **Enter** to save your changes.

Uniform dial plan

You can also set up a Uniform Dialing Plan that can be shared among a group of servers. For more information, see *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205.

Multi-Location Dial Plan

When a customer migrates from a multiple independent node network to a single distributed server whose gateways are distributed across a data network, it may initially appear as if some dial plan functions are no longer available.

The multi-location dial plan feature preserves dial plan uniqueness for extensions and attendants that were provided in a multiple independent node network, but appear to be unavailable when customers migrate to a single distributed server. This feature is available beginning with Communication Manager, release 2.0.

For example, in a department store with many locations, each location might have had its own switch with a multiple independent node network. The same extension could be used to represent a unique department in all stores (extension 123 might be the luggage department). If the customer migrates to a single distributed server, a user could no longer dial 123 to get the luggage department in their store. The user would have to dial the complete extension to connect to the proper department.

Instead of having to dial a complete extension, the multi-location dial plan feature allows a user to dial a shorter version of the extension. For example, a customer can continue to dial 123 instead of having to dial 222-123.

Communication Manager takes leading digits of the location prefix and adds some or all of its leading digits (specified on the **Uniform Dial Plan** screen) to the front of the dialed number. The switch then analyzes the entire dialed string and routes the call based on the administration on the **Dial Plan Parameters** and **Dial Plan Analysis** screens.

Note:

Before you can administer the multi-location dial plan feature, the **Multiple Locations** field on the **Optional Features** screen must be enabled. To check if this is enabled, use the **`display system-parameters customer-options`** command. The **Multiple Locations** field is on page 3 of the **Optional Features** screen.

Prepending the location prefix to dialed numbers

Use the **Insert Digits** field on the **Uniform Dial Plan Table** screen to assign the location prefix from the caller's location on the **Locations** screen. The system adds some or all of its leading digits (specified on the **Uniform Dial Plan** screen) to the front of the dialed number. The switch then analyzes the entire dialed string and routes the call based on the administration on the **Dial Plan Parameters** screen.

- Non-IP phones and trunks inherit the location number of the hardware they are connected to (for example, the cabinet, remote office, or media gateway).
- IP phones indirectly obtain their location number.
  - A location number is administered on the **IP Network Region** screen that applies to all phones in that IP region.
  - If a **Location** field is left blank on an **IP Network Region** screen, an IP telephone derives its location from the cabinet where the CLAN board is that the telephone registered through.
- IP trunks obtain their location from the location of its associated signaling group. Either direct administration (only possible for signaling groups for remote offices), or the ways described for IP phones, determines the location.

Other options for the dial plan

You can establish a dial plan so that users only need to dial one digit to reach another extension. You can also establish a dial plan that allows users to dial, for example, two digits to reach one extension, and three digits to reach another. This is particularly useful in the hospitality industry, where you want users to be able to simply dial a room number to reach another guest.

Adding feature access codes

As your needs change, you may want to add a new set of FAC for your system. Before you can assign a FAC on the **Feature Access Code** screen, it must conform to your dial plan.

In our example, if you want to assign a feature access code of 33 to **Last Number Dialed**, first you need to add a new FAC range to the dial plan.

To add a FAC range from 30 to 39:

1. Type **`change dialplan analysis`** and press **Enter**.

   The **Dial Plan Analysis Table** screen appears.

2. Move the cursor to an empty row.
3. Type **3** in the **Dialed String** column. Press **Tab** to move to the next field.

4. Type **2** in the **Total Length** column. Press **Tab** to move to the next field.
5. Type **fac** in the **Call Type** column.
6. Press **Enter** to save your changes.

Changing feature access codes

Feature access codes (FAC) allow users to activate and deactivate features from their phones. A user who knows the FAC for a feature does not need a programmed button to use the feature. For example, if you tell your users that the FAC for the Last Number Dialed is *33, then users can redial a telephone number by entering the FAC, rather than requiring a Last Number Dialed button.

Many features already have factory-set feature access codes. You can use these default codes or you can change them to codes that make more sense to you. However, every FAC must conform to your dial plan and must be unique. For more information about the dial plan, refer to Understanding the Dial Plan on page 50.

Let us try an example. If you want to change the feature access code for Call Park to *72:

1. Type **`change feature-access-codes`** and press **Enter**.

   The **Feature Access Code** screen appears.
2. Move the cursor to the **Call Park Access Code** field.
3. Type ***72** in the access code field over the old code.
4. Press **Enter** to save your changes.

   If you try to enter a code that is assigned to a feature, the system warns you of the duplicate code and does not allow you to proceed until you change one of them.

Tip:

To remove any feature access code, merely delete the existing FAC and leave the field blank.

Controlling the Features Your Users Can Access

Avaya Communication Manager offers a wide range of features and functions. Some of these you can administer differently from one user to the next. For example, you can give one user a certain set of telephone buttons, and the next user a completely different set, depending on what each person needs to get his/her job done. You decide on these things as you administer the phones for these individuals. Refer to Telephone feature Buttons on page 123 for more information.

You can also establish classes of service (COS) to control the Communication Manager features that users can access. For example, you can permit users to forward their calls, or restrict them from placing priority calls. Once you have set permissions for a class of service, you assign this COS to a user's telephone or other device.

Classes of service are very similar to classes of restriction. COR and COS do not overlap in the access or restrictions they control. Refer to Class of Service on page 614 for more information.

Class of service and class of restriction give you great flexibility with what you allow users to do. If you are in doubt about the potential security risks associated with a particular permission, read further in this document, consult the *Avaya Products Security Handbook*, or contact your Avaya technical support representative.

System-Wide Settings

There are some settings that you enable or disable for the entire system, and these settings effect every user. You may want to look over the various System Parameters screens and decide which settings best meet the needs of your users.

To see a list of the different types of parameters that control your system, type **`display system-parameters`** and press **Help**. You can change some of these parameters yourself. Type **`change system-parameters`** and press **Help** to see which types of parameters you can change. In some cases, an Avaya technical support representative is the only person who can make changes, such as to the **System-Parameters Customer-Options** screen.

Type **`list usage`** to see all the instances of an object, such as an extension or ip address, in your system. This is useful when you attempt to change administration and receive an "in use" error. See *Maintenance Commands*, 03-300191, for more information.

This chapter offers a few examples of how you establish these system-wide settings. The Screen Reference contains explanations of each of the system parameters screens, and how to use them.

Changing Feature Parameters

You can modify the system parameters that are associated with some of the system features. For example, you can use the system parameters to allow music to play if callers are on hold or to allow trunk-to-trunk transfers on the system.

Generally, Avaya sets your system parameters when your system is installed. However, you can change these parameters as your organization's needs change. For example, let us say that you are told that the number of rings between each point for new coverage paths should change from 4 to 2 rings.

To change the number of rings:

1. Type **change system-parameters coverage/forwarding** and press **Enter**.

   The **System Parameters Call Coverage/Call Forwarding** screen appears.

Figure 15: System-Parameters Call Coverage/Call Forwarding screen

```
                                                          Page 1 of x
               SYSTEM PARAMETERS -- CALL COVERAGE / CALL FORWARDING

CALL COVERAGE/FORWARDING PARAMETERS
          Local Cvg Subsequent Redirection/CFWD No Ans Interval (rings): 2
        Off-Net Cvg Subsequent Redirection/CFWD No Ans Interval (rings): 2
                         Coverage - Caller Response Interval (seconds): 4


COVERAGE
                                        Keep Held SBA At Coverage Point? y
             External Coverage Treatment For Transferred Incoming Calls? n


FORWARDING
                                                 Call Forward Override? n
                                               Coverage After Forwarding? y
```

2. In the **Local Coverage Subsequent Redirection/CFWD No Answer Interval** field, type **2.**
3. Press **Enter** to save your changes.

   Each telephone in a Call Coverage path now rings twice before the call routes to the next coverage point. The **Local Cvg Subsequent Redirection/CFWD No Ans Interval** field also controls the number of rings before the call is forwarded when you use Call Forwarding for busy/don't answer calls. This applies only to calls covered or forwarded to local extensions. Use Off-Net to set the number of rings for calls forwarded to public network extensions.

Setting WAN Bandwidth Limits Between Network Regions

Using the Communication Manager Call Admission Control: Bandwidth Limitation (CAC-BL) feature, you can specify a VOIP bandwidth limit between any pair of IP network regions, and then deny calls that need to be carried over the WAN link that exceed that bandwidth limit. Bandwidth limits may be administered in terms of:

- Kbit/sec WAN facilities
- Mbit/sec WAN facilities
- Explicit number of connections
- No limit

It is highly recommended that you have the following design information before setting the bandwidth limits and mapping the connections:

1. Network topology and WAN link infrastructure.
2. An understanding of the Committed Information Rate (CIR) for the WAN infrastructure.
3. Overlay/design of the Network Regions mapped to the existing topology.
4. Codec sets administered in the system.
5. Bandwidth is assumed to be full duplex.

The following table can be used to help assess how much bandwidth (in Kbits /sec) is used for various types of codeecs and packet sizes. The values shown assume a 7 byte L2 WAN header (and are rounded up).

Table 1: Bandwidth usage (in Kbits/sec) based on packet size and codec selection

Packet Size	10 ms	20 ms	30 ms	40 ms	50 ms	60 ms
G.711	102	83	77	74	72	71
G.729	46	27	21	18	16	15
G.723-6.3	NA	NA	19	NA	NA	13
G.723-5.3	NA	NA	18	NA	NA	12

These values, when compared to the actual bandwidth used for 8 byte as well as 10 byte L2 WAN headers are not significantly different. In some cases, the rounded up values shown above are greater than values used for 10 bytes.

The bandwidth usage numbers shown above assume 6 bytes for Multilink Point-to-Point Protocol (MP) or Frame Relay Forum (FRF), 12 Layer 2 (L2) header, and 1 byte for the end-of-frame flag on MP and Frame Relay frames for a total of 7 byte headers only. They do not account for silence suppression or header compression techniques, which might reduce the actual bandwidth. For other types of networks (such as Ethernet or ATM) or for cases where there is a lot of silence suppression or header compression being used, the network may be better modeled by administering the CAC-BL limits in terms of number of connections rather than bandwidth used.

Instructions

Note:

All DIRECT links must be administered first, and INDIRECT links administered last.

To set bandwidth limitations between directly-connected network regions:

1. Type **`change ip-network region <n>`**, where ***n*** is the region number you want to administer, and press **Enter**.
2. The **IP Network Region** screen appears.
3. Scroll to page 3 of the form, **Inter Network Region Connection Management**.
4. In the **codec-set** field, enter the number (1-7) of the codec set to be used between the two regions.
5. In the **Direct WAN** field, enter **y**.
6. In the **WAN-BW-limits** field, enter the number and unit of measure (Calls, Kbits, Mbits, No Limit) that you want to use for bandwidth limitation.
7. Press **Enter** to save your changes.

To set bandwidth limitations between indirectly-connected network regions:

1. Type **`change ip-network region <n>`**, where ***n*** is the region number you want to administer, and press **Enter**.

   The **IP Network Region** screen appears.
2. Scroll to page 3 of the form, **Inter Network Region Connection Management**.
3. In the **codec-set** field, enter the number (1-7) of the codec set to be used between the two regions.
4. In the **Direct WAN** field, enter **n**.
5. In the **Intervening-regions** fields, enter up to four intervening region numbers between the two indirectly connected regions.

6. In the **Dynamic CAC Gateway** field, set the gateway that reports the bandwidth limit for this link. The gateway must be configured to be a CAC gateway.
7. Press **Enter** to save your changes.

Note:

Type **`display ip-network region <n>`** to view the current settings of inter-network region connections. Type **`status ip-network region <n>`** to view the current status (i.e., bandwidth and number of connections being used) of network-region connections.

Administering Treatment for Denied or Invalid Calls

You can administer your system to reroute denied or invalid calls to an announcement, the attendant, or to another extension.

Instructions

In this example, we want:

- all outward restricted call attempts to route to an announcement at extension 2040
- all incoming calls that are denied to route to the attendant
- all invalid dialed numbers to route to an announcement at extension 2045

1. Type **`change system-parameters features`** and press **Enter**.

   The **Feature-Related System Parameters** screen appears.
2. In the **Controlled Outward Restriction Intercept Treatment** field, type **announcement**.
3. Another blank field appears.
4. In this blank field, type **2040**.
5. This is the extension of an announcement you recorded earlier.
6. In the **DID/Tie/ISDN Intercept Treatment** field, type **attd**.
7. This allows the attendant to handle incoming calls that have been denied.
8. In the **Invalid Number Dialed Intercept** field, type **announcement**.
9. Another blank field appears.
10. In this blank field, type **2045**.
11. This is the extension of an announcement you recorded earlier.
12. Press **Enter** to save your changes.

Setting up Music-on-Hold

Music-on-Hold automatically provides music to a caller placed on hold. Providing music lets the caller know that the connection is still active. The system does not provide music to callers in a multiple-party connection who are in queue, on hold, or parked.

Before you start

You need to determine the music source you will use, and obtain the necessary circuit pack. Refer to the *Hardware Guide for Avaya Communication Manager* for more information about required hardware.

Note:

If you use equipment that rebroadcasts music or other copyrighted materials, you may be required to obtain a copyright license from or pay fees to a third party.

Instructions

In this example, we administer the system to allow local callers and incoming trunk callers to hear music while on hold. Note that if you use Tenant Partitioning, you cannot set up Music on Hold this way. See Providing Service for Multiple Tenants on page 63 for more information.

To set up Music-on-Hold:

1. Type **`change system-parameters features`** and press **Enter**.

   The **Feature-Related System Parameters** screen appears.

Figure 16: Feature-Related System Parameters screen

```
change system-parameters features                              Page 1 of x
                        FEATURE-RELATED SYSTEM PARAMETERS

                                  Trunk-to-Trunk Transfer? none
 Automatic Callback - No Answer Timeout Interval (rings): 4_
                    Call Park Timeout Interval (minutes): 10
    Off-Premises Tone Detect Timeout Interval (seconds): 20_
                               AAR/ARS Dial Tone Required? y
                                     Music/Tone On Hold: music  Port: _______
          Music (or Silence) On Transferred Trunk Calls: all
                       DID/Tie/ISDN Intercept Treatment: attd
              Messaging Service Adjunct (MSA) Connected? y
 Internal Auto-Answer of Attd-Extended/Transferred Calls: y
              Automatic Circuit Assurance (ACA) Enabled? y
                                       ACA Referral Calls: local
                                ACA Referral Destination: ________
           ACA Short Holding Time Originating Extension: ______
            ACA Long Holding Time Originating Extension: ______
         Abbreviated Dial Programming by Assigned Lists:
    Auto Abbreviated/Delayed Transition Interval(rings):
```

2. In the **Music/Tone On Hold** field, type **music**.

   The **Port** field appears.

3. In the **Port** field, type **6040**.

   This is the port address of the music source.

4. In the **Music (or Silence) on Transferred Trunk Calls**, type **all**.

5. Press **Enter** to save your changes.

6. Now administer a class of restriction with **Hear System Music on Hold** set to **y**, to allow your local users to hear music on hold.

More information

If a call with either Data Privacy or Data Restriction activated is placed on hold, the Music/Tone on Hold is withheld. This is to prevent transmission of a musical tone that a connected data service might falsely interpret as a data transmission.

If you administer the **Music/Tone on Hold** field to provide music, the system provides the music after a hunt group or Automatic Call Distribution (ACD) split delayed announcement.

Music on Hold may sound distorted when played to IP trunks or to IP phones through certain codecs, particularly the G.723 codec. You can provide different on-hold materials for these endpoints. Using the instructions for Providing Service for Multiple Tenants, create one tenant partition for all endpoints that do not use the G.723 codec and administer Music on Hold for this tenant. Create another tenant partition for endpoints that use the G.723 codec and administer silence, news, or other material that does not sound distorted for these endpoints.

Providing Service for Multiple Tenants

If you manage the switching system for an entire office building, you may need to provide individualized telephone service for each of the firms who are tenants. You can set up your system so that each tenant can have its own attendant, and can chose to have music or play special announcements while callers are on hold.

Before you start

Before you can administer tenants in your system, Tenant Partitioning must be enabled on the **System-Parameters Customer-Options** screen. To allow tenants to select their own Music on Hold, you must purchase and administer equipment. Refer to the **Hardware Guide for Avaya Communication Manager** for more information about required hardware. Your Avaya technical support representative can help you administer this hardware.

Instructions

In this example, we are going to administer the system to allow one tenant to play Country music for callers on hold, and another to play Classical music. We will assign these music types to two new tenants.

1. Type **`change music-sources`** and press **Enter**.

   The **Music Sources** screen appears.

Figure 17: Music Sources screen

```
                          Music Sources                        Page 1 of X

              Source    Type      Port       Description

                1      music     01A1001    Country
                2      tone                 Tone-on-Hold
                3      music     01A1003    Classical
                4      none
                5      none
                6      none
                7      music     12B1301    Oldies
                8      none
                9      none
               10      none
               11      music     04C2003    Rock
               12      none
               13      none
               14      none
               15      none
```

2. For **Source 1**, enter **music** for the **Type**, **01A1001** for the **Port**, and **Country** for the **Description**.
3. Move to **Source 3**, and enter **music** for the Type, **01A1003** for the **Port**, and **Classical** for the **Description**.
4. Press **Enter** to save your changes.
5. Type **change tenant 1** and press **Enter**.

   The **Tenant** screen appears.

Figure 18: Tenant screen

```
Tenant 18

              Tenant Description: ____________________________________________

                 Attendant Group: 1

           Ext Alert Port (TAAS): _______ Ext Alert (TAAS) Extension: ____

               Night Destination: _____

                    Music Source: 1
```

6. In the **Tenant Description** field, type **Dentist**.

   This identifies the client in this partition.

7. In the **Attendant Group** field, type **1**.

   The attendant group number must also appear in the **Group** field of the attendant console screen for this tenant.

8. In the **Music Source** field, type **1**.

   Callers to this tenant will now hear country music while on hold.

9. Press **Enter** to save your changes.
10. To administer the next partition, type **change tenant 2** and press **Enter**.
11. Administer this tenant, Insurance Agent, to use Attendant Group 2 and Music Source 3. Be sure to change the **Attendant Console** screen so that this attendant is in group 2.

    This tenant's callers will hear classical music on hold.

Receiving Notification in an Emergency

If one of your users calls an emergency service such as the police or ambulance, someone, perhaps the receptionist, security or the front desk, needs to know who made the call. Thus, when the emergency personnel arrive, they can be directed to the right place.

You can set up Communication Manager to alert the attendant and up to ten other extensions whenever an end-user dials an emergency number. The display on the notified user's telephone shows the name and number of the person who placed the emergency call. The phones also ring with a siren-type alarm, which users must acknowledge to cancel.

Before you start

Decide if you want one user to be able to acknowledge an alert, or if all users must respond before an alert is cancelled.

Verify that the **ARS** field is **y** on the **System-Parameters Customer-Options** screen.

Make sure that the extensions you notify belong to physical digital display phones. Refer to Telephone Reference on page 647 for a list of telephone types. When you assign crisis alert buttons to the phones, check the **Type** field on the **Station** screen to be sure you are not using a virtual extension.

Instructions

In this example, we will set up the system to notify the attendant and the security guards at all 3 entrances when someone dials the emergency number 5555. All three guards must acknowledge the alert before it is silent.

1. Type **change ars analysis 5** and press **Enter**.

   The **AAR and ARS Digit Analysis Table** screen appears.

Figure 19: AAR and ARS Digit Analysis Table screen

```
ARS DIGIT ANALYSIS TABLE
                        Location: all          Percent Full:  6

     Dialed         Total   Route     Call   Node  ANI
     String       Min Max  Pattern   Type   Num   Reqd
5555_____________  4_  4_    1____    alrt   ___    n
_________________  __  __   _____    ____   ___    n
_________________  __  __   _____    ____   ___    n
_________________  __  __   _____    ____   ___    n
_________________  __  __   _____    ____   ___    n
_________________  __  __   _____    ____   ___    n
_________________  __  __   _____    ____   ___    n
_________________  __  __   _____    ____   ___    n
_________________  __  __   _____    ____   ___    n
_________________  __  __   _____    ____   ___    n
_________________  __  __   _____    ____   ___    n
_________________  __  __   _____    ____   ___    n
_________________  __  __   _____    ____   ___    n
_________________  __  __   _____    ____   ___    n
_________________  __  __   _____    ____   ___    n
_________________  __  __   _____    ____   ___    n
_________________  __  __   _____    ____   ___    n
```

2. In the **Dialed String** field, type **5555**.

   This is the number that end-users dial to reach emergency services.

3. In the **Total Min** and **Max** fields, type **4**.

   In this example, the user must dial all 4 digits for the call to be treated as an emergency call.

4. In the **Route Pattern** field, type **1**.

   In this example, we use route pattern 1 for local calls.

5. In the **Call Type** field, type **alrt**.

   This identifies the dialed string 5555 as one that activates emergency notification.

6. Press **Enter** to save your changes.

Now set up the attendant console to receive emergency notification.

7. Type **change attendant 1** and press **Enter**.

   The **Attendant Console** screen appears.

8. In the feature button area, assign a **crss-alert** button.
9. Press **Enter** to save your changes.
10. Assign a **crss-alert** button to each security guard's telephone.

    You cannot assign this button to a soft key. See Adding Feature Buttons on page 121 for more information.

Finally, we make sure that all security personnel and the attendant will have to acknowledge the alert.

11. Type **change system-parameters crisis-alert** and press **Enter**.

    The **Crisis Alert System Parameters** screen appears.

12. Go to the **Every User Responds** field and type **y**.
13. Press **Enter** to save your changes.

More information

Attendants cancel an alert by pressing the crisis alert button three times. The first button push turns off the siren, the second stops the lamp from flashing, and the third clears the display.

Digital telephone users cancel the siren by pushing the crisis alert button. If you have set the system so that only one user needs to respond, this stops the alerting at all phones. If all users must respond, each telephone continues to alert until that user presses the crisis alert button to acknowledge the alarm. The emergency caller's name and extension remain on the display at this point. To completely cancel an alert and clear their displays, users press the Normal button.

Once you administer Crisis Alert, Communication Manager still records each emergency call and sends a record to the journal printer, if available. If not, you can view the emergency log with the command **list emergency**.

Related topics

To determine what types of digital phones have displays, see Telephone Reference on page 647.

For information on setting up Centralized Automatic Message Accounting (CAMA) trunks and providing Caller's Emergency Service Identification (CESID) information to your local community's Enhanced 911 system, see CAMA Numbering Format.

For information about updating station location information without having to change the USA 911 system's Automatic Location Identification database, see the **Emergency Location extension** field in Station on page 1409.

For information on how to administer IP phones to make emergency calls, see Setting up emergency calls on IP phones on page 106.

For more information on individual features, see *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205.

Notifying a Digital Pager of an Emergency

You also have the option of having your emergency calls go to a digital pager. When someone dials an emergency number (for example, 911), the system sends the extension and location (that originated the emergency call) to the administered pager.

Before you start

- You need to administer a **crss-alert** button on at least one of the following.
  - Attendant Console (use the **`change attendant`** command)
  - Digital telephone set (use the **`change station`** command)
- The **ARS Digit Analysis Table** must have emergency numbers in the **Call Type** column set to **alrt** (crisis alert).
- You need a digital numeric pager.

Instructions

To set up crisis alert to a digital pager:

1. Type **`change system-parameters crisis-alert`** and press **Enter**.

   The **Crisis Alert System Parameters** screen appears.

2. In the **Alert Pager** field, type **y**.

   This allows you to use the Crisis Alert to a Digital Pager feature and causes additional crisis alert administration fields to appear.

3. In the **Originating Extension** field, type a valid unused extension to send the crisis alert message.

   We'll type **7768**.

4. In the **Crisis Alert Code** field, type **911**.

   This is the number used to call the crisis alert pager.

5. In the **Retries** field, type **5**.

   This is the number of additional times the system tries to send out the alert message in case of an unsuccessful attempt.

6. In the **Retry Interval (sec)** field, type **30**.

   This is length of time between retries.

7. In the **Main Number** field, type the number that is to be displayed at the end of the pager message.

   We'll type **303-555-0800**.

8. In the **Pager Number** field, type the number for the pager.

   We'll type **303-555-9001**.

9. In the **Pin Number** field, type **pp77614567890**.

   This is the PIN number, if required, for the pager. Insert any pause digits (pp) as needed to wait for announcements from the pager service to complete before sending the PIN.

10. In the **DTMF Duration - Tone (msec)** field, type **100**.

    This is the length of time the DTMF tone is heard for each digit.

11. In the **Pause (msec)** field, type **100**.

    This is the length of time between DTMF tones for each digit.

12. Press **Enter** to save your changes.

Related topics

Refer to the Crisis Alert feature in *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205 for more detailed information.

Other Useful Settings

There are many settings that control how your system operates and how your users phones work. Most of these you administer through one of the System Parameters screens. This section describes a few of the items you can enable in your system to help your users work more efficiently. See Feature-Related System Parameters for a more detailed description of the available system settings.

Automatic callback if an extension is busy

You can allow users to request that the system call them back if they call a user whose telephone is busy. For more information, see "Automatic Callback" in *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205.

Automatic hold

You can set a system-wide parameter that allows your users to initiate a call on a second line without putting the first call on Hold. This is called Automatic Hold, and you enable it on the **Feature-Related System Parameters** screen. If you do not turn this on, the active call drops when a the user presses the second line button.

Bridging onto a call that has gone to coverage

You can allow users to join (bridge) on to a call that rang at their extension and then went to coverage before they could answer. For more information, see "Temporary Bridged Appearance" in *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205.

Distinctive ringing

You can establish different ringing patterns for different types of calls. For example, you can administer your system so that internal calls ring differently from external calls or priority calls. For more information, see "Distinctive Ringing" in *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205.

Warning when phones are off-hook

You can administer the system so that if a telephone remains off-hook for a given length of time, Communication Manager sends out a warning. This is particularly useful in hospitals, where the telephone being off-hook may be an indication of trouble with a patient. See **Class of Service** for more information.

Warning users if their calls are redirected

You can warn analog telephone users if they have features active that may redirect calls. For example, if the user has activated send all calls or call forwarding, you can administer the system to play a special dial tone when the user goes off-hook. See "Distinctive Ringing" in *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205 for more information.

Controlling the Calls Your Users Can Make and Receive

The Avaya Communication Manager provides several ways for you to restrict the types of calls your users can make, and the features that they can access.

You use Class of Restriction (COR) to define the types of calls your users can place and receive. Your system may have only a single COR, a COR with no restrictions, or as many CORs as necessary to effect the desired restrictions.

You will see the **COR** field in many different places throughout Communication Manager when administering phones, trunks, agent logins, and data modules, to name a few. You must enter a COR on these screens, although you control the level of restriction the COR provides.

Strategies for assigning CORs

The best strategy is to make it as simple as possible for you and your staff to know which COR to assign when administering your system. You can create a unique COR for each type of user or facility, for example, call center agents, account executives, administrative assistants, Wide Area Telecommunications Service (WATS) trunks, paging zones or data modules.

You can also create a unique COR for each type of restriction, for example, toll restriction, or outward restriction. If you have a number of people who help you administer your system, using this method would also require the additional step of explaining where you wanted to use each type of restriction. See "Class of Restriction" in *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205 for more information.

To find out what CORs are administered in your system already, type **list cor**. You can also display information for a single COR by typing **list cor #**.

Allowing users to change CORs

You can allow specific users to change their Class of Restriction (COR) from their telephones using a Change COR feature access code. You can also limit this feature by insisting that the user enter a password as well as a feature access code before they can change their COR. The Station Lock feature also allows users to change their own COR. For more information on the Station Lock feature, see Station Lock on page 74.

Before you start

- Be sure that **Change COR by FAC** field is set to **y** on the **System-Parameters Customer-Options** screen. Note that you cannot have both **Change COR by FAC** and **Tenant Partitioning** enabled.
- Be sure that each user (who you want to allow to change a COR) has a class of service with console permissions. For more information about console permissions, refer to Class of Service on page 830.

Instructions

To allow users to change their own class of restriction, you must define a feature access code and can, optionally, create a password. For example, we will create a change COR feature access code of *55 and a password of 12344321.

1. Type **change feature-access-codes** and press **Enter**.

   The **Feature Access Code** screen appears.

2. Move the cursor to the **Change COR Access Code** field.
3. Type ***55** in the access code field.
4. Press **Enter** to save your changes.

   Now we have to define the password.

5. Type **change system-parameters features** and press **Enter**.

   The **Feature-Related System Parameters** screen appears. Press **Next Page** to find the **Automatic Exclusion Parameters** section.

Figure 20: Feature-Related System Parameters screen

```
change system-parameters features                                  page 12 of X
                         FEATURE-RELATED SYSTEM PARAMETERS

AUTOMATIC EXCLUSION PARAMETERS

                       Automatic Exclusion by COS? y
                  Automatic Exclusion Coverage/Hold? y
              Automatic Exclusion with Whisper Page? y
                            Recall Rotary Digit: 2
                     Password to Change COR by FAC: *
          Duration of Call Timer Display (seconds): 3
WIRELESS PARAMETERS
  Radio Controllers with Download Server Permission (enter board location)
    1.         2.         3.         4.         5.

IP PARAMETERS
                    Direct IP-IP Audio Connections? n
                              IP Audio Hairpinning? n
RUSSIAN MULTI-FREQUENCY PACKET SIGNALING
                                                Re-try?
     T2 (Backward Signal) Activation Timer (secs):
```

6. Move to the **Password to Change COR by FAC** field and enter **12344321**.

   This field determines whether or not Communication Manager requires the user to enter a password when they try to change their COR. Avaya recommends that you require a password.

7. Press **Enter** to save your changes.

Station Lock

Station Lock provides users with the capability to manually lock their stations, using a button or feature access code, in order to prevent unauthorized external calls from being placed.

Station Lock can prevent unauthorized external calls. Phones can be remotely locked and unlocked. Station Lock allows users to:

- change their Class of Restriction (COR); usually the lock COR is set to fewer calling permissions than the station's usual COR
- lock their phones to prevent unauthorized outgoing calls.
- block outgoing calls and still receive incoming calls.
- block all outgoing calls except for emergency calls.

Station Lock is activated by pressing a telephone button, which lights the button indicator, or dialing a FAC.

Analog and XMOBILE stations must dial a FAC to activate the feature. The user hears a special dial tone on subsequent origination attempts from the telephone to indicate that the lock feature is active.

Digital stations (including DCP, BRI, IP hardphones and softphones) access Station Lock with a feature button or via a FAC. If a digital station has a Station Lock button but activates the feature with the FAC, the LED for the button lights and no special dial tone is provided. However, if a digital station does not have a Station Lock button and activates the feature with the FAC, a special dial tone is provided.

A station can be locked or unlocked from any other station if the FAC is used and the Station Security Code is known. The attendant console can never be locked but can be used to lock or unlock other stations. A station also can be locked or unlocked via a remote access trunk.

Interactions

- Attendant Console

  Station Lock cannot be used for attendant consoles but it can be assigned to regular digital stations that might also have console permissions. The FAC cannot be used to activate Station Lock for the attendant console, but the FAC can be dialed from the attendant console in an attempt to remotely activate or deactivate Station Lock for another station.

- Personal Station Access (PSA)

  Station Lock can be used for PSA stations as long as they are associated with an extension. When stations are disassociated, Station Lock cannot be activated.

- Remote Access

  After a remote user dials a valid barrier code, the user receives system dial tone. To activate/deactivate Station Lock, the user must dial the FAC, then the extension number, then the security code number.

Fiber Link Administration

For information concerning system fiber link configurations, see *Upgrades and Additions for Avaya DEFINITY Server R*.

Chapter 3: Managing Phones

Installing New Phones

Simple administration allows you to plug a telephone into a jack and dial a sequence to start up service to the telephone. The dialing sequence sets up an association between the telephone and the corresponding station administration.

SECURITY ALERT:
If you do not manage this feature carefully, its unauthorized use may cause you security problems. Consult the Avaya Products Security Handbook for suggestions on how to secure your system and find out about obtaining additional security information.

For traditional instructions, see Installing New Phones.

Before you start

On the **Feature-Related System Parameters** screen, be sure the **Customer Telephone Activation (CTA) Enabled** field is **y** and the **TTI Enabled** field is **y**.

Complete the **Station** screen for the new telephone and type x in the port field. Note that the telephone type must match the board type. For example, match a two-wire digital telephone with a port on a two-wire digital circuit pack. Use this procedure with all circuit-switched phones except BRI (ISDN) and model 7103A.

Tip:

See Completing the station screens for more information. See Using Templates to Add Phones if you want to add a number of phones with similar settings.

Instructions

CAUTION:

You can destroy your hardware if you attempt to connect an analog telephone to a digital port.

To associate a telephone with existing x-port station administration, complete the following steps from the telephone you want to install:

1. Plug the telephone into the wall jack.
2. Lift the receiver and continue if you hear dial tone.
3. Dial ***#*nnnn***, where ***nnnn*** is the extension number of the telephone you are installing.
4. Hang up after you receive confirmation tone.
5. Dial a test call to confirm that the telephone is in service.

   If possible, call a telephone with a display so the person answering can confirm that you entered the correct extension number.
6. Repeat the process until all new phones have been installed.
7. For security reasons, you should disable this feature when you are done. At the system administration terminal type **`change system-parameters features`** to access the **Feature-Related System Parameters** screen.
8. Type **n** in the **Customer Telephone Activation (CTA) Enabled** field.
9. Press **Enter** to save your changes.
10. Type **`save translations`** and press **Enter** to permanently save the changes.

    a. Fixing problems

    If you misdial and the wrong extension is activated for the telephone you are using, use the terminal translation initialization (TTI) unmerge feature access code to "uninstall" the telephone before you try again.

Adding New Phones

When you are asked to add a new telephone to the system, what do you do first? To connect a new telephone you need to do three things:

- find an available port
- wire the port to the cross-connect field or termination closet
- tell the telephone system what you’re doing

Before you can determine which port to use for the new telephone, you need to determine what type of telephone you are installing, what ports are available, and where you want to install the telephone.

To add an IP telephone, see Adding an IP Telephone on page 103.

Gathering necessary information

1. Determine whether the telephone is an analog, digital, ISDN, or hybrid set. You may also administer a virtual telephone, one without hardware at the time of administration.

   You need this information to determine the type of port you need, because the port type and telephone type must match. If you do not know what type of telephone you have, see the **Type** field on the **Station** screen for a list of phones by model number.

2. Record the room location, jack number, and wire number.

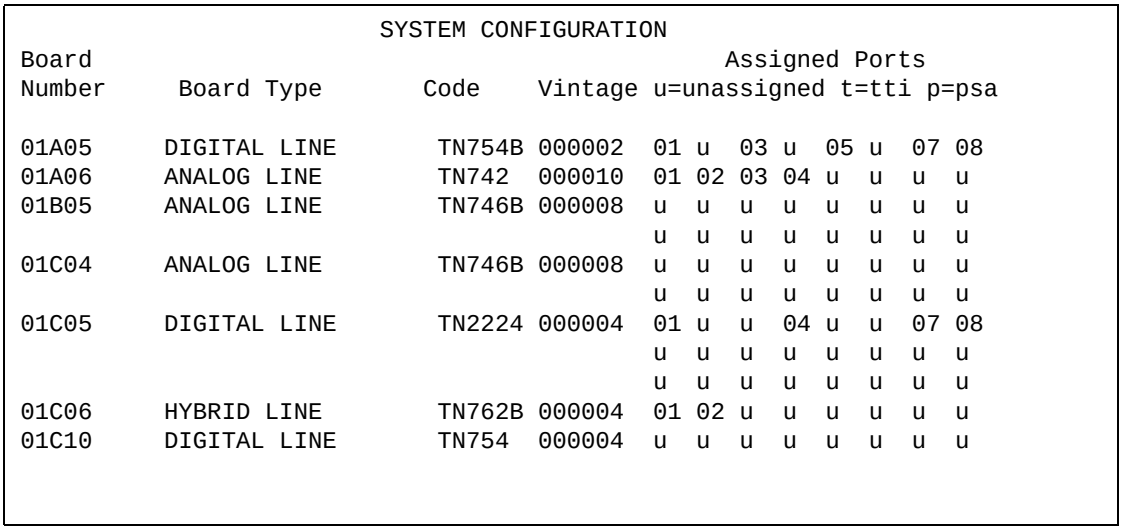
   You may find this information on the jack where you want to install the telephone, recorded in your system records, or from the technician responsible for the physical installation.

3. Display the available boards (cards) and ports.

   To view a list of boards on your system, type **`list configuration station`** and press **Enter**.

   The **System Configuration** screen appears.

Figure 21: System Configuration screen



```
                        SYSTEM CONFIGURATION
Board                                          Assigned Ports
Number     Board Type       Code    Vintage u=unassigned t=tti p=psa

01A05     DIGITAL LINE       TN754B 000002  01 u  03 u  05 u  07 08
01A06     ANALOG LINE        TN742  000010  01 02 03 04 u  u  u  u
01B05     ANALOG LINE        TN746B 000008  u  u  u  u  u  u  u  u
                                            u  u  u  u  u  u  u  u
01C04     ANALOG LINE        TN746B 000008  u  u  u  u  u  u  u  u
                                            u  u  u  u  u  u  u  u
01C05     DIGITAL LINE       TN2224 000004  01 u  u  04 u  u  07 08
                                            u  u  u  u  u  u  u  u
                                            u  u  u  u  u  u  u  u
01C06     HYBRID LINE        TN762B 000004  01 02 u  u  u  u  u  u
01C10     DIGITAL LINE       TN754  000004  u  u  u  u  u  u  u  u
```

The **System Configuration** screen shows all the boards on your system that are available for connecting phones. You can see the board number, board type, circuit-pack type, and status of each board's ports.

4. Choose an available port and record its port address.

   Each port that is available or unassigned is indicated by a ‘u.’ Choose an available port from a board type that matches your telephone type (such as a port on an analog board for an analog telephone).

   Every telephone must have a valid port assignment, also called a port address. The combined board number and port number is the port address. So, if you want to attach a telephone to the 3rd port on the 01C05 board, the port address is 01C0503 (01=cabinet, C=carrier, 05=slot, 03=port).

   Tip:

   If you add several phones at one time, you may want to print a paper copy of the **System Configuration** screen. To print the screen to a printer attached to the system terminal, type **`list configuration station print`** and press **Enter**. To print to the system printer that you use for scheduled reports, type **`list configuration station schedule immediate`** and press **Enter**.

5. Choose an extension number for the new telephone.

   The extension you choose must not be assigned and must conform to your dial plan. You should also determine whether this user needs an extension that can be directly dialed (DID) or reached via a central telephone number.

   Be sure to note your port and extension selections on your system’s paper records.

Physically connecting the telephone

Once you have collected all the information, you are ready to physically wire the port to the cross-connect field.

If you have an Avaya technical support representative or on-site technician who completes the physical connections, you need to notify them that you are ready to add the telephone to the system. To request that Avaya install the new connections, call your Avaya technical support representative to place an order.

If you are responsible for making the connections yourself and if you have any questions about connecting the port to the cross-connect field, see your system installation guide.

Now you are ready to configure the system so that it recognizes the new telephone.

Before you start

To download language display labels for phones, set the **Display Language** field on the **Station** screen to **english**, **spanish**, **italian**, **french**, **user-defined**, or **unicode**.

Note:

Unicode display is only available for Unicode-supported telephones. Currently, only the 4620SW and 4610SW support Unicode display.

For a Eurofont character display for the 2420/2410 telephone, set the **Display Character Set** field on the **System-Parameters Country-Options** screen to **Eurofont**.

For a Katakana character display for the 2420/2410 telephone, set the **Display Character Set** field on the **System-Parameters Country-Options** screen to **Katakana**.

Completing the station screens

The information that you enter on the station screen advises the system that the telephone exists and indicates which features you want to enable on the telephone.

To access the **Station** screen for the new telephone:

1. Type **`add station nnnn`**, where ***nnnn*** is the extension for the new telephone. Press **Enter**.

   Make sure the extension conforms to your dial plan. You can also use the **`add station next`** command to add a telephone to the **`next`** available extension.

   Note:

   If you have Terminal Translation Initialization (TTI) enabled, you may receive the following error message when attempting to add a new station: "**No station/TTI port records available; 'display capacity' for their usage.**" If this occurs, try one or more of the following:

   - Remove any DCP or Analog circuit packs that have no ports administered on them.
   - If you are not using TTI or any related feature (such as PSA or ACTR), set the **Terminal Translation Initialization (TTI) Enabled?** field on the **Feature Related System Parameters** screen to **n**.
   - Contact your Avaya technical support representative.

   For more information on TTI, see "Terminal Translation Initialization" in *Feature Description and Implementation for Avaya Communication Manager*, 555-245-205.

   For more information on the **System Capacity** screen, see *Maintenance Commands for Avaya Communication Manager, Media Gateways and Servers*, 03-300191.

   When the **Station** screen appears, you see the extension number and some default field values. For example, the following screen is for a new telephone at extension 2345.

Figure 22: Station screen

```
                              STATION

Extension: 2345___                Lock Messages? n          BCC: _
     Type: 6480D+                 Security Code: _________   TN: 1_
     Port: _______________      Coverage Path 1: ____       COR: 1
     Name: ____________________ Coverage Path 2: ____       COS: 1
                                Hunt-to Station: ________

STATION OPTIONS
          Loss Group: none      Personalized Ringing Pattern: 1
         Data Module? n                   Message Lamp Ext: 2345
        Speakerphone: 2-way             Mute Button Enabled? y
    Display Language: english
 Survivable GK Node Name:                        edia Complex Ext:________
                                                  IP SoftPhone? n
                                           Remote Office Phone? y
```

2. Type the model number of the telephone into the **Type** field.

   For example, to install a 6508D+ telephone, type **6480D+** in the **Type** field. Note that the displayed fields may change depending on the model you add.

3. Type the port address in the **Port** field.

   Note:

   Port 1720 is turned off by default to minimize denial of service situations. This applies to all IP softphones release 5.2 or later. You can change this setting, if you have root privileges on the system, by typing the command: **`/opt/ecs/sbin ACL 1720 on`** or **`off`**.

4. Type a name to associate with this telephone in the **Name** field. The name you enter displays on called phones that have display capabilities. Also, some messaging applications, such as INTUITY, recommend that you enter the user's name (last name first) and their extension to identify the telephone.

   Note:

   For 4620SW or 4610SW telephones, the **Name** field is supported by Unicode language display. You must be using ASA or MSA. For more information on Unicode language display, see Administering Unicode display on page 181.

5. Press **Enter** to save your changes.

To make changes to this new telephone, such as assigning coverage paths or feature buttons, type **`change station`** ***nnnn***, where ***nnnn*** is the extension of the new telephone. Press **Enter**. See Adding Feature Buttons on page 121 for more information.

Using Templates to Add Phones

A quick way to add phones is to copy the information from an existing telephone and modify it for each new telephone. For example, you can configure one telephone as a template for an entire work group. Then, you merely duplicate the template station screen to add all the other extensions in the group.

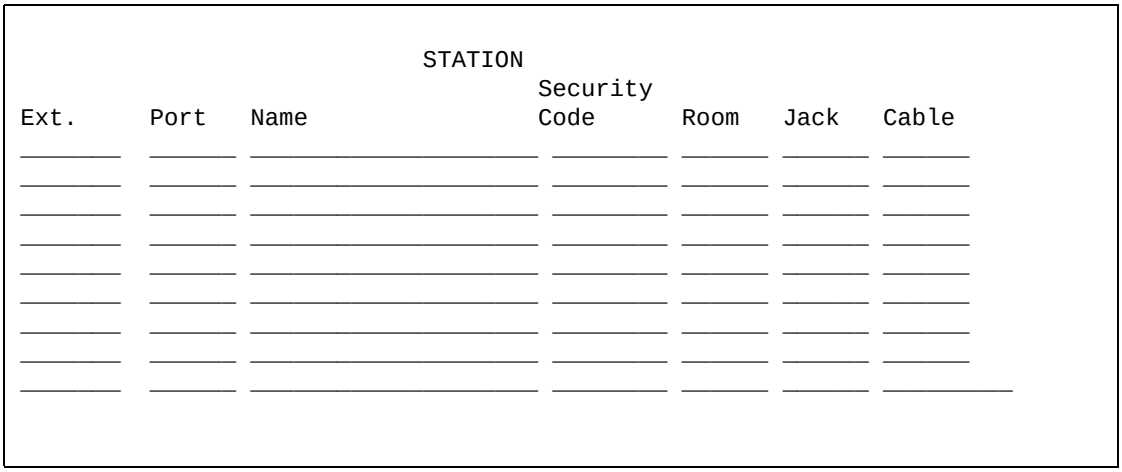
Note that only phones of the same model can be duplicated. The duplicate command copies all the feature settings from the template telephone to the new telephones.

To duplicate an existing telephone:

1. Type **`display station`** ***nnnn***, where ***nnnn*** is the extension of the station screen you want to duplicate to use as a template. Press **Enter**. Verify that this extension is the one you want to duplicate.
2. Press **Cancel** to return to the command prompt.
3. Type **`duplicate station`** ***nnnn*****,** where ***nnnn*** is the extension you want to duplicate. Press **Enter**.

   The system displays a blank duplicate **Station** screen.

Figure 23: Station screen



```
                              STATION
                                      Security
Ext.      Port    Name                Code      Room    Jack    Cable
_______   ______  ____________________ ________ ______ ______ ______
_______   ______  ____________________ ________ ______ ______ ______
_______   ______  ____________________ ________ ______ ______ ______
_______   ______  ____________________ ________ ______ ______ ______
_______   ______  ____________________ ________ ______ ______ ______
_______   ______  ____________________ ________ ______ ______ ______
_______   ______  ____________________ ________ ______ ______ ______
_______   ______  ____________________ ________ ______ ______ ______
_______   ______  ____________________ ________ ______ ______ _________
```

4. Type in the extension, port address, and telephone name for each new telephone you want to add.

   The rest of the fields are optional. You can complete them at any time.
5. Press **Enter** to save your changes to system memory.

To make changes to these phones, such as assigning coverage paths or feature buttons, type **`change station`** ***nnnn***, where ***nnnn*** is the extension of the telephone that you want to modify. Press **Enter**

Related topics

You can also add multiple call center agents, all with the same settings based on an agent that is already administered. Enter **command duplicate agent-loginID** and the extension of the agent you want to duplicate, then **start** and the extension you want to use for the first new agent, then **count** and the number of agents you want to add. Fill in the information on the **Agent LoginID** screen. For more information, see the *Avaya Communication Manager Contact Center Guide to ACD Contact Centers*.

Using an Alias

Not every telephone model or device has a unique station screen in the system. You might have to use an available model as an "alias" for another. If you need to enter a telephone type that the system does not recognize or support, use an alias. Defining aliases is also a useful method to identify items that act as analog stations on Communication Manager, such as fax machines, modems, or other analog device.

If you purchase a telephone model that is newer than your system, you can alias this telephone to an available model type that best matches the features of your new telephone. See your telephone's manual to determine which alias to use. If your manual does not have this information, you can contact the DEFINITY helpline for an appropriate alias.

For example, we will create two aliases: one to add a new 6220 telephone and one to add modems to our system.

1. See your new telephone's manual to find the correct alias.

   In our example, we find that the 6220 should be administered on an older system as a 2500 telephone.

2. Type **change alias station** and press **Enter**.

   The **Alias Station** screen appears.

Figure 24: Alias Station screen

```
                ALIAS STATION
   Alias Set Type            Supported Set Type
      6220                       2500
      modem                      2500
      _____                      _____
      _____                      _____
      _____                      _____
      _____                      _____
      _____                      _____
      _____                      _____
      _____                      _____
      _____                      _____
      _____                      _____
'#' indicates previously aliased set type is now native
```

3. Enter **6220** in the **Alias Set Type** field.

   This is the name or model of the unsupported telephone.

4. Enter **2500** in the **Supported Set Type** field.

   Enter the supported model in this field.

5. Enter **modem** in the second **Alias Set Type** field.

   You can call the alias set anything you like. Once you define the alias, you can use the alias set in the **Type** field on the **Station** screen.

6. Enter **2500** in the second **Supported Set Type** field.

   Entering 2500 indicates to the system that these models are basic analog devices.

7. Press **Enter** to save your changes.

Now you can follow the instructions for adding a new telephone (or adding a fax or modem). Avaya Communication Manager now recognizes the new type (6220 or modem) that you enter in the **Type** field.

Be sure to see your telephone's manual for instructions on how to set feature buttons and call appearance buttons. Note that if you need to use an alias for a telephone, you may not be able to take advantage of all the features of the new telephone.

Customizing Your telephone

This section provides recommendations for setting up or enhancing your personal telephone. You need a telephone that is powerful enough to allow you to use all the features you may give to other employees. You may want to add feature buttons that allow you to monitor or test the system, so that you can troubleshoot the system from your telephone.

It will be much easier to monitor and test your system if you have a telephone with:

- a large multi-button display (such as 8434D or 8410D)
- a class of service (cos) that has console permissions
- the following feature buttons
  - ACA and Security Violations (assign to lamp buttons)
  - Busy verify
  - Cover message retrieval button
  - Major/minor alarm buttons
  - Trunk ID buttons
  - Verify button

Once you select a telephone, you'll want to determine if you want to place this telephone at your desk or in the server room. If the telephone is in the server room (near the system administration terminal), you can quickly add or remove feature buttons to test features and facilities. You may decide that you want a telephone at both your desk and in the server room — it's up to you.

You may also find it handy to set up multiple phones for testing applications and features before you provide them to users. You may want to have a telephone that mimics each type of user telephone in your organization. For example, if you have four basic telephone templates, one for executives, one for marketing, one for technicians, and one for other employees, you may want to have examples of each of these phones so you can test new features or options. Once you are satisfied that a change works on the test telephone, you can make the change for all the users in that group.

Upgrading Phones

If you want to change telephone types for a user and do not need to change locations, you can just access the station screen for that extension and enter the new model number.

Tip:

This method can be used only if the new telephone type matches the existing port type (such as digital telephone with a digital port).

For example, if a user at extension 4556 currently has a 7410+ telephone and you want to replace it with a new 8411D telephone:

1. Type **`change station 4556`** and press **Enter**.

   The **Station** screen for 4556 appears.

2. Overwrite **7410+** with **8411D** in the **Type** field.

   Now you can access the functions and feature buttons that correspond to an 8411D telephone.

Swapping Phones

You will often find that you need to move or swap phones. For example, employees moving from one office to another may want to bring their phones. In this case, you can use X ports to easily swap the phones.

In general, to swap one telephone (telephone A) with another telephone (B), you change telephone A's port assignment to **x**, change telephone B's port assignment to A's old port, and, finally, change the **x** for telephone A to B's old port. Note that these swapping instructions work only if the two phones are the same type (both digital or both analog, etc.).

For example, to swap phones for extension 4567 (port 01C0505) and extension 4575 (port 01C0516), complete the following steps:

1. Type **`change station 4567`** and press **Enter**.
2. Record the current port address (`01C0505`) and type **x** in the **Port** field.
3. Press **Enter** to save your changes.
4. Type **`change station 4575`** and press **Enter**.
5. Record the current port address (`01C0516`).
6. Type **01C0505** in the **Port** field.
7. Update the **Room** and **Jack** fields.

8. Press **Enter** to save your changes.
9. Type **change station 4567** again and press **Enter**.
10. Type **01C0516** in the **Port** field.

    This is the port that used to be assigned to extension 4575.
11. Update the **Room** and **Jack** fields.
12. Press **Enter** to save your changes.
13. Physically unplug the phones and move them to their new locations.

When you swap phones, the system keeps the old button assignments. If you are swapping to a telephone with softkeys, the telephone could have duplicate button assignments, because softkeys have default assignments. You may want to check your button assignments and modify them as necessary.

Using ACTR to Move Phones

Automatic Customer Telephone Rearrangement (ACTR) allows a telephone to be unplugged from one location and moved to a new location without additional administration in Avaya Communication Manager. Communication Manager automatically associates the extension to the new port. ACTR works with 6400 Serialized phones and with the 2420/2410 phones. The 6400 Serialized telephone is stamped with the word "Serialized" on the faceplate for easy identification. The 6400 Serialized telephone memory electronically stores its own part ID (comcode) and serial number, as does the 2420/2410 telephone. ACTR uses the stored information and associates the telephone with new port when the telephone is moved.

ACTR is an enhancement to Terminal Translation Initialization (TTI), Personal Station Access (PSA), Customer Telephone Activation (CTA). ACTR makes it easy to identify and move phones.

CAUTION:

When a telephone is unplugged and moved to another physical location, the Emergency Location Extension field must be changed for that extension or the USA Automatic Location Identification data base must be manually updated. If the Emergency Location Extension field is not changed or if the USA Automatic Location Identification data base is not updated, the DID number sent to the Public Safety Acess Point (PSAP) could send emergency response personnel to the wrong location.

Detailed description

On the **Feature-Related System Parameters** screen, set the **Terminal Translation Initialization (TTI) Enabled** field to **y** and the **TTI State** field to **voice**.

Note:

When a telephone is moved, if there is any local auxiliary power (a power supply plugged into a local AC outlet), the telephone must be plugged into an AC outlet at the telephone's new location. A telephone with remote auxiliary power must be supplied remote auxiliary power at its new location. If you do not supply auxiliary power in either case after a telephone is moved, some optional adjuncts (for example, an expansion module) do not operate.

When you enter **always** or **once** in the **Automatic Moves** field on the **Station** screen, Communication Manager adds the extension to its ACTR Move List database. When the telephone is plugged in, Communication Manager asks the telephone for its serial number and records the serial number on the ACTR Move List. If you change the entry in the **Automatic Moves** field from **always** or **once** to **no**, Communication Manager removes the extension from the Move List.

Call processing

When a telephone is unplugged while on a call, and a 6400 Serialized telephone or a 2420/2410 telephone that is administered for automatic moves is plugged into the port within 60 seconds:

- both extensions are placed in idle state
- active calls on either extension are dropped, unless the call is active on a bridged appearance at some other telephone
- held calls remain in a hold state
- any calls ringing on either extension instantly proceed to the next point in coverage or station hunting path, unless the call is ringing on a bridged appearance at some other telephone
- user actions that were pending when the new telephone was plugged in are aborted

Design considerations

You can use the **`list station movable`** command to keep track of extensions on the move list. Once you reach the maximum number, Communication Manager does not allow additional extensions.

Administration

Before you start

- Be sure the **TTI** field on the **Feature-Related System Parameters** screen is set to **y**.

Tasks

Before you move a telephone in your system, set the **TTI State** field to voice on the **Feature-Related System Parameters** screen.

Moving phones

You can allow a telephone to be unplugged from one location and moved to a new location without additional administration on Avaya Communication Manager.

For example, to allow moves anytime for a telephone at extension 1234:

1. Type **change station 1234** and press **Enter**.
2. Move to the **Automatic Moves** field.

Figure 25: Station screen

```
change station 1234                                          Page 2 of X
                                  STATION

FEATURE OPTIONS
           LWC Reception? spe                Auto Select Any Idle Appearance? n
          LWC Activation? y                            Coverage Msg Retrieval? y
  LWC Log External Calls? n                                    Auto Answer: none
             CDR Privacy? n                                 Data Restriction? n
   Redirect Notification? y                       Idle Appearance Preference? n
 Per Button Ring Control? n
   Bridged Call Alerting? n                         Restrict Last Appearance? y
  Active Station Ringing: single

        H.320 Conversion? n            Per Station CPN - Send Calling Number? _
       Service Link Mode: as-needed
         Multimedia Mode: basic
    MWI Served User Type: ______                   Display Client Redirection? n
         Automatic Moves: always
              AUDIX Name:                        Select Last Used Appearance? n
                         _                         Coverage After Forwarding? _
     Recall Rotary Digit? n                          Multimedia Early Answer? n

Remote Softphone Emergency Calls: as-on-local   Direct IP-IP Audio Connections? n
Emergency Location Ext: 75001                          IP Audio Hairpinning? n
```

3. Type **always** in the **Automatic Moves** field.
4. Press **Enter** to save your changes.

Using TTI to Move Phones

Terminal Translation Initialization (TTI) allows you to merge an x-ported station to a valid port by dialing a TTI merge code, a system-wide security code, and the x-port extension from a telephone connected to that port. TTI also allows you to separate an extension from its port by dialing a similar separate digit sequence. This action causes the station to revert to an x-port.

TTI can be used for implementing telephone and data module moves from office to office. That is, you can separate a telephone from its port with TTI, unplug the telephone from the jack, plug in the telephone in a jack in a different office, and merge the telephone to its new port with TTI.

If you are moving phones and concerned about security, you may also want to see Setting up Personal Station Access on page 369 for more information about setting the security code for each extension.

SECURITY ALERT:

If you do not manage this feature carefully, its unauthorized use may cause you security problems. For example, someone who knows the TTI security code could disrupt normal business functions by separating telephones or data terminals. You can help protect against this action by frequently changing the TTI security code. You can further enhance system security by removing the feature access code (FAC) from the system when it does not need to be used (for example, there are no moves going on at present). Consult the *Avaya Products Security Handbook* for additional steps to secure your system and find out about obtaining information regularly about security developments.

Before you start

Before you can merge a telephone, you must set the **TTI State** field to **voice** on the **Feature-Related System-Parameters** screen. You also must set the extension to match the port type of the TTI port making the merge request. For example, a digital telephone type can merge only to a port on a digital board.

CAUTION:

When a telephone is unplugged and moved to another physical location, the **Emergency Location Extension** field must be changed for that extension or the USA Automatic Location Identification data base must be manually updated. If the **Emergency Location Extension** field is not changed or if the USA Automatic Location Identification data base is not updated, the DID number sent to the Public Safety Network could send emergency response personnel to the wrong location.

Note:

You cannot use TTI to change a virtual extension.

Instructions

TTI merge from a voice TTI port

CAUTION:

You can destroy your hardware if you attempt to connect an analog telephone to a digital port.

To merge an extension to a telephone with TTI, complete the following steps from the telephone you want to merge:

1. Dial the TTI merge FAC.
   - If the code is correct, you receive dial tone.
   - If the code is not correct, you receive intercept tone.
2. Dial the TTI security code from the telephone you want to merge.
   - If the code is correct, you receive dial tone.
   - If the code is not correct, you receive intercept tone.

3. Dial the extension of the telephone you want to merge.
   - If the extension is valid, you receive confirmation tone, which may be followed by dial tone. (It is possible to receive intercept tone immediately following the confirmation tone. If this happens, you need to attempt the merge again.)
   - If the extension is valid, but the extension is being administered, you receive reorder tone. Try the merge again later.
   - If the extension is invalid, you receive intercept tone.
   - If the system is busy and cannot complete the merge, you receive reorder tone. Try the merge again later.
   - If the telephone has a download status of pending, you receive reorder tone. You need to change the download status to complete to successfully complete the TTI merge.

TTI separation from a telephone

To complete a TTI separation, complete the following steps from the telephone that needs to be separated:

1. Dial the TTI separate FAC.
2. Dial the TTI security code.
   - If the code is correct, you receive dial tone.
   - If the code is not correct, you receive intercept tone.
3. Dial the extension of the telephone to be separated.
   - If you have dialed the extension of the telephone currently merged with this telephone, you receive confirmation tone.
   - If you have dialed the extension of the telephone currently merged with this telephone, but the extension is being administered, you receive reorder tone. Try the separation again later.
   - If you have not dialed the extension of the telephone currently merged with this telephone, you receive intercept tone.
   - If the system is busy and cannot complete the separation, you receive reorder tone. Try the separation again later.

Fixing problems

If you are having difficulty using TTI, you may want to review the following system restrictions:

- The **TTI Ports** field on the **System Capacity** screen (type **`display capacity`**) shows the number of TTI ports used in a server running Communication Manager. This field shows only the number of TTI ports being administered. If a TTI exceeds the maximum number of ports, the port is not administered and cannot be added. In that case, a telephone cannot be added. For details on the **System Capacity** screen, see *Maintenance Commands for Avaya Communication Manager, Media Gateways and Servers*, 03-300191.

  BRI endpoints are only counted as one TTI port. For example, for every two BRI endpoints, one TTI port is counted. As such, you can have two telephones assigned to one port. If either endpoint is administered, the TTI port count is reduced by 1.

- The total number of translated telephones and Voice TTI ports in a system is limited to the maximum number of administered telephones supported in the system. The total number of translated data terminals and Data TTI ports in a system is limited to the maximum number of administered data modules allowed in the system.

- Set the **TTI State** field to **voice** and then set the **TTI State** field to **data**. When you use this order, voice and then data, you reduce the chance of a user trying to use TTI on a data-only terminal that does not have TTI port translation. This can happen when the number of telephones allowed by the system is twice the number of data terminals. For example, if the system limit for telephones is 15,000 and 7,500 for data, then when TTI was turned on for data first, only the first 7,500 unadministered ports would get TTI port translations.

- When TTI is activated for the system, the following actions take place:

  - If the **TTI State** field was previously activated but in a different state (such as, a voice to data state), the old TTI translations are removed and the new ones added on a board by board basis.

  - If the **TTI State** field is set to **voice**, then default TTI translations are generated for every unadministered port on all digital, hybrid, and analog boards.

  - If the **TTI State** field is set to **data**, then default TTI translations are generated for every unadministered port on all digital and data line boards in the system.

  - Whenever a new digital board is inserted when the system is in TTI Data mode, or when a digital, hybrid, or analog board is inserted when the system is in TTI Voice mode, the unadministered ports on the board become TTI ports.

  - When TTI is deactivated, all translation for the TTI ports are removed in the system; the ports return to an unadministered state.

Removing Phones

Before you physically remove a telephone from your system, check the telephone's status, remove it from any group or usage lists, and then delete it from the system's memory.

For example, to remove a telephone at extension 1234:

1. Type **`status station 1234`** and press **Enter**.

   The **General Status** screen appears.

2. Make sure that the telephone:

   - is plugged into the jack
   - is idle (not making or receiving calls)
   - has no messages waiting
   - has no active buttons (such as Send All Calls or Call Forwarding)

3. Type **`list groups-of-extension 1234`** and press **Enter**.

   The **Extension Group Membership** screen shows whether the extension is a member of any groups on the system.

4. Press **Cancel**.

5. If the extension belongs to a group, access the group screen and delete the extension from that group.

   For example, if extension 1234 belongs to pickup group 2, type **`change pickup group 2`** and delete the extension from the list.

6. Type **`list usage extension 1234`** and press **Enter**.

   The **Usage** screen shows where the extension is used in the system.

7. Press **Cancel**.

8. If the extension appears on the **Usage** screen, access the appropriate feature screen and delete the extension.

   For example, if extension 1234 is bridged onto extension 1235, type **`change station 1235`** and remove the appearances of 1234.

9. Type **`change station 1234`** and press **Enter**.

10. Delete any bridged appearances or personal abbreviated dialing entries and press **Enter**.

11. Type **`remove station 1234`** and press **Enter**.

    The system displays the station screen for this telephone so you can verify that you are removing the correct telephone.

    Tip:

    Be sure to record the port assignment for this jack in case you want to use it again later.

12. If this is the correct telephone, press **Enter**.

    If the system responds with an error message, the telephone is busy or still belongs to a group. Press **Cancel** to stop the request, correct the problem, and enter **`remove station 1234`** again.

13. Remove the extension from voice mail service if the extension has a voice mailbox.

14. Type **`save translations`** and press **Enter** to save your changes.

Note that you do not need to delete the extension from coverage paths. The system automatically adjusts coverage paths to eliminate the extension.

Now you can unplug the set from the jack and store it for future use. You do not need to disconnect the wiring at the cross-connect field. The extension and port address remain available for assignment at a later date.

Once you successfully remove a set, that set is permanently erased from system memory. If you want to reactivate the set, you have to add it again as though it were a new telephone.

Adding a Fax or Modem

Connecting a fax machine or modem to your system is similar to adding a telephone, with a few important exceptions. If you have not added a telephone, you may want to read Adding New Phones on page 78.

Because the system does recognize the concept of "fax" or "modem," you need to administer these items as basic analog stations. You can merely use the supported station type 2500 (analog, single line).

Alternatively, you can create aliases to the 2500 for fax machines and modems. If you want to be able to create reports that indicate which stations are faxes or modem, you should create aliases for these items. For more information about aliasing, see Using an Alias on page 84.

For this example, let us assume that we have already defined an alias for ‘fax’ as a 2500 and that we now want to add a fax machine to extension 4444.

To add a fax machine as extension 444, complete the following steps:

1. Type **`add station 4444`** and press **Enter**.
2. In the **Type** field, type **fax**.

3. In the **Port** field, type the port address.
4. In the **Name** field, type a name to associate with this fax.
5. Move to the **Data Restriction** field and type **y**.

   Entering **y** in this field prevents calls to and from this extension from being interrupted by tone signals. This is important for fax machines and modems as these signals can disrupt transmissions of data.
6. In the **Distinctive Audible Alert** field, type **n**.

   This eliminates the distinct 2-burst ring for external calls, which often interferes with the auto-answer function on fax machines or modems.
7. Press **Enter** to save your changes.

Enabling transmission over IP networks for modem, TTY, and fax calls

Prerequisites

The ability to transmit fax, modem, and TTY calls over IP trunks or LANs and WANs assumes that the endpoints sending and receiving the calls are connected to a private network that uses H.323 trunking or LAN connections between gateways and/or port networks. This type of transmission also assumes that calls can either be passed over the public network using ISDN-PRI trunks or passed over an H.323 private network to Communication Manager switches that are similarly enabled.

As a result, it is assumed that you have assigned, or will assign, to the network gateways the IP codec you define in this procedure. For our example, the network region 1 will be assigned codec set 1, which you are enabling to handle fax, modem, and TTY calls.

To enable transmission over IP networks for modem, TTY, and fax calls:

8. Type **`change ip-codec-set 1`** and press **Enter**.

   The **IP Codec Set** screen appears.

Complete the fields as required for each media type you want to enable, and press **Enter**.

Adding an IP Softphone

Avaya IP Softphones enable the end user to control telephone calls directly from a personal computer (PC). An end user can log in remotely to your company's server running Avaya Communication Manager and then make and receive telephone calls from the telephone extension.

Avaya IP Softphone supports the following modes:

- Road-Warrior

  You typically use this mode for laptop users who are travelling. In this mode, the PC LAN connection carries both the call control signaling and the voice path. Because the audio portion of the voice call is handled by the PC, you must have some kind of audio device (e.g., handset, headset) PC to provide the audio connection.

- Telecommuter or Avaya IP Agent

  For the telecommuter or Avaya IP Agent mode, you make two separate connections to the Avaya DEFINITY server. The signaling path is carried over an IP network and the voice path is carried over the standard circuit-switched telephone network (PSTN). Since you are using a telephone for audio, you do not need an H.323 PC audio application.

  The telecommuter mode uses the Avaya IP Softphone interface (on the user's PC) and a standard telephone. The Avaya IP Agent mode uses the Avaya IP Agent interface (on the agent's PC) and a call center telephone, such as a CallMaster.

- Native H.323 (only available with Avaya IP Softphone R2)

  The stand-alone H.323 mode enables travelers to use some Communication Manager features from a remote location. This mode uses a PC running an H.323 v2-compliant audio application, such as Microsoft NetMeeting. The H.323 mode controls the call signaling and the voice path. However, since it does not use the IP Softphone interface, this configuration is capable of operating only as an analog or single-line telephone making one call at a time without any additional assigned features. You can provide stand-alone H.323 users only features that can they can activate with dial access codes.

- Control of IP Telephone (only available with IP Softphone R4 and later)

  This mode allows you to make and receive calls under the control of the IP Softphone - just like in the Telecommuter or Road Warrior mode. The big difference is that you have a real digital telephone under your control. In the Road Warrior mode, there is no telephone. In the Telecommuter mode, the telephone you are using (whether analog, digital, or IP telephone is brain dead). In this mode (if you have an IP telephone), you get the best of both worlds.

- Control of DCP Telephone (only available with IP Softphone R5 and later)

  This feature provides a registration endpoint configuration that will allow an IP softphone and a non-softphone telephone to be in service on the same extension at the same time. In this new configuration, the call control is done by both the softphone and the telephone endpoint. The audio is done by the telephone endpoint.

Tip:

Use **`status station`** to show the part (product) ID, serial number, and the audio connection method used by existing stations.

Note:

Beginning with the November 2003 release of Communication Manager, R1 and R2 IP Softphone and IP Agent, which use a dual connect (two extensions) architecture, are no longer supported. R3 and R4 IP Softphone and IP Agent, which use a single connect (one extension) architecture, continue to be supported. This applies to the RoadWarrior and the Telecommuter configurations for the IP Softphone. Native H.323 registrations for R1 and R2 Softphones continue to be supported.

Before you start

Be sure that your system has been enabled to use IP Softphones. Display the **System Parameters Customer-Options** screen and verify the following field settings:

- **Maximum Concurrently Registered IP Stations** is greater than **0**
- **IP Stations** field is **y**
- Information has been entered in the fields on the **Maximum IP Registrations by Product ID** page

Be sure that your DEFINITY Server R, CSI, or SI has a CLAN board and an IP Media Processor board.

Once you're finished administering Avaya Communication Manager, you need to install the IP Softphone software on each user's PC.

Adding a road-warrior mode

You can use the road-warrior mode when you have only a single telephone line available to access Avaya Communication Manager over the IP network.

You also can "take over" an IP telephone. Typically you would not have a different extension for your softphone. When you log in, the softphone takes over the existing telephone extension (turn the DCP or IP telephone off). During this time, that DCP or IP telephone is out of service. This is accomplished if, on the **Station** screen, the **IP Softphone** field is **y**.

We will add a road-warrior mode at extension 3001. Except for single-connect IP phones, you have to actually administer two extensions for each road-warrior mode. We will first add an H.323 extension at 3000.

To assign an H.323 extension, complete the following steps:

1. Type **add station 3000** and press **Enter.**

   The **Station** screen appears.

Figure 26: Station screen

```
add station 3000                                          Page   1 of x
                                  STATION

Extension: 3000                      Lock Messages? n             BCC: 0
     Type: H.323                     Security Code:                TN: 1
     Port: IP                      Coverage Path 1:               COR: 1
     Name:                         Coverage Path 2:               COS: 1
                                   Hunt-to Station:             Tests? y

STATION OPTIONS
              Loss Group: 2
```

2. In the **Type** field, enter **H.323**.
3. Press **Enter** to save your work.

Now, you need to administer the telephone (DCP) extension. To do so, complete the following steps:

1. Type **add station 3001** and press **Enter**.

   The **Station** screen appears.

Figure 27: Station screen

```
add station 3001                                          Page   1 of   x
                                STATION

Extension: 3001                        Lock Messages? n           BCC: 0
     Type: 6408D                       Security Code: *            TN: 1
     Port: X                         Coverage Path 1:             COR: 1
     Name:                           Coverage Path 2:             COS: 1
                                     Hunt-to Station:

STATION OPTIONS
              Loss Group: 2         Personalized Ringing Pattern: 1
            Data Module? n                    Message Lamp Ext: 3000
           Speakerphone: 2-way             Mute Button Enabled? y
       Display Language: english

                                           Media Complex Ext: 3000
                                                   IP Softphone? y
```

Note that you choose to change an existing DCP extension by using **`change station`** ***nnnn*** in this step, where ***nnnn*** is the existing DCP extension.

2. In the **Type** field, enter the model of telephone you want to use, such as **6408D**.
3. In the **Port** field, type **x** for virtual telephone or enter the port number if there is hardware.

   Note:

   Port 1720 is turned off by default to minimize denial of service situations. This applies to all IP softphones release 5.2 or later. You can change this setting, if you have root privileges on the system, by typing the command: **`/opt/ecs/sbin ACL 1720 on`** or **`off`**.

4. In the **Security Code** field, enter the password for this remote user, such as **1234321**.

   This password can be 3-8 digits in length.

5. In the **Media Complex Ext** field, type **3000**.

   This is the H.323 extension just administered.

6. In the **IP Softphone** field, type **y**.
7. On page 2, in the **Service Link Mode** field, type **as-needed**.

   Set this field to **permanent** only for extremely busy remote telephone users, such as call center agents.

8. In the **Multimedia Mode** field, type **enhanced**.
9. Press **Enter** to save your work.

Now you can install and configure the software on the user's PC. In this example, the user will login by entering their DCP extension (3001) and password (1234321).

Fixing problems

Problem	Possible causes	Solutions
Display characters on the telephone can not be recognized.	Microsoft Windows is not set to use Eurofont characters.	Set the Microsoft Windows operating system to use Eurofont.

Adding a telecommuter mode

Assign this configuration to remote users who have two available telephone lines. For example, to administer a telecommuter mode for a home user at extension 3010, complete the following steps:

1. Type **add station 3010** and press **Enter.**

   The **Station** screen appears.

   Note:

   Use the **add station** command if this is a new DCP extension. Use the **change station** command for an existing DCP extension and ignore steps 2 and 3.)

2. In the **Port** field, type **x** for virtual telephone or enter the port number if there is hardware.
3. In the **Security Code** field, enter the password for this remote user, such as **1234321**.

   This password can be up to 7 digits in length.

4. In the **IP Softphone** field, type **y**.
5. On page 2, in the **Service Link Mode** field, type **as-needed**.

   Set this field to **permanent** only for extremely busy remote telephone users, such as call center agents.

6. In the **Multimedia Mode** field, type **enhanced**.
7. Press **Enter** to save your work.

Now you can install and configure the software on the user's PC. In this example, the user will login by entering their DCP extension (3010) and password (1234321).

Related topics

See the online help and to *Avaya IP Softphone Overview and Troubleshooting* for customer information on Avaya IP Softphone modes. This document is a Portable Document Format (PDF) document that is located in the Overview Document folder on the Avaya IP Softphone CD.

Also see *Getting Started*, located on the Avaya IP Softphone CD for more information on how to install and configure the IP Softphone software.

Adding an IP Telephone

The 4600-series IP Telephones are physical sets that connect to Avaya Communication Manager via TCP/IP.

CAUTION:

An Avaya IP endpoint can dial emergency calls (for example, 911 calls in the U.S.). It only reaches the local emergency service in the Public Safety Answering Point area where the telephone system has local trunks. Please be advised that an Avaya IP endpoint cannot dial to and connect with local emergency service when dialing from remote locations that do not have local trunks. You should not use an Avaya IP endpoint to dial emergency numbers for emergency services when dialing from remote locations. Avaya Inc. is not responsible or liable for any damages resulting from misplaced emergency calls made from an Avaya endpoint. Your use of this product indicates that you have read this advisory and agree to use an alternative telephone to dial all emergency calls from remote locations.

Before you start

Verify the system has a:

- TN2302 IP Media Processor circuit pack for audio capability
- TN799 Control-LAN circuit pack for signaling capability (for Servers R, CSI, and SI only)

Be sure that your system has been enabled to use IP Telephones. Display the **System-Parameters Customer-Options** screen and verify the following field settings:

- **Maximum Concurrently Registered IP Stations** is greater than 0
- **IP Stations** field is **y**
- Information has been entered in the fields on the **Maximum IP Registrations by Product ID** page

Adding an IP Telephone

Let us add an IP telephone at extension 4005.

To assign an extension, complete the following steps:

1. Type **add station 4005** and press **Enter.**

   The **Station** screen appears.

Figure 28: Station screen

```
                                                  Page 1 of x
                              STATION

Extension: 2345                    Lock Messages? n          BCC: _
     Type: 4620+                   Security Code: _________   TN: 1_
     Port: IP                    Coverage Path 1: ____       COR: 1
     Name: _____________________ Coverage Path 2: ____       COS: 1
                                 Hunt-to Station: _____

STATION OPTIONS
          Loss Group: none       Personalized Ringing Pattern: 1
                                             Message Lamp Ext: 2345
        Speakerphone: 2-way                Mute Button Enabled? y
    Display Language: english                 Expansion Module? y

                                           MM Complex Data Ext:
                                                  IP SoftPhone? n
```

2. In the **Type** field, type **4620**.
3. The **Port** field is display-only, and **IP** appears.
4. In the **Security Code** field, enter a password for the IP telephone user.

   Note:

   Although the system accepts a null password, the IP telephone will not work unless you assign a password.

5. Press **Enter** to save your work.

Changing from dual-connect to single-connect IP phones

When you have a dual extension telephone and you upgrade to a single extension telephone, you can remove the connection that is no longer used for that telephone.

To remove the H.323 connection that is no longer needed, first record the media complex extension number:

1. Type **`change station nnnn`**, where nnnn is the extension number of the original dual-connect telephone that you are replacing with a single-connect telephone.

   The **Station** screen appears.

2. Move to the **Media Complex Extension** field.
3. Write down the number in the **Media Complex** field, then delete the number from the field.
4. Press **Enter** to save your changes.

Now remove the extension you recorded. Before you remove an H.323 extension from your system, check the status, remove it from any group or usage lists, and then delete it from the system's memory.

For example, if you wrote down extension 1234 before you removed it from the Media Complex field on the Station screen, then remove extension 1234 using these steps:

1. Type **`status station 1234`** and press **Enter**.

   The **General Status** screen appears.

2. Make sure that the extension:
   - is idle (not making or receiving calls)
   - has no messages waiting
   - has no active buttons (such as Send All Calls or Call Forwarding)
3. Type **`list groups-of-extension 1234`** and press **Enter**.

   The **Extension Group Membership** screen shows whether the extension is a member of any groups on the system.

4. Press **Cancel**.
5. If the extension belongs to a group, access the group screen and delete the extension from that group.

   For example, if extension 1234 belongs to pickup group 2, type **`change pickup group 2`** and delete the extension from the list.

6. Type **`list usage extension 1234`** and press **Enter**.

   The **Usage** screen shows where the extension is used in the system.

7. Press **Cancel**.

8. If the extension appears on the Usage screen, access the appropriate feature screen and delete the extension.

   For example, if extension 1234 belongs to hunt group 2, type **change hunt group 2** and delete the extension from the list.

9. Type **change station 1234** and press **Enter**.
10. Delete any bridged appearances or personal abbreviated dialing entries and press **Enter**.
11. Type **remove station 1234** and press **Enter**.

    The system shows the station screen for this telephone so you can verify that you are removing the correct telephone.

12. If this is the correct telephone, press **Enter**.

    The system responds with **command successfully completed**.

    If the system responds with an error message, the telephone is busy or still belongs to a group. Press **Cancel** to stop the request, correct the problem, and type **remove station 1234** again.

13. Remove the extension from voice mail service if the extension has a voice mailbox.
14. Type **save translations** and press **Enter** to save your changes.

Note that you do not need to delete the extension from coverage paths. The system automatically adjusts coverage paths to eliminate the extension.

Once you successfully remove the extension, it is permanently erased from system memory. If you want to reactivate the extension, you have to add it again as though it were new.

Tasks

Setting up emergency calls on IP phones

Set up which "calling number" to send to the public safety access point when an emergency call is placed from an IP telephone.

Instructions

You use the **Station** screen to set up emergency call handling options for IP phones. As an example, we'll administer the option that prevents emergency calls from an IP telephone.

To prevent an emergency call from an IP telephone:

1. Type **change station nnnn**, where ***nnnn*** is the extension of the telephone you want to modify. Press **Enter**.

   The **Station** screen appears.

Figure 29: Station screen

```
change station 75001                                          Page 2 of X
                                    STATION

FEATURE OPTIONS
           LWC Reception? spe                 Auto Select Any Idle Appearance? n
          LWC Activation? y                           Coverage Msg Retrieval? y
  LWC Log External Calls? n                                   Auto Answer: none
            CDR Privacy? n                                Data Restriction? n
   Redirect Notification? y                       Idle Appearance Preference? n
 Per Button Ring Control? n
   Bridged Call Alerting? n                         Restrict Last Appearance? y
  Active Station Ringing: single

       H.320 Conversion? n             Per Station CPN - Send Calling Number? _
      Service Link Mode: as-needed
        Multimedia Mode: basic
    MWI Served User Type: ______                    Display Client Redirection? n
         Automatic Moves:
              AUDIX Name:                         Select Last Used Appearance? n
                                 _                  Coverage After Forwarding? _
     Recall Rotary Digit? n                           Multimedia Early Answer? n

Remote Softphone Emergency Calls: block   Direct IP-IP Audio Connections? n
Emergency Location Ext: 75001                        IP Audio Hairpinning? n
```

2. Type **block** in the **Remote Softphone Emergency calls** field and press **Enter** to save your changes.

CAUTION:

An Avaya IP endpoint can dial emergency calls (for example, 911 calls in the U.S.). It only reaches the local emergency service in the Public Safety Answering Point area where the telephone system has local trunks. Please be advised that an Avaya IP endpoint cannot dial to and connect with local emergency service when dialing from remote locations that do not have local trunks. You should not use an Avaya IP endpoint to dial emergency numbers for emergency services when dialing from remote locations. Avaya Inc. is not responsible or liable for any damages resulting from misplaced emergency calls made from an Avaya endpoint. Your use of this product indicates that you have read this advisory and agree to use an alternative telephone to dial all emergency calls from remote locations. Please contact your Avaya respresentative if you have questions about emergency calls from IP phones.

Setting up Remote Office

Avaya Remote Office provides IP processing capabilities to traditional call handling for voice and data between Avaya Communication Manager and offices with Remote Office hardware. You need to add the information about Remote Office as a node in Communication Manager, add its extensions, and set up the trunk and signaling groups.

Before you start

Be sure the following fields on the **Feature Related System Parameters Customer Options** screen are set to **y** or completed. If not, contact your Avaya representative.

- **Maximum Administered Remote Office Trunks**
- **Maximum Administered Remote Office Stations**
- **Product ID registration limit**
- **Remote Office**
- **IP station**
- **ISDN-PRI**

Also, be sure your Remote Office hardware is installed and administered at the remote location. You need the following information from the remote administration:

- IP address
- Password

Adding Remote Office to Avaya Communication Manager

In our example, we'll set up a remote-office location using Avaya R300 Remote Office Communicator hardware in our branch office in Santa Fe. We'll add a new node, and set up the signaling group and trunk group.

Adding a node

Instructions

To add the remote office node to Communication Manager:

1. Type **`change node-names IP`** and press **Enter**.

   The **Node Name** screen appears.

Figure 30: Node Name screen

```
change node-names ip                                          Page 1 of 1
                              IP NODE NAMES

  Name                  IP Address            Name           IP Address
default              0  .0  .0  .0       ____________________  ___.___.___.___
remote office 1      134.23.107.22       ____________________  ___.___.___.___
____________________  ___.___.___.___    ____________________  ___.___.___.___
____________________  ___.___.___.___    ____________________  ___.___.___.___
____________________  ___.___.___.___    ____________________  ___.___.___.___
____________________  ___.___.___.___    ____________________  ___.___.___.___
____________________  ___.___.___.___    ____________________  ___.___.___.___
____________________  ___.___.___.___    ____________________  ___.___.___.___
____________________  ___.___.___.___    ____________________  ___.___.___.___
____________________  ___.___.___.___    ____________________  ___.___.___.___
```

2. In the **Name** field, type in a word to identify the node.

   In our example, type **Remote 6**.

3. In the IP address field, type in the IP address to match the one on the Avaya R300 administration.

4. Press **Enter** to save your changes.

5. Type **`add remote office`** and the number for this remote office, and press **Enter**.

   The **Remote Office** screen appears.

Figure 31: Remote Office screen

```
add remote-office 6                                           Page 1 of 1
                                REMOTE OFFICE 6

     Node Name:  Remote Office 6
Network Region:  22
      Location:  1
     Site Data:  Contact: Joe Smith
                 Phone: xxx-yyy-zzz
                 _____________________________________
```

6. Fill in the following fields:
   - **Node Name** - match the name on the IP Node Names screen.
   - **Network Region** - this must match the network region on the IP-Interface screen for the circuit packs that connect this remote office. Use **`display ip-interfaces`** to find this information.
   - **Location** - match the one set up on the **Location** screen for this remote office.
   - **Site Data** - identify the street address or identifier you want to use
7. Press **Enter** to save your changes.

Tip:

Use **`status remote office`** to verify that your server running Communication Manager recognizes the Remote Office information. It also displays the extensions and signaling group you administer next.

Setting up a trunk group

You can modify an existing trunk group or add a new one. In our example, we'll add trunk group 6.

Before you start

Perform Setting up a signaling group on page 111.

To set up the trunk group for your remote office:

1. Type **`add trunk group 6`**.

   The **Trunk Group** screen appears.

Figure 32: Trunk Group screen

```
add trunk-group next                                          Page 1 of x
                                TRUNK GROUP

 Group Number: 6___                 Group Type: _________      CDR Reports: _
   Group Name: ________________________________ COR: __       TN: ___      TAC: __6__
    Direction: _________       Outgoing Display? _    Trunk Signaling Type: ____
  Dial Access? _                 Busy Threshold: ___         Night Service: _________
 Queue Length: ___                                    Incoming Destination: _________
Service Type: _____                 Auth Code? _
                                   Trunk Flash? _
          BCC: _                           ITC? ____
 TRUNK PARAMETERS
    Trunk Type (in/out): _____________     Incoming Rotary Timeout(sec): __
     Outgoing Dial Type: _________                  Incoming Dial Type: _________
                                                Disconnect Timing(msec): ____
        Digit Treatment: ___________                              Digits: ____
                                                   Sig Bit Inversion: none
      Analog Loss Group: ___                       Digital Loss Group: ___
     Incoming Dial Tone? _
Incoming Calling Number - Delete: ___ Insert: ________ Format: ________
               Bit Rate: _____        Synchronization: _____    Duplex: ____
  Disconnect Supervision - In? _  Out? _
  Answer Supervision Timeout: ___           Receive Answer Supervision? _
```

2. In the **Group Type** field, type **ISDN**.

   ISDN-PRI or **ISDN-BRI** must be **y** on the **System Parameters Customer Options** screen.
3. In the **TAC** field, type in the trunk access code that conforms to your dial plan.
4. In the **Carrier Medium** field, type **IP** (Medpro).
5. In the **Dial Access** field, type **y**.
6. In the **Service Type** field, type **tie**.
7. In the **Signaling Group** field, type in the signaling group you created.
8. Press **Enter** to save your changes.

Setting up a signaling group

Each Remote Office has own listen port and signaling group. Set up a new trunk group, or use an existing trunk group administered for H.323 signaling.

Instructions

Set up the signaling group for remote office:

1. Type **add signaling-group** and the number of the group you want to add.

   The **Signaling Group** screen appears.

Figure 33: Signaling Group screen

```
add signaling-group 6                                        Page 1 of 5
                              SIGNALING GROUP

Group Number  6             Group Type:  H.323
                          Remote Office?  y     Max Number of NCA TSC: ___

                                                  Max number of CA TSC:  ___
                                               Trunk Group for NCA TSC:  ___
     Trunk Group for Channel Selection:  6
        Supplementary Service Protocol: _


  Near-end Node Name:  clan              Far-end Node Name: remote office 6
Near-end Listen Port:  5001            Far-end Listen Port:5055

        LRQ Required? _          Calls Share IP Signaling Connection?  _
        RRQ Required? y
                                      Bypass If IP Threshold Exceeded?  _

                                          Interworking Message: ________
```

2. In the **Group Type** field, type **H.323**.
3. In the **Remote Office** field, type **y**.
4. In the **Trunk Group for Channel Selection** field, type the number of the trunk you set up for the remote office.
5. In the **Near-end Node Name** field, identify the node name assigned to the CLAN that supports the R300.
6. In the **Far-end Node Name** field, identify the node name assigned to the CLAN that supports the R300.
7. In the **Near-end Listen Port** field, type a port number in the 5000-9999 range.
8. In the **Far-end Listen Port** field, type **1720**. See *Getting Started with the Avaya R300 Remote Office Communicator* for more details.
9. In the **RRQ** field, type **y**.
10. Tab to the **Direct IP-IP Audio Connection** field on another page of this screen and type **y**.
11. Press **Enter** to save your changes.

Setting up Remote Office on network regions

Now we will set up a network region and show the connections between regions.

Instructions

Set up network region 1:

1. Type **`add ip-network-region 1`** and press **Enter**.

   The **IP Network Region** screen appears.

Figure 34: IP Network Region screen

```
add ip-network-region 1                                  Page 1 of 19
                            IP NETWORK REGION
  Region: 1
Location:                 Home Domain:
    Name:
                                Intra-region IP-IP Direct Audio: no
AUDIO PARAMETERS                Inter-region IP-IP Direct Audio: no
   Codec Set: 1                           IP Audio Hairpinning? y
UDP Port Min: 2048
UDP Port Max: 3049                        RTCP Reporting Enabled? y
                              RTCP MONITOR SERVER PARAMETERS
DIFFSERV/TOS PARAMETERS       Use Default Server Parameters? y
 Call Control PHB Value: 34                Server IP Address:   .   .   .
        Audio PHB Value: 46                      Server Port: 5005
802.1P/Q PARAMETERS                   RTCP Report Period(secs): 5
 Call Control 802.1p Priority: 7
        Audio 802.1p Priority: 6      AUDIO RESOURCE RESERVATION PARAMETERS
H.323 IP ENDPOINTS                                    RSVP Enabled? y
  H.323 Link Bounce Recovery? y              RSVP Refresh Rate(secs): 15
 Idle Traffice Interval (sec): 20     Retry upon RSVP Failure Enabled? y
    Keep-Alive Interval (sec): 6               RSVP Profile: guaranteed-service
              Keep-Alive Count: 5     RSVP unreserved (BBE) PHB Value: 40
```

2. In the **Name** field, describe the region you are setting up.
3. In the **Codec Set** field, type the codec set you want to use in this region.
4. In the **UDP Port Range** field, type the range of the UDP port number to be used for audio transport.
5. In the **Intra-region IP-IP Direct Audio** field, type **y**.
6. In the **Inter-region IP-IP Direct Audio** field, type **y**.
7. Move to page 3 to set up connections between regions and assign codecs for inter-region connections.

   Note:

   Page 2 of the **IP Network Region** screen shows a list of LSP servers for the network region, and pages 4 through 19 are duplicates of page 3, providing the ability to administer up to 250 locations. For details, see the Screen Reference section on the **IP Network Region** screen.

Figure 35: IP Network Region screen

```
add ip-network-region 3                                      Page   3 of x

                  Inter Network Region Connection Management

 src dst
 rgn rgn      codec-set  direct-WAN  WAN-BW limits  Intervening-regions
 3   1           1           y          256:Kbits
 3   2           1           n                      1    ___  ___  ___
 3   3           1
 3   4           1           n                      1    ___  ___  ___
 3   5           1           n                      6    ___  ___  ___
 3   6           1           y             :NoLimit
 3   7           1           y           10:Calls
 3   8
 3   9
 3   10
 3   11
 3   12
 3   13
 3   14
 3   15
```

The following connections are administered in this example:

- codec-set 2 is used between region 1 and region 4
- codec-set 5 is used between region 1 and region 99
- codec-set 6 is used between region 1 and region 193.

Now you need to assign the region number to the CLAN circuit pack. All the endpoints registered with a specific CLAN circuit pack belong to the CLAN's region. See *Administration for Network Connectivity for Avaya Communication Manager* for more information.

Adding phones to Remote Office

Be sure the extensions you add fit your dialing plan.

Instructions

1. Type **`add station`** ***nnnn***, where ***nnnn*** is the extension you are adding. Press **Enter**.

   The **Station** screen appears.

Figure 36: Station screen

```
add station 6001                                          Page 1 of X
                                    STATION

Extension: 1014                      Lock Messages? n         BCC: 0
     Type: 8410D                     Security Code: 1234567    TN: 1
     Port: x                         Coverage Path 1: ___     COR: 1
     Name: Remote main               Coverage Path 2: ___     COS: 1
                                       Map-to Station:
                                     Hunt-to-Station: _______

STATION OPTIONS
              Loss Group: _          Personalized Ringing Pattern:
             Data Module? n                    Message Lamp Ext: 6001
           Speakerphone: 2-way               Mute button enabled? y
       Display Language? English
                                               Media Complex Ext:
                                         IP Station? n
                                            Remote Office Phone? y
```

2. In the **Type** field, type in the model of the telephone you are adding.
3. In the **Port** field, type **x**.

   This indicates that there is no hardware associated with the port assignment.
4. In the **Name** field, identify the telephone for your records.
5. In the **Security Code** field, match the password set up on the Remote Office administration.
6. In the **Remote Office Phone** field, type **y**.
7. Press **Enter** to save your changes.

Tip:

You can set up a telnet session on your remote office administration program to verify that the telephone is registered. See *Getting Started with the Avaya R300 Remote Office Communicator* for details.

2420/2410 DCP telephone

Downloading firmware to a 2420 or 2410 DCP telephone

You can copy updated application code into Communication Manager via TFTP over a TCP/IP connection. This eliminates the need to physically remove the telephone and send it to the factory for the firmware update. This feature is available on all of the servers running Avaya Communication Manager.

Before you start

1. Type **`change node-name ip`** and press **Enter**.

   The **IP Node Names** screen appears.
2. Administer the TFTP server node name and the local node name (CLAN) and IP address.
3. Press **Enter** to save your changes.
4. Type **`change ip-interfaces`** and press **Enter**.

   The **IP Interfaces** screen appears.
5. Administer the CLAN ethernet interface or processor CLAN.
6. Press **Enter** to save your changes.

Downloading the firmware file to Communication Manager

1. Place the file on the TFTP server using TFTP, FTP, HTTP or another file transfer program.
2. From the Web Interface menu, click the **Set LAN Security** link.
3. Click **Advanced**. A list of settings that can be enabled or disabled through the use of check boxes appears.
4. Scroll to **tftp** and check the box enabling *inbound* tftp traffic.
5. Click **Submit**.
6. Log into SAT and enter **`change tftp-server`** and press **Enter**.

   The **TFTP Server Configuration** screen appears.

Figure 37: TFTP Server Configuration screen

```
change tftp-server                                              Page 1 of 1
                        TFTP Server Configuration

      Local Node Name:
TFTP Server Node Name:
     TFTP Server Port: 69
     File to Retrieve:

          File Status:
            File Size:
   Filename in Memory:
```

7. In the **Local Node Name** field, enter the valid local node name from the **IP Node Names** screen.

   The node must be assigned to a CLAN ip-interface or procr (processor CLAN).

8. In the **TFTP Server Node Name** field, enter the valid TFTP server node name from the **IP Nodes Names** screen.
9. In the **TFTP Server Port** field, enter the TFTP server port number from where the file download begins.
10. In the **File to Retrieve** field, enter the name of the file to be retrieved.
11. Press **Enter** to save your changes.

    The file transfer begins.

12. Type **`display tftp-server`** and press **Enter** to view the status of the file transfer.

    A **File download successful** message appears when the file transfer completes. It also displays the file size and the file name in memory.

Downloading firmware to a single station

You must have console permissions to download someone else's phones.

Note:

Steps 1 through 3 need be done only once to set up the FAC for file downloads. Thereafter, start at step 4 to download files.

To set up a FAC for file downloads:

1. Type **`change feature-access-codes`** and press **Enter**.

   The **Feature Access Code (FAC)** screen appears. Click **Next** until you see the **Station Firmware Download Access Code** field.

Figure 38: Feature Access Code (FAC) screen

```
change feature-access-codes                                  Page 3 of x

                              FEATURE ACCESS CODE (FAC)

            Priority Calling Access Code:
                     Program Access Code:
 Refresh Terminal Parameters Access Code:
        Remote Send All Calls Activation: Deactivation:
         Self Station Display Activation:Deactivation:
               Send All Calls Activation:Deactivation:
   Station Firmware Download Access Code:
Station Security Code Change Access Code:
        Station User Admin of FBI Assign:Remove:
```

2. In the **Station Firmware Download Access Code** field, enter a valid FAC as defined in the dial plan.
3. Press **Enter** to save your changes.
4. Take the 2410 or 2420 DCP telephone off-hook.
5. Dial the Station Firmware Download FAC (for instance, *36).
6. Press # if you are dialing from the target station (or dial the telephone's extension to be downloaded).
7. Place the telephone on-hook within 4 seconds after the confirmation tone.

   The telephone is placed in a busy-out state (not able to make or receive calls) and displays **Firmware Download in Progress**, the amount of the file downloaded, and a timer. The telephone displays error messages and a success message before rebooting.
8. When the download completes, the telephone reboots and is released from the busy-out state.

Downloading firmware to multiple stations

You can download firmware to multiple stations of the same type, either 2410 or 2420 DCP telephones. Download firmware to as many as 1000 stations per download schedule. You can schedule a specific time for the download, or you can administer the download to run immediately.

To download 2420 DCP station firmware to multiple stations:

1. Type **`change firmware station-download`** and press **Enter**.

   The **Firmware Station Download** screen appears.

Figure 39: Firmware Station Download screen

```
change firmware station-download

                              FIRMWARE STATION DOWNLOAD

Source File:

Schedule Download? y
     Start Date/Time:/: Stop Date/Time:/:
Continue Daily Until Completed? y
Download Set Type: 2420

Beginning Station:         Ending Station:
```

2. In the **Source File** field, enter the name of the file specified in the **File to Retrieve** field on the **TFTP Server Configuration** screen.
3. In the **Schedule Download** field, type **y**. The **Start Date/Time** and **Stop Date/Time** fields appear.
4. In the **Start Date/Time** field, enter the month (mm), day (dd), year (yyyy), and time (hh:mm) that you want the download to begin.
5. In the **Stop Date/Time** field, enter the month (mm), day (dd), year (yyyy), and time (hh:mm) that you want the download to begin.
6. In the **Continue Daily Until Completed** field, enter **y** if you want the system to execute the firmware download each day at the scheduled time until all specified telephones have received the firmware.
7. In the **Beginning Station** field, enter the first extension number in the range of telephones to which you want to download the firmware. Up to 1000 stations can be included in a scheduled download.

8. In the **Ending Station** field, enter the last extension number in the range of telephones to which you want to download firmware. Up to 1000 stations can be included in a scheduled download.
9. Press **Enter** to save your changes. The firmware download is set to run at the scheduled time. If you entered **n** in the **Schedule Download?** field, pressing **Enter** immediately initiates the download to the specified range of telephones.

Displaying firmware download status

You can use the **status firmware download** command to display status information for an active download schedule.

To display download status:

- Type **status firmware download**.

  The **Status Firmware Station Download** screen appears.

Figure 40: Status Firmware Download screen

```
 status firmware station-download

                              STATUS FIRMWARE STATION DOWNLOAD

Image file:
Schedule Download?               Continue daily until completed?
Start Date/Time:                 Stop Date/Time:

Terminal type for download:
Extension range:     to:         Number of stations in range:
Stations completed:              Stations unsuccessful:
```

Note:

If you add the qualifier **last** to the **status firmware download** command, status information on the last download schedule is displayed.

Disabling firmware downloads

You can use the **disable firmware download** command to disable any active download schedule.

To disable active downloads:

- Type **disable firmware download**.

  This command disables any active download schedule and the system displays **Command successfully completed** at the bottom of the screen.

Chapter 4: Managing Telephone Features

Adding Feature Buttons

Once you add a telephone to the system, you can use the station screen to change the settings for the telephone, such as adding or changing feature button assignments. The system allows you to assign features or functionality to each programmable button. It is up to you to decide which features you want for each telephone and which feature you want to assign to each button.

Note:

If you have 6400-series phones, your users can administer some of their own feature buttons. See Setting up Terminal Self Administration on page 155 for more information.

Note:

An NI-BRI telephone with Communication Manager has only the **Conference**, **Transfer**, **Hold**, and **Drop** feature buttons, none of which requires administration. On an NI-BRI telephone, you may assign additional feature buttons only as call appearances. As a result, NI-BRI telephone users must access all other features of Communication Manager using feature access codes.

Additionally, the number of call appearance buttons administered in Communication Manager (the default is three) must match the number of call appearances programmed on the telephone.

Finally, Communication Manager does not support bridged call appearances for NI-BRI phones

Instructions

To assign feature buttons:

1. Type **`change station`** ***nnnn***, where ***nnnn*** is the extension for the telephone you want to modify. Press **Enter**.

   The **Station** screen appears.

2. Press **Next Page** until you locate the **Feature Button Assignment** fields.

   Some phones have several feature button groups. Make sure that you are changing the correct button. If you do not know which button on the telephone maps to which button-assignment field, see your telephone's manual, or see Telephone Reference on page 647.

Figure 41: Station screen

```
                                        STATION
SOFTKEY BUTTON ASSIGNMENTS

 1: lwc-store
 2: lwc-cancel
 3: auto-cback
 4: timer
 5: call-fwd    Ext: _________
 6: call-park
 7: date-time
 8: priority
 9: abr-prog
10: abr-spchar  Char: ~p
11: abr-spchar  Char: ~m
12: abr-spchar  Char: ~w
```

3. Move the cursor to the field you want to change.
4. Type the button name that corresponds to the feature you want to add.

   To determine feature button names, press **Help** or refer to Telephone feature Buttons on page 123.
5. Press **Enter** to save your changes.

Some phones have default assignments for buttons. For example, the following figure shows that the 8411D includes defaults for 12 softkey buttons. It already has assignments for features like Leave Word Calling and Call Forwarding.

If you do not use an alias, you can easily assign different features to these buttons if you have different needs.

If you use an alias you must leave the default softkey button assignments. The system allows you to change the button assignments on the screen and the features work on the alias telephone, however the labels on the display do not change.

Telephone feature Buttons

The following table provides descriptions of the feature buttons that you can administer on multiappearance telephones. It also lists the administrable software names and recommended button label names. Display buttons support telephones equipped with alphanumeric displays. Note that some buttons may require 1-lamp or 2-lamp buttons. Some buttons are not allowed on some systems and on some phones.

Note:

An NI-BRI telephone with Communication Manager has only the **Conference**, **Transfer**, **Hold**, and **Drop** feature buttons, none of which requires administration. On an NI-BRI telephone, you may assign additional feature buttons only as call appearances. As a result, NI-BRI telephone users must access all other features of Communication Manager using feature access codes.

Additionally, the number of call appearance buttons administered in Communication Manager (the default is three) must match the number of call appearances programmed on the telephone.

Finally, Communication Manager does not support bridged call appearances for NI-BRI phones.

Table 2: Telephone feature buttons *1 of 21*

Button name	Button label	Description	Maximum
#	Autodial	You can administer the # button as an autodial feature button by entering the Audix number in the **Button Assignments** field on the **Station** screen.	1 per station
abr-prog	AbrvDial Program	Abbreviated Dialing Program: allows users to program abbreviated dialing and autodial buttons or to store or change numbers in a personal list or group list associated with the station.	1 per station
abr-spchar	AbrvDial (char)	Abbreviated Dialing Special Character: allows users to enter an associated special character [~, ~m (mark), ~p (pause), ~s (suppress), ~w (wait for dial tone), or ~W (wait forever)] when programming an abbreviated dialing list entry.	1 each per station

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Table 2: Telephone feature buttons *2 of 21*

Button name	Button label	Description	Maximum
abrdg-appr (Ext: _____)	(extension)	Bridged Appearance of an analog telephone: allows the user to have an appearance of a single-line telephone extension. Assign to a 2-lamp appearance button.	Depends on station type
abrv-dial (List: __ DC: __)	AD	Abbreviated Dialing: dials the stored number on the specified abbreviated dialing list. List: specify the list number 1 to 3 where the destination number is stored DC: specify the dial code for the destination number	1 per AD list per dial code
abrv-ring	AR	Abbreviated and Delayed Ringing: allows the user to trigger an abbreviated or delayed transition for calls alerting at an extension.	
ac-alarm	AC Alarm	Administered Connection alarm notification: allows the user to monitor when the number of failures for an administered connection has met the specified threshold.	1 per station
aca-halt	Auto-Ckt Assure	Automatic Circuit Assurance (*display button*): allows users of display telephones to identify trunk malfunctions. The system automatically initiates a referral call to the telephone when a possible failure occurs. When the user presses ACA Halt, the system turns off ACA monitoring for the entire system. The user must press ACA Halt again to restart monitoring.	1 per system
account	Acct	Account: allows users to enter Call Detail Recording (CDR) account codes. CDR account codes allow the system to associate and track calls according to a particular project or account number.	

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Table 2: Telephone feature buttons *3 of 21*

Button name	Button label	Description	Maximum
admin	Admin	Administration: allows a user to program the feature buttons on their 6400-series telephone.	
after-call Grp:___	After Call Work	After Call Work Mode: allows an agent to temporarily be removed from call distribution in order for the agent to finish ACD-related activities such as completing paperwork. Grp: specify the ACD split group number.	1 per split group
alrt-agchg	Alert Agent	Alert Agent: indicates to the agent that their split/skill hunt group changed while active on a call. This button blinks to notify the agent of the change.	1 per station
alt-frl	Alt FRL	Alternate Facility Restriction Level (FRL): activates or deactivates an alternate facility restriction level for the extension.	1 per system
ani-requst	ANI Request	Automatic Number Identification Request: allows the user to display the calling party's number from incoming trunks during the voice state of call. The trunk must support this functionality.	1 per station
assist (Group: __)	Assist	Supervisory Assistance: used by an ACD agent to place a call to a split supervisor. Group: specify the ACD split group number.	1 per split group
asvn-halt	asvn-halt	Authorization Code Security Violation Notification: activates or deactivates call referral when an authorization code security violation is detected.	1 per system
atd-qcalls	AQC	Attendant Queue Calls (display button): tracks the number of calls in the attendant group's queue and displays the queue status. Assign this button to any user who you want to backup the attendant.	1 per station

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Table 2: Telephone feature buttons *4 of 21*

Button name	Button label	Description	Maximum
atd-qtime	AQT	Attendant Queue Time (display button): tracks the calls in the attendant group's queue according to the oldest time a call has been queued, and obtains a display of the queue status.	1 per station
audix-rec	Audix Record	Audix One-Step Recording (display button): activates/deactivates recording of the current call. An Audix hunt group extension that is valid for the user must be entered in the **Ext:** field after the name.	1 per station
aut-msg-wt (Ext: ___)	Message (name or ext #)	Automatic Message Waiting: associated status lamp automatically lights when an LWC message has been stored in the system for the associated extension (can be a VDN). This lamp will not light on the mapped-to physical station for messages left for virtual extensions.	1 per aut-mst-ext
auto-cback	Auto CallBack	Automatic Call Back: when activated, allows inside user who placed a call to a busy or unanswered telephone to be called back automatically when the called telephone becomes available to receive a call.	1 per station
auto-icom (Group: __)	Auto (name or ext #)	Automatic Intercom: places a call to the station associated with the button. The called user receives a unique alerting signal, and a status lamp associated with a Intercom button flashes. Grp: Intercom — Auto-Icom group number. This extension and destination extension must be in the same group.	1 per group per dial code
auto-in (Group: __)	Auto In	Auto-In Mode: allows the user to become automatically available for new ACD calls upon completion of an ACD call. Grp: The split group number for ACD.	1 per split group

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Table 2: Telephone feature buttons *5 of 21*

Button name	Button label	Description	Maximum
auto-wkup	Auto Wakeup	Automatic Wakeup (*display button*): allows attendants, front-desk users, and guests to request a wakeup call to be placed automatically to a certain extension (may not be a VDN extension) at a later time.	1 per station
autodial	Autodial	Allows a user to dial a number that is not part of a stored list.	
aux-work (Group: __)	Auxiliary Work	Auxiliary Work Mode: removes agent from ACD call distribution in order to complete non-ACD-related activities. Grp: The split group number for ACD.	1 per split group
brdg-appr (Btn: __ Ext: ___)	(extension)	Bridged Call Appearance: provides an appearance of another user's extension on this telephone. For example, an assistant might have a bridged appearance of their supervisor's extension. The bridged appearance button functions exactly like the original call appearance, for instance it indicates when the appearance is active or ringing. You can assign brdg-appr buttons only to 2-lamp appearance buttons. You must indicate which extension and which call appearance button the user wants to monitor at this telephone.	Depends on station type

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Table 2: Telephone feature buttons *6 of 21*

Button name	Button label	Description	Maximum
btn-view	Button View	Button View: allows users to view, on the telephone's display, the contents of any feature button. Button View does more than the "View" or "stored-num" feature button; these only display what is contained in abbreviated dialing and autodial buttons. When the user presses the btn-view button and then a specific feature button, they see the feature name and any auxiliary data for that button. This allows users to review the programming of their feature buttons. You can assign this soft-key button to any 6400-, 7400-, or 8400-series display telephone.	
busy-ind (TAC/Ext: __)	Busy	Busy Indication: indicates the busy or idle status of an extension, trunk group, terminating extension group (TEG), hunt group, or loudspeaker paging zone. Users can press the busy-ind button to dial the specified extension. You can assign this button to any lamp button and must specify which Trunk or extension the user wants to monitor.	1 per TAC/ Ext
call-appr	*extension*	Call Appearance: originates or receives calls. Assign to a 2-lamp appearance button.	Depends on station type
call-disp	Return Call	Call Displayed Number (*display button*): initiates a call to the currently displayed number. The number may be from a leave word calling message or a number the user retrieved from the Directory.	1 per station
call-fwd (Ext: ___)	Call Forwarding	Activates or deactivates Call Forwarding All Calls.	1 per station
call-park	Call Park	Allows the user to place the current call in the call park state so it can be retrieved from another telephone.	1 per station

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Table 2: Telephone feature buttons *7 of 21*

Button name	Button label	Description	Maximum
call-pkup	Call Pickup	Allows the user to answer a call that is ringing in the user's pickup group.	1 per station
call-timer	CTime	Used only on the 6400 sets. Allows users to view the duration of the call associated with the active call appearance button.	1 per station
callr-info	Caller Info	(*display button*) Used with Call Prompting to allow users to display information collected from the originator.	1 per station
cas-backup	CAS Backup	Centralized Attendant Service Backup: used to redirect all CAS calls to a backup extension in the local branch if all RLTs are out-of-service or maintenance busy. The associated status lamp indicates if CAS is in the backup mode.	1 per station
cdr1-alrm	CDR 1 Failure	CDR Alarm: associated status lamp is used to indicate that a failure in the interface to the primary CDR output device has occurred.	1 per station
cdr2-alrm	CDR 2 Failure	CDR Alarm: associated status lamp is used to indicate that a failure in the interface to the secondary CDR output device has occurred.	1 per station
cfwd-bsyda	Call Forwarding bsyda (Ext)	Call Forward Busy/Don't Answer: activates and deactivates call forwarding for calls when the extension is busy or the user does not answer.	
check-in	Check In	Check In (*display button*): changes the state of the associated guest room to occupied and turns off the outward calling restriction for the guest room's station.	1 per station

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Table 2: Telephone feature buttons *8 of 21*

Button name	Button label	Description	Maximum
check-out	Check Out	Check Out (*display button*): Changes the state of the associated guest room to vacant and turns on the outward calling restriction for the guest room's station. Also clears (removes) any wake-up request for the station.	1 per station
clk-overid	Clocked Override	Clocked Manual Override (*display button*): used in association with Time of Day Routing to override the routing plan in effect for the activating user. The routing plan is overridden for a specified period of time.	1 per station
conf-dsp	Conferenc e Display	Allows a user to display information about each party of a conference call. This button can be assigned to stations and attendant consoles.	1 per station
consult	Consult	The Consult button allows a covering user, after answering a coverage call, to call the principal (called party) for private consultation. Activating Consult places the caller on hold and establishes a private connection between the principal and the covering user. The covering user can then add the caller to the conversation, transfer the call to the principal, or return to the caller.	1 per station
cov-cback	Coverage Callback	Allows a covering party to store a leave word calling message for the principal (called party).	1 per station
cov-msg-rt	Covr Msg Retrieve	Coverage Message Retrieval (*display button*): places a covering station into the message retrieval mode for the purposes of retrieving messages for the group.	1 per station
cpn-blk	CPN Block	Blocks the sending of the calling party number for a call.	1 per station
cpn-unblk	CPN Unblock	Deactivates calling party number (CPN) blocking and allows the CPN to be sent for a single call.	1 per station

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Table 2: Telephone feature buttons *9 of 21*

Button name	Button label	Description	Maximum
crss-alert	Crisis	Crisis Alert (*display button*): provide this button to the telephones or consoles that you want to notify when any user makes an emergency call. (You define which calls are emergency calls on the **AAR/ARS Analysis** screen by setting the **Call Type** to **alrt**.) After a user receives an alert, they can press the crss-alert button to disable the current alert. If tenant partitioning is active, the attendants within a partition can receive emergency notification only from callers in the same partition.	1 per station 10 per system
data-ext	Data (data ext #)	Data Extension: sets up a data call. May be used to pre-indicate a data call or to disconnect a data call. May not be a VDN or ISDN-BRI extension.	1 per data-extension group
date-time	Date Time	Date and Time (*display button*): displays the current date and time. Do not assign this button to 6400-series display phones as they normally show the date and time.	1 per station
delete-msg	Delete Message	Delete message (*display button*): deletes a stored message that is currently on the display.	1 per station
dial-icom (Grp: ___)	Dial Icom	Dial Intercom: accesses the intercom group assigned to the button. Grp: Intercom — Dial (Dial Icom) group number.	1 per group
did-remove	DID Remove	DID Remove (*display button*): allows DID assignments to be removed.	1 per station
did-view	DID View	DID View (*display button*): allows DID assignments to be displayed and changed. Allows choice between XDID and XDIDVIP numbers.	1 per station

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Table 2: Telephone feature buttons *10 of 21*

Button name	Button label	Description	Maximum
directory	Directory	Directory (*display button*): allows users with display telephones to access the system directory, use the touch-tone buttons to key in a name, and retrieve an extension from the directory. The directory contains the names and extensions that you have assigned to the telephones administered in your system. If you assign a directory button, you should also assign a Next and Call-Disp button to the telephone. These buttons allow the user to navigate within the directory and call an extension once they find the correct one. Note that Vector Directory Numbers do not appear in the Directory.	1 per station
dir-pkup	dir-pkup	Directed call pickup: allows the user to answer a call ringing at another extension without having to be a member of a pickup group.	
disp-chrg	Display Charge	Provides your display telephone with a visual display of accumulated charges on your current telephone call. Used exclusively outside the U.S. and Canada.	1 per station
disp-norm	Local/ Normal	Normal (*display button*): Toggles between LOCAL display mode (displays time and date) and NORMAL mode (displays call-related data). LED off = LOCAL mode and LED on = NORMAL.	1 per station
dn-dst	Do Not Disturb	Places the user in the do not disturb mode.	1 per station
drop	Drop	Allows users to drop calls. Users can drop calls from automatic hold or drop the last party they added to a conference call.	

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Table 2: Telephone feature buttons *11 of 21*

Button name	Button label	Description	Maximum
exclusion	Exclusion	Exclusion: allows multiappearance telephone users to keep other users with appearances of the same extension from bridging onto an existing call. If the user presses the **Exclusion** button while other users are already bridged onto the call, the other users are dropped. There are two means of activating exclusion. • Manual Exclusion — when the user presses the **Exclusion** button (either before dialing or during the call). • Automatic Exclusion — as soon as the user picks up the handset. To turn off Automatic Exclusion during a call, the user presses the **Exclusion** button. To use Automatic Exclusion, set the **Automatic Exclusion by COS** field to **y** on the **Feature-Related System Parameters** screen.	1 per station
ext-dn-dst	Do Not Disturb Ext	**Extension — Do Not Disturb** (*display button*): used by the attendant console or hotel front desk display telephone to activate do not disturb and assign a corresponding deactivate time to an extension.	1 per station
fe-mute	Far End Mute for Conf	Allows a user to mute a selected party on a conference call. This button can be assigned to stations and attendant consoles.	1 per station
			11 of 21

Table 2: Telephone feature buttons *12 of 21*

Button name	Button label	Description	Maximum
flash	Flash	1) Allows a station on a trunk call with Trunk Flash to send a Trunk Flash signal to the far end (e.g., Central Office); 2) allows a station on a CAS main call to send a Trunk Flash signal over the connected RLT trunk back to the branch to conference or transfer the call.	1 per station
goto-cover	Go To Cover	Go To Coverage: sends a call directly to coverage instead of waiting for the called inside-user to answer. Go to Cover forces intercom and priority calls to follow a coverage path. NOTE: Go to Cover cannot be activated for calls placed to a Vector Directory Number extension. Go to Cover can be used to force a call to cover to a VDN if the called principal has a VDN as a coverage point.	1 per station
grp-dn-dst	Do Not Disturb Grp	Group Do Not Disturb (*display button*): places a group of phones into the do not disturb mode.	1 per station
grp-page (Number:___)	GrpPg	Allows users to make announcements to groups of stations by automatically turning on their speakerphones. Number: The extension of the page group.	
headset	Headset	Signals onhook/offhook state changes to Communication Manager. The green LED is on for offhook state and off (dark) for onhook state.	
hunt-ns (Grp: ___)	Hunt Group	Hunt-Group Night Service: places a hunt-group into night service. Grp: Hunt group number.	3 per hunt group
			12 of 21

Table 2: Telephone feature buttons *13 of 21*

Button name	Button label	Description	Maximum
in-call-id (Type: __ Grp: ___)	Coverage (group #, type, name, or ext #)	The Coverage Incoming Call Identification (ICI) button allows a member of a coverage answer group or hunt group to identify an incoming call to that group even though the member does not have a display telephone. In the **Type** field, enter **c** for coverage answer groups and type of **h** for a hunt group. In the **Grp** field, enter the group number.	1 per group-type per group
inspect	Inspect Mode	Inspect (*display button*): allows users on an active call to display the identification of an incoming call. Inspect also allows users to determine the identification of calls they placed on Hold.	1 per station
int-aut-an	IntAutoAns	Internal Automatic Answer: causes any hybrid or digital station to automatically answer incoming internal calls.	1 per station
last-numb	LastNumb Dialed	Last Number Dialed (redial): originates a call to the number last dialed by the station.	1 per station
lic-error	License Error	License-Error: indicates a major License File alarm. Pressing the button does not make the light go out. The button goes out only after the error is cleared and Communication Manager returns to License-Normal Mode. You can administer this button on phones and attendant consoles.	1 per telephone 20 per system (Server CSI/ Server SI) 30 per system (Server R)
link-alarm (link# ___)	Link Failure (link #)	Link Alarm: associated status lamp indicates that a failure has occurred on one of the Processor Interface circuit pack data links. Link: Link number — 1 to 8 for multi-carrier cabinets or 1 to 4 for single-carrier cabinets.	8 per station

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Table 2: Telephone feature buttons *14 of 21*

Button name	Button label	Description	Maximum
lsvn-halt	Login SVN	Login Security Violation Notification: activates or deactivates referral call when a login security violation is detected.	1 per system
lwc-cancel	Cancel LWC	Leave Word Calling Cancel: cancels the last leave word calling message originated by the user.	1 per station
lwc-lock	Lock LWC	Leave Word Calling Lock: locks the message retrieval capability of the display module on the station.	1 per station
lwc-store	LWC	Leave Word Calling Store: leaves a message for the user associated with the last number dialed to return the call to the originator.	1 per station
major-alrm	Major Hdwe Failure	Major Alarm: assign to a status lamp to notify the user when major alarms occur. Major alarms usually require immediate attention.	1 per station
man-msg-wt (Ext: ___)	Msg Wait (name or ext #)	Manual Message Waiting: allows a multiappearance telephone user to press a button on their telephone in order to light the Manual Message Waiting button at another telephone. You can administer this feature only to pairs of telephones, such as an assistant and an executive. For example, an assistant can press the man-msg-wt button to signal the executive that they have a call.	None
man-overid (TOD: _)	Immediate Override	Immediate Manual Override (*display button*): allows the user (on a system with Time of Day Routing) to temporarily override the routing plan and use the specified TOD routing plan. TOD: specify the routing plan the user wants to follow in override situations.	1 per station

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Table 2: Telephone feature buttons *15 of 21*

Button name	Button label	Description	Maximum
manual-in (Group: __)	Manual In	Manual-In Mode: prevents the user from becoming available for new ACD calls upon completion of an ACD call by automatically placing the agent in the after call work mode. Grp: The split group number for ACD.	1 per split group
mct-act	MCT Activation	Malicious Call Trace Activation: sends a message to the MCT control extensions that the user wants to trace a malicious call. MCT activation also starts recording the call, if your system has a MCT voice recorder.	1 per station
mct-contr	MCT Control	Malicious Call Trace Control: allows the user to take control of a malicious call trace request. Once the user becomes the MCT controller, the system stops notifying other MCT control extensions of the MCT request. NOTE: To add an extension to the MCT control group, you must also add the extension on the **Extensions Administered to have an MCT-Control Button** screen. When the user presses the MCT Control button, the system first displays the called party information. Pressing the button again displays the rest of the trace information. The MCT controller must dial the MCT Deactivate feature access code to release control.	1 per station
mf-da-intl	Directory Assistance	Multifrequency Operator International: allows users to call Directory Assistance.	1 per station
mf-op-intl	CO attendant	Multifrequency Operator International: allows users to make international calls to the CO attendant.	1 per station

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Table 2: Telephone feature buttons *16 of 21*

Button name	Button label	Description	Maximum
mj/mn-alrm	Maj/Minor Hdwe Failure	Minor Alarm: assign to a status lamp to notify the user when minor or major alarms occur. Minor alarms usually indicate that only a few trunks or a few stations are affected.	1 per station
mm-basic	MM Basic	Multimedia Basic: used to place a multimedia complex into the "Basic" mode or to return it to the "Enhanced" mode.	1 per station
mm-call	MM Call	Multimedia Call: used to indicate a call is to be a multimedia call.	1 per station
mm-cfwd	MM CallFwd	Multimedia Call Forward: used to activate forwarding of multimedia calls as multimedia calls, not as voice calls.	1 per station
mm-datacnf	MM Datacnf	Multimedia Data Conference: used to initiate a data collaboration session between multimedia endpoints; requires a button with a lamp.	1 per station
mm-multnbr	MM MultNbr	Indicate that the user wants to place calls to 2 different addresses using the 2 B-channels.	1 per station
mm-pcaudio	MM PCAudio	Switches the audio path from the telephone (handset or speakerphone) to the PC (headset or speakers/ microphone).	1 per station
msg-retr	Message Retrieve	Message Retrieval (*display button*): places the station’s display into the message retrieval mode.	1 per station
mwn-act	Message Waiting Act.	Message Waiting Activation: lights a message waiting lamp on an associated station.	1 per station
mwn-deact	Message Waiting Deact	Message Waiting Deactivation: dims a message waiting lamp on an associated station.	1 per station
			16 of 21

Table 2: Telephone feature buttons *17 of 21*

Button name	Button label	Description	Maximum
next	Next	Next (*display button*): steps to the next message when the telephone's display is in Message Retrieval or Coverage Message Retrieval mode. Shows the next name when the telephone's display is in the Directory mode.	1 per station
night-serv	Night Serv	Night Service Activation: toggles the system in or out of Night Service mode.	1 per station
noans-alrt	RONA	Redirection on No Answer Alert: indicates a Redirection on No Answer timeout has occurred for the split.	1 per hunt group
no-hld-cnf	No Hold Conferenc e	No Hold Conference: can automatically conference another party while continuing the existing call.	1 per station
normal	Normal Mode	Normal (*display button*): places the station's display into normal call identification mode.	1 per station
off-bd-alm	Off board alarm	Off board Alarm: associated status lamp lights if an off-circuit pack major, minor, or warning alarm is active on a circuit pack. Off-board alarms (loss of signal, slips, misframes) relate to problems on the facility side of the DS1, ATM, or other interface.	1 per attendant
per-COline (Grp: ___)	CO Line (line #)	Personal CO Line: allows the user to receive calls directly via a specific trunk. Grp: CO line group number.	1 per group
pms-alarm	PMS Failure	Property Management System alarm: associated status lamp indicates that a failure in the PMS link occurred. A major or minor alarm condition raises the alarm.	1 per station
pr-awu-alm	Auto Wakeup Alm	Automatic Wakeup Printer Alarm: associated status lamp indicates that an automatic wakeup printer interface failure occurred.	1 per station

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Table 2: Telephone feature buttons *18 of 21*

Button name	Button label	Description	Maximum
pr-pms-alm	PMS Ptr Alarm	PMS Printer Alarm: associated status lamp indicates that a PMS printer interface failure occurred.	1 per station
pr-sys-alm	Sys Ptr Alarm	System Printer Alarm: associated status lamp indicates that a system printer failure occurred.	1 per station
print-msgs	Print Msgs	Print Messages: allows users to print messages for any extension by pressing the button and entering the extension and a security code.	1 per station
priority	Priority Call	Priority Calling: allows a user to place priority calls or change an existing call to a priority call.	1 per station
q-calls (Grp: ___)	NQC	Queue Calls: associated status lamp flashes if a call warning threshold has been reached. Grp: Group number of hunt group.	1 per hunt group per station
q-time (Grp: ___)	OQT	Queue Time: associated status lamp flashes if a time warning threshold has been reached. Grp: Group number of hunt group.	1 per hunt group per station
release	Release	Releases an agent from an ACD call.	1 per station
ringer-off	Ringer Cutoff	Ringer-Cutoff: silences the alerting ringer on the station.	1 per station
rs-alert	System Reset Alert	The associated status lamp lights if a problem escalates beyond a warm start.	1 per station
rsvn-halt	rsvn-halt	Remote Access Barrier Code Security Violation Notification Call: activates or deactivates call referral when a remote access barrier code security violation is detected.	1 per system
scroll	Scroll	Scroll (*display button*): allows the user to select one of the two lines (alternates with each press) of the 16-character LCD display. Only one line displays at a time.	1 per station

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Table 2: Telephone feature buttons *19 of 21*

Button name	Button label	Description	Maximum
send-calls (Ext: ___)	Send All Calls	Send All Calls allows users to temporarily direct all incoming calls to coverage regardless of the assigned call-coverage redirection criteria. Assign to a lamp button.	1 per station
send-term	Send All Calls-TEG	Send All Calls For Terminating Extension Group: allows the user to forward all calls directed to a terminating extension group.	1 per TEG
serv-obsrv	Service Observing	Service Observing: activates Service Observing. Used to toggle between a listen-only and a listen-talk mode.	1 per station
signal (Ext: ___)	Signal (name or ext #)	Signal: allows the user to use one button to manually signal the associated extension. The extension cannot be a VDN extension.	1 per signal extension
ssvn-halt	ssvn-halt	Toggle whether or not station security code violation referrals are made to the referral destination.	1 per station
sta-lock	Station Lock	Station Lock: when enabled, no outgoing calls can be made from the telephone.	1 per station
stored-num	Stored Number	(*display button*) Places the station's display into the stored number mode.	1 per station
stroke-cnt (Code:_)	Stroke Count (#)	Automatic Call Distribution Stroke Count # (0, 1, 2, 3, 4, 5, 6, 7, 8, or 9) sends a message to CMS to increment a stroke count number.	up to 10 per station
term-x-gr (Grp: ___)	Term Grp (name or ext #)	Terminating Extension Group: provides one or more extensions. Calls may be received but not originated with this button. Grp: TEG number.	1 per TEG
timer	Timer	Used only on the 6400 sets. Allows users to view the duration of the call associated with the active call appearance button.	1 per station

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Table 2: Telephone feature buttons *20 of 21*

Button name	Button label	Description	Maximum
togle-swap	Conf/Trans Toggle-Swap	Allows a user to toggle between two parties before completing a conference or a transfer.	1 per station
trk-ac-alm	FTC Alarm	Facility Test Call Alarm: associated status lamp lights when a successful Facility Test Call (FTC) occurs.	1 per station
trk-id	Trunk ID	Trunk Identification (*display button*): identifies the tac (trunk access code) and trunk member number associated with a call.	1 per station
trunk-name	Trunk Name	(*display button*) Displays the name of the trunk as administered on the CAS Main or on a server without CAS.	1 per station
trunk-ns (Grp: ___)	Trunk Grp	Trunk-Group Night Service: places a trunk-group into night service. Grp: Trunk group number.	3 per trunk group
verify	Verify	Busy Verification: allows users to make test calls and verify a station or a trunk.	1 per station
vip-chkin	VIP Check-in	VIP Check-in (*display button*): allows user to assign the XDIDVIP number to the room extension.	1 per station
vip-retry	VIP Retry	VIP Retry: starts to flash when the user places a VIP wakeup call and continues to flash until the call is answered. If the VIP wakeup call is not answered, the user can press the VIP Retry button to drop the call and reschedule the VIP wakeup call as a classic wakeup call. To assign this button, you must have both Hospitality and VIP Wakeup enabled.	1 per station
vip-wakeup	VIP Wakeup	VIP Wakeup: flashes when a VIP wakeup reminder call is generated. The user presses the button to place a priority (VIP) wakeup call to a guest. To assign this button, you must have both Hospitality and VIP Wakeup enabled.	1 per station

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Table 2: Telephone feature buttons *21 of 21*

Button name	Button label	Description	Maximum
voa-repeat	VOA repeat	VDN of Origin Announcement. VDN of Origin Announcement must be enabled.	1 per station
vu-display (format: __ ID: __)	VuStats #	VuStats Display: allows the agent to specify a display format for the statistics. If you assign a different VuStats display format to each button, the agent can use the buttons to access different statistics. You can assign this button only to display phones. format: specify the number of the format you want the button to display ID (optional): specify a split number, trunk group number, agent extension, or VDN extension	limited to the number of feature buttons on the telephone
whisp-act	Whisper Page Activation	Whisper Page Activation: allows a user to make and receive whisper pages. A whisper page is an announcement sent to another extension who is active on a call where only the person on the extension hears the announcement; any other parties on the call cannot hear the announcement. The user must have a class of restriction (COR) that allows intraswitch calling to use whisper paging.	1 per station
whisp-anbk	Answerback	Whisper Page Answerback: allows a user who received a whisper page to respond to the user who sent the page.	1 per station
whisp-off	Whisper Page Off	Deactivate Whisper Paging: blocks other users from sending whisper pages to this telephone.	1 per station
work-code	Work Code	Call Work Code: allows an ACD agent after pressing "work-code" to send up to 16 digits (using the dial pad) to CMS.	1 per station

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Adding Abbreviated Dialing Lists

Abbreviated dialing is sometimes called speed dialing. It allows you to dial a short code in place of an extension or telephone number.

When you dial abbreviated-dialing codes or press abbreviated-dialing buttons, you access stored numbers from special lists. These lists can be personal (a list of numbers for an individual telephone), group (a department-wide list), system (a system-wide list), or enhanced numbers (allows for a longer list of numbers). The version and type of your system determine which lists are available and how many entries you can have on each list.

Note:

You can designate all group-number lists, system-number lists, and enhanced-number lists as "privileged." Calls automatically dialed from a privileged list are completed without class of restriction (COR) or facility restriction level (FRL) checking. This allows access to selected numbers that some telephone users might otherwise be restricted from manually dialing. For example, a user may be restricted from making long-distance calls. However, you can program the number of a branch office that is long distance into an AD list as privileged. Then, the user can call this office location using AD, while still being restricted from making other long-distance calls.

SECURITY ALERT:

Privileged group-number, system-number, and enhanced-number lists provide access to numbers that typically would be restricted.

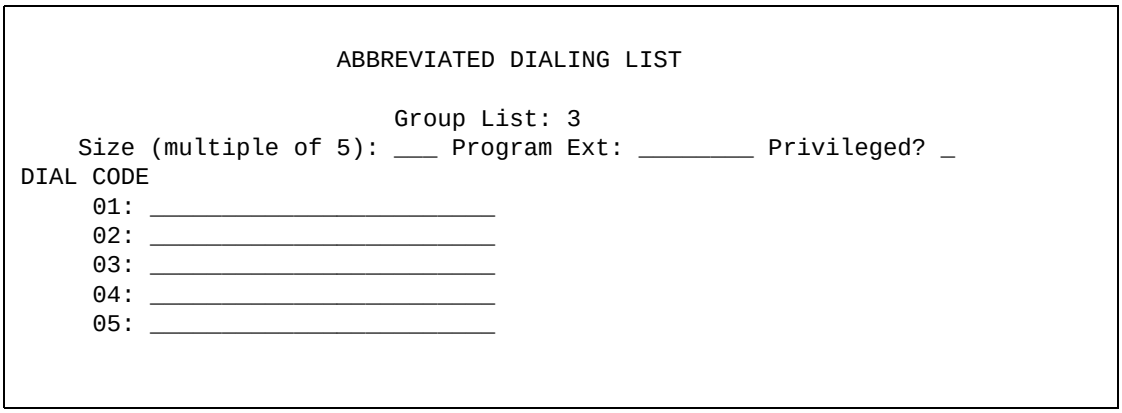
Instructions

As an example, we will program a new group list:

1. Type **`add abbreviated-dialing group next`** and press **Enter**.

   The **Abbreviated Dialing List** screen appears. In our example, the next available group list is group 3.

Figure 42: Abbreviated Dialing List screen



2. Enter a number (in multiples of 5) in the **Size** field. This number defines the number of entries on your dialing list.

   For example, if you have 8 telephone numbers you want to store in the list, type **10** in the **Size** field.

3. If you want another user to be able to add numbers to this list, enter their extension in the **Program Ext** field.

   For example, if you want the user at 4567 to be able to change group list 3, enter **4567** in this field.

4. Enter the telephone numbers you want to store, one for each dial code.

   Each telephone number can be up to 24 digits long.

5. Press **Enter** to save your changes.

You can display your new abbreviated-dialing list to verify that the information is correct or print a copy of the list for your paper records.

Once you define a group list, you need to define which stations can use the list. For example, we will set up station 4567 so it has access to the new group list.

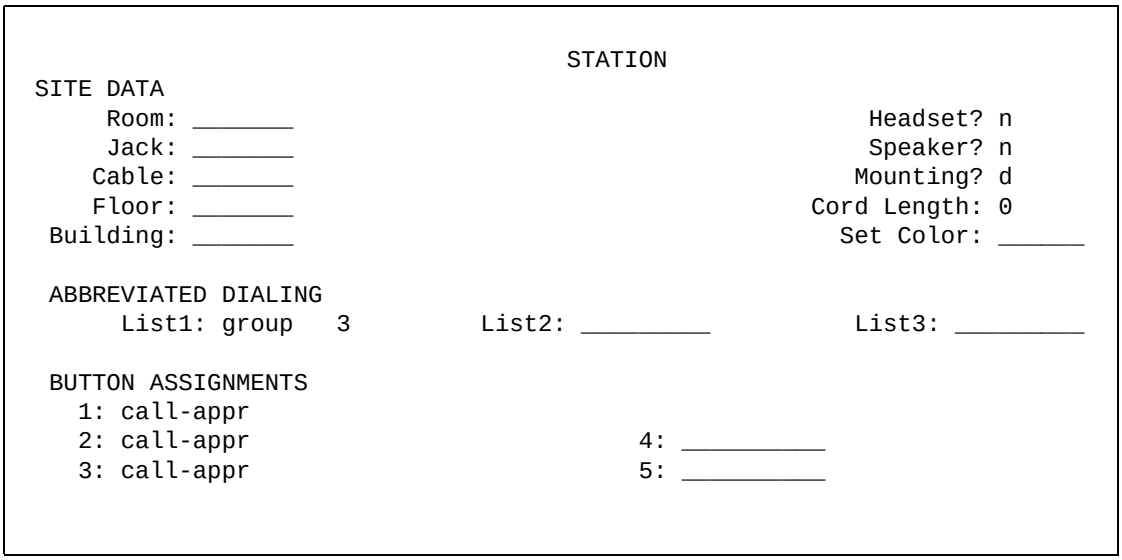
To give station 4567 access to the group list:

1. Type **`change station 4567`** and press **Enter**.

   The **Station** screen for extension 4567 appears.

2. Press **Next Page** to access the screen containing the Abbreviated Dialing List fields.

Figure 43: Station screen



```
                                    STATION
SITE DATA
     Room: _______                                    Headset? n
     Jack: _______                                    Speaker? n
    Cable: _______                                   Mounting? d
    Floor: _______                                 Cord Length: 0
 Building: _______                                   Set Color: ______

 ABBREVIATED DIALING
      List1: group   3        List2: _________          List3: _________

 BUTTON ASSIGNMENTS
   1: call-appr
   2: call-appr                            4: __________
   3: call-appr                            5: __________
```

3. Type **group** in any of the **List** fields and press **Enter**.

   A blank list number field appears.

4. Type **3** in the list number field.

   When you assign a group or personal list, you must also specify the personal list number or group list number.

5. Press **Enter** to save your changes.

The user at extension 4567 can now use this list by dialing the feature access code for the list and the dial code for the number they want to dial. Alternatively, you can assign an abbreviated dialing button to this station that allows the user press one button to dial a specific stored number on one of their three assigned abbreviated lists.

Fixing problems

Problem	Possible causes	Solutions
A user cannot access a dial list	• The specific list may not be assigned to the user's telephone.	Resolution: 1. Type **display station nnnn**, where ***nnnn*** is the user's extension. Press **Enter**. 2. Review the current settings of the **List1**, **List2**, and **List3** fields to determine if the list the user wants to access is assigned to their telephone.
	• If the user attempted to use a feature access code to access the list, they may have dialed the incorrect feature access code.	Resolution: 1. Type **display feature-access-codes** and press **Enter**. 2. Verify that the user is dialing the appropriate feature access code.
	• If the user attempted to press a feature button, they may have pressed the incorrect feature button.	Resolution: 1. Type **display station nnnn**, where ***nnnn*** is the user's extension. Press **Enter**. 2. Review the current feature button assignments to determine if the user was pressing the assigned button.

1 of 2

Problem	Possible causes	Solutions
A user cannot access a dial list - *continued*	• If the user attempted to press the correct feature button, the button may not be set up correctly.	Resolution: 1. Type **display station nnnn**, where ***nnnn*** is the user's extension. Press **Enter**. 2. Review the current feature button assignments to see if the list number and dial code are correct.
A user complains that using an abbreviated dial list dials the wrong number.	• The user could be using the wrong dial code. • The dial code could be defined incorrectly.	Resolution: 1. Ask the user what number they dialed or button they pressed to determine which list and dial code they attempted to call. 2. Access the dialing list and verify that the number stored for the specific dial code corresponds to the number the user wanted to dial. (For example to access a group list, type **display abbreviated-dialing group *x*** and press **Enter**, where ***x*** is a group list number.) 3. If the user dialed the wrong code, give them the correct code. 4. If the dial code is wrong, press **Cancel** and use the appropriate change command to re-access the abbreviated dialing list. Correct the number and press **Enter**.
		2 of 2

More information

There are limits to the total number of abbreviated dialing list entries, the number of personal dial lists, and the number of group dial lists that your system can store. Because of these limitations, you should avoid storing the same number in more than one list. Instead, assign commonly dialed numbers to the system list or to a group list. You can determine the abbreviated dialing storage capacity, by referring to the **System Capacity** screen for the abbreviated dialing values (type **`display capacity`**). For details on the **System Capacity** screen, see *Maintenance Commands for Avaya Communication Manager, Media Gateways and Servers*, 03-300191.

Related topics

For more information, see "Abbreviated Dialing" in *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205.

Setting up Bridged Call Appearances

Think of a bridged call appearance as a telephone (the primary set) with an extension (the bridged-to appearance). Both phones can be used to call in and out and both show when a line is in use. A call to the primary telephone is bridged to a specific appearance, or button, on the secondary telephone. The secondary telephone retains all its functions, and a specific button is dedicated as the bridged-to appearance from the primary telephone.

Bridged call appearances have to be assigned to phones with double-lamp buttons, or lights. The telephone types do not need to match, but as much consistency as possible is recommended for all phones in a bridged group. When a call comes in on bridged phones, the buttons assigned to the bridged appearances flash. You can assign as many bridged appearances as there are line appearances on the primary telephone, and you can assign ringing (alerting) to one or more of the phones.

Instructions

To create a bridged call appearance:

1. Note the extension of the primary telephone.

   A call to this telephone lights the button and, if activated, rings at the bridged-to appearance on the secondary telephone.

2. If you want to use a new telephone for the bridged-to extension, duplicate a station.

   For information, see Using Templates to Add Phones on page 83.

3. Type **change station** and the bridged-to extension and press **Enter**.

   The **Station** screen appears.

Figure 44: Station screen

```
change station 75001                                          Page 2 of X
                                    STATION

FEATURE OPTIONS
           LWC Reception? spe                 Auto Select Any Idle Appearance? n
          LWC Activation? y                           Coverage Msg Retrieval? y
  LWC Log External Calls? n                                   Auto Answer: none
             CDR Privacy? n                               Data Restriction? n
   Redirect Notification? y                       Idle Appearance Preference? n
 Per Button Ring Control? n
   Bridged Call Alerting? n                         Restrict Last Appearance? y
  Active Station Ringing: single

        H.320 Conversion? n             Per Station CPN - Send Calling Number? _
       Service Link Mode: as-needed
         Multimedia Mode: basic
    MWI Served User Type: ______                    Display Client Redirection? n
         Automatic Moves:
              AUDIX Name:                         Select Last Used Appearance? n
                                 _                  Coverage After Forwarding? _
     Recall Rotary Digit? n                           Multimedia Early Answer? n

Remote Softphone Emergency Calls: as-on-local   Direct IP-IP Audio Connections? n
Emergency Location Ext: 75001                           IP Audio Hairpinning? n
```

4. Press **Next Page** until **Per Button Ring Control** appears (digital sets only).
   - If you want to assign ringing separately to each bridged appearance, type **y**.
   - If you want all bridged appearances to either ring or not ring, leave the default **n**.
5. Move to **Bridge Call Alerting**.

   If you want the bridged appearance to ring when a call arrives at the primary telephone, type **y**. Otherwise, leave the default **n**.

6. Complete the appropriate field for your telephone type.

If. . .	Then. . .
your primary telephone is analog	move to the **Line Appearance** field and enter **abrdg-appr**
your primary telephone is digital	move to the **Button Assignments** field and enter **brdg-appr**

7. Press **Enter**.

   Btn and **Ext** fields appear. If **Per Button Ring Control** is set to **y** on the digital screen, **Btn**, **Ext**, and **Ring** fields appear.

Figure 45: Station screen (analog set)

```
                                    STATION
 SITE DATA
      Room: _______                                      Headset? n
      Jack: _______                                      Speaker? n
     Cable: _______                                     Mounting? d
     Floor: _______                                   Cord Length: 0
  Building: _______                                     Set Color: ______

ABBREVIATED DIALING
     List1: ___________       List2: _________          List3: _________

HOT LINE DESTINATION

         Abbreviated Dialing List Number (From above 1, 2 or 3):
                                                        Dial Code:
      Line Appearance:  brdg-appr   Btn:     Ext:
```

Figure 46: Station screen (digital set)

```
                                    STATION
 SITE DATA
       Room: _______                                    Headset? n
       Jack: _____                                      Speaker? n
      Cable: _____                                     Mounting: d
      Floor: _______                                Cord Length: 0
   Building: _______                                  Set Color: ______


ABBREVIATED DIALING
     List1: ________           List2: ________           List3: ________

BUTTON ASSIGNMENTS
  1: brdg-appr  Btn:   Ext:      Ring:
  1: brdg-appr  Btn:   Ext:      Ring:
```

8. Enter the primary telephone's button number that you want to assign as the bridged call appearance.

   This button flashes when a call arrives at the primary telephone.

9. Enter the primary telephone extension.
10. If the **Ring** field appears:
    - If you want the bridged appearance to ring when a call arrives at the primary telephone, type **y**.
    - If you do not want the bridged appearance to ring, leave the default **n**.
11. Press **Enter** to save your changes.

To see if an extension has any bridged call appearances assigned, type **`list bridge`** and the extension, and press **Enter**.

More information

Following are a list of example situations where you might want to use bridged appearances.

- A secretary making or answering calls on an executive's primary extension

  These calls can be placed on hold for later retrieval by the executive, or the executive can simply bridge onto the call. In all cases, the executive handles the call as if he or she had placed or answered the call. It is never necessary to transfer the call to the executive.

- Visitor telephones

  An executive may have another telephone in their office that is to be used by visitors. It may be desirable that the visitor be able to bridge onto a call that is active on the executive's primary extension number. A bridged call appearance makes this possible.

- Service environments

  It may be necessary that several people be able to handle calls to a particular extension number. For example, several users may be required to answer calls to a hot line number in addition to their normal functions. Each user may also be required to bridge onto existing hot line calls. A bridged call appearance provides this capability.

- A user frequently using telephones in different locations

  A user may not spend all of their time in the same place. For this type of user, it is convenient to have their extension number bridged at several different telephones.

Setting up an Extension to Cellular Feature Access Button

Extension to Cellular provides the capability to administer an Extension to Cellular feature access button on the user's office telephone to enable and disable the feature. You can also configure an optional timer. You administer this feature button on page 3 of the **Station** screen for the "host" office extension to which Extension to Cellular is linked.

The process described below explains how to administer an Extension to Cellular feature button and include the optional Extension to Cellular timer. The Extension to Cellular feature button is available on phones which support administrable feature buttons.

To set up an Extension to Cellular feature button with optional timer:

1. Type **`change station`** ***n*** (where ***n*** is the extension of an Extension to Cellular-enabled station - in this example, **`1034`**) and press **Enter**.

   The **Station** screen appears.

2. Press the **Next Page** button twice to display page 3 of the **Station** screen.

Figure 47: Station screen - page 3

```
change station 1034                                        Page   3 of 4
                                    STATION
 SITE DATA
       Room:                                            Headset? n
       Jack:                                            Speaker? n
      Cable:                                           Mounting: d
      Floor:                                       Cord Length: 0
   Building:                                         Set Color:




ABBREVIATED DIALING
     List1:                    List2:                    List3:

BUTTON ASSIGNMENTS
 1: call-appr                            5:
 2: call-appr                            6:
 3: call-appr                            7:
 4: ec500    Timer? y                    8:
```

3. Select an available feature button under the **Button Assignments** header (button **4** was used in this example) and type **ec500** (lower case) to administer an Extension to Cellular feature button on the office telephone.
4. Press **Enter**.

   Note:

   The **Timer** subfield displays, and defaults to **n**. Leaving the default setting of **n** excludes the timer state.

5. Set the optional **Timer** subfield to **y** to include an Extension to Cellular timer state for the administered feature button.

   When the timer state is included, the Extension to Cellular user can activate a one-hour timer to temporarily disable Extension to Cellular through this administered feature button.

6. Press **Enter**.

   The corresponding feature button on the office telephone is now administered for Extension to Cellular.

   Note:

   The feature status button on the office telephone indicates the current state of Extension to Cellular regardless of whether the feature was enabled remotely or directly from the office telephone.

For additional information, see the *Avaya Extension to Cellular Administrator's Guide, 210-100-500.*

Setting up Terminal Self Administration

Terminal self-administration (TSA) allows users to administer some of their own feature buttons from their telephones. TSA is available for 6400-series, and 4612 and 4624 telephones. Users are prompted, via the telephone's display, to choose features to assign to buttons on their telephones.

Before you start

To prevent users from changing another user's telephone administration, you can enable the system-wide option that requires users to enter a station security code before they can administer their telephone. To enable this option:

1. Set the **Station Security Code for Terminal Self-Administration Required** on the **Security-Related System Parameters** screen to **y**.

   If you enable this option, the user is prompted for the station security code when they press the **Admin** button. The user must enter the security code, followed by the pound (#) button or the **Done** softkey.

Instructions

You need to assign a security code to the user's **Station** screen for each user you want to enable access to TSA. You also need to assign the user an Admin feature button.

For example, to assign a security code of 12345678 to extension 4234, complete the following steps:

1. Type **`change station 4234`** and press **Enter**.

   The **Station** screen for extension 4234 appears.

2. In the **Security Code** field, type **12345678**.

   You should assign unique security codes for each user. Once you enter the code and move off the field, the system changes the field to '*' for extra security.

3. In one of feature button fields, type **admin**.

   You can assign this button to a feature button or a softkey.

4. Press **Enter** to save your changes.

More information

Before a user can enter the TSA Admin mode, their telephone must be completely idle. After a user presses the **Admin** button and enters a security code (if necessary), they are prompted, via the telephone's display, to choose features to administer to buttons on their telephone.

The user can add, replace, or delete any of the following feature-button types from their telephone.

- CDR Account Code
- Automatic Dial
- Blank
- Call Forwarding
- Call Park
- Call Pickup
- Directed Call Pickup
- Group Page
- Send All Calls
- Toggle Swap
- Activate Whisper Page
- Answerback for Whisper Page
- Whisper Page Off

End-user button changes are recorded to the Communication Manager server's history log so that remote services can know what translations are changed.

Fixing problems

- When a telephone is in the Admin mode, the telephone cannot accept any calls — the telephone is treated as if it were busy. Also, a user cannot make calls while in the Admin mode.
- Any button state a telephone is in when the telephone enters the Admin mode stays active while the telephone is in the Admin mode.
- ACD agents who wish access to the Admin mode of TSA must be logged off before pressing the Admin button. If they are not logged off when they attempt to enter the Admin mode, they receive a denial (single-beep) tone.
- Call Forwarding can be active and works correctly in the Admin mode. An active Call Forwarding button cannot be removed when the telephone is in the Admin mode.

- Since the telephone must be on-hook to go into the Admin mode, the Headset On/Off button must be in the OFF position.
- A telephone that is in the Admin mode of TSA cannot be remotely unmerged by the PSA feature.

  If a user has Abbreviated and Delayed Ringing active, a call can be silently ringing at a telephone and the user may not realize it. This ringing prevents the user from entering the Admin mode of TSA.

Chapter 5: Managing Attendant Consoles

Attendant Consoles

The attendant console is the main answering position for your organization. The console operator is responsible for answering incoming calls and for efficiently directing or "extending" calls to the appropriate telephone.

The attendant console also can allow your attendants to monitor:

- system problems
- toll fraud abuse
- traffic patterns

The number of consoles you can have in your organization varies depending on your Avaya solution.

302 attendant consoles

Avaya Communication Manager supports the following 302 attendant consoles: the 302A/B, 302C, and 302D consoles. You may have a basic or enhanced version of these consoles. Figure 48 shows the 302A/B console and Figure 49 shows the 302C console. The next two figures show the button layouts on the Feature area and on the optional Selector console.

Figure 48: 302A and 302B1 attendant console

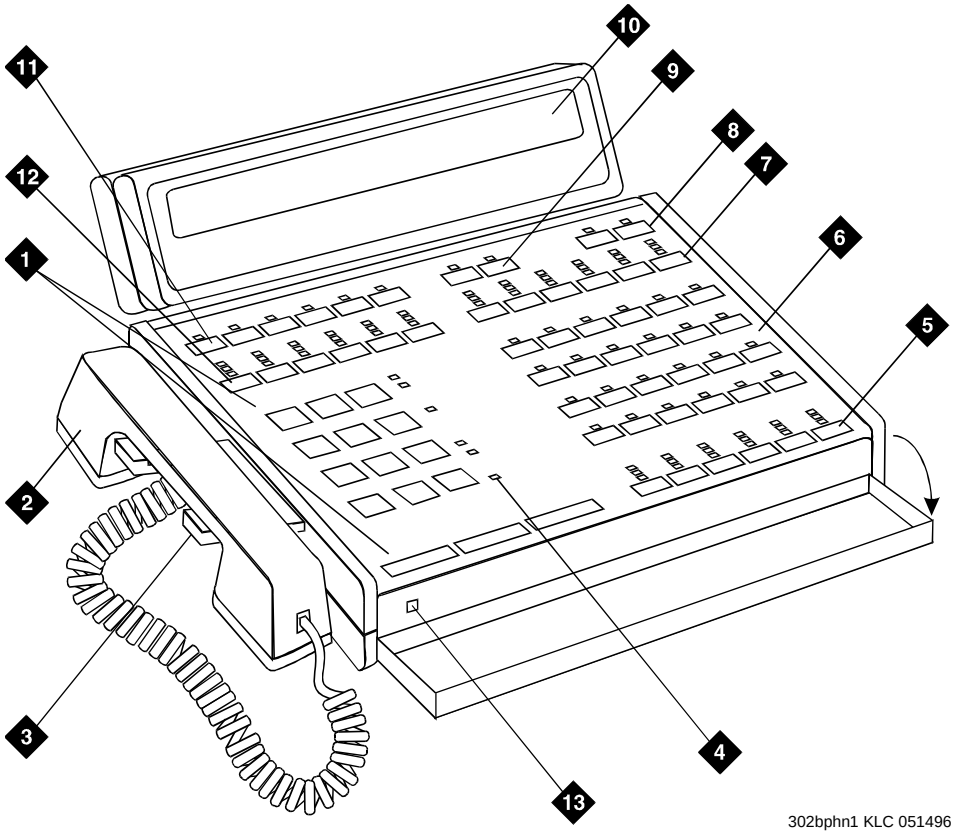


Figure notes:

1. Call processing area
2. Handset
3. Handset cradle
4. Warning lamps and call waiting lamps
5. Call appearance buttons
6. Feature area
7. Trunk group select buttons
8. Volume control buttons
9. Select buttons
10. Console display panel
11. Display buttons
12. Trunk group select buttons
13. Lamp Test Switch

Figure 49: 302C attendant console

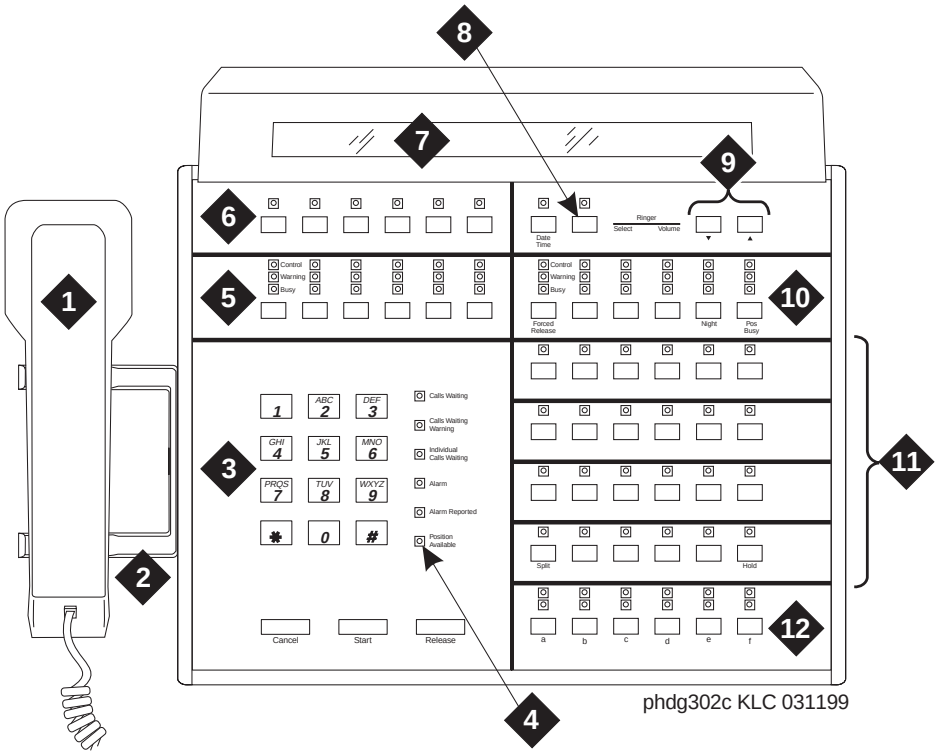


Figure notes:

1. **Handset**
2. **Handset cradle**
3. **Call processing area**
4. **Warning lamps and call waiting lamps**
5. **Outside-line buttons**
6. **Display buttons**
7. **Display**
8. **Select buttons**
9. **Volume control buttons**
10. **Outside-line buttons**
11. **Feature buttons**
12. **Call appearance buttons**

Figure 50: Console feature button layout

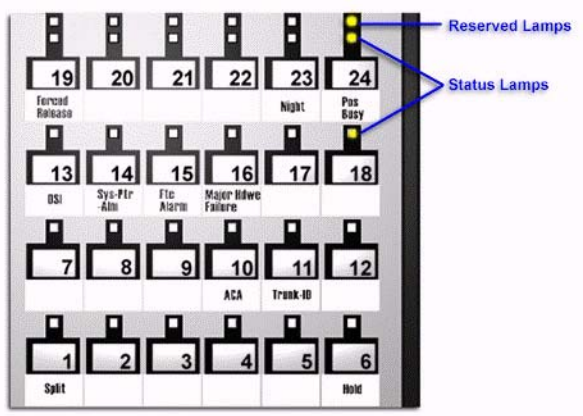
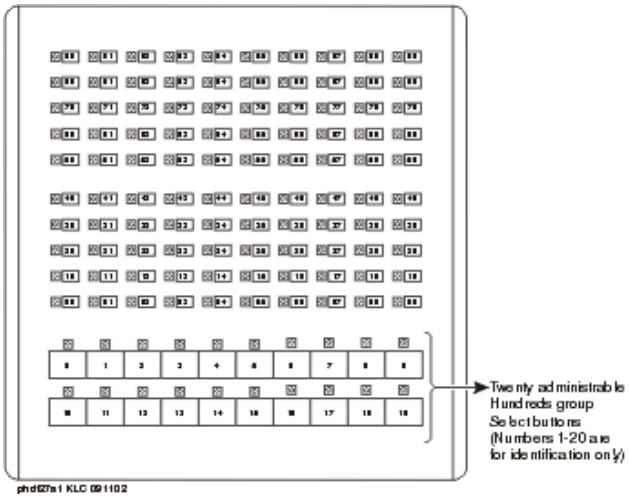


Figure 51: Enhanced Selector Console



302D Console

The 302D console provides the following enhancements to the 302C console:

- Modular handset/headset connection

  The console accepts a standard RJ11, 4-pin modular handset or headset. This connection replaces the quarter-inch, dual-prong handset/headset connection.

- Activate/deactivate push-button

  You can use the push-button on the left side of the console to activate or deactivate the console. A message appears on the console identifying that the button must be pressed to activate the console.

- Two-wire DCP compatibility

  The console is compatible with two-wire DCP circuit packs only, not four-wire DCP circuit packs.

- Headset volume control

  The console can now control the volume of an attached headset.

- Noise expander option

  The console has circuitry to help reduce background noise during pauses in speech from the console end of a conversation. This option is normally enabled.

- Support for Eurofont or Katakana character set

  The console can show the Eurofont or Katakana character set. Administration of these character sets must be coordinated with the characters sent from Avaya Communication Manager.

Avaya PC consoles

The Avaya PC Console is a Microsoft Windows-based call handling application for Avaya Communication Manager attendants. It provides an ideal way to increase your productivity and to better serve your customers.

PC Console offers all the call handling capabilities of the hardware-based Avaya 302 attendant console with a DXS module, plus several enhanced features and capabilities. The enhanced features provide you with the ability to see up to six calls at once, and to handle all calls more efficiently.

PC Console also provides a powerful directory feature. You are able to perform searches, display user information, including a photo. You are able to place a call immediately from the directory.

And, because PC Console resides on a Windows-based PC, you are able to use other software applications at the same time. If a call comes in while you are in another application, you are able to handle it immediately.

For more information about the Avaya PC Console, contact your Avaya account team or representative.

SoftConsole IP Attendant

The SoftConsole is a Windows-based application that can replace the 302B hard console. The SoftConsole is similar to PC Console, but it performs call answering and routing through a PC interface via IP. For more information, contact your Avaya account team or representative.

Adding an Attendant Console

Usually Avaya connects and administers your primary attendant console during cutover. However, you might find a need for a second attendant console, such as a backup console that is used only at night. Note that these instructions do not apply to adding a PC Console or SoftConsole. For more information, see the appropriate console documentation.

To add a night-only attendant console, complete the following steps:

1. Type **add attendant 2** and press **Enter**.

   The **Attendant Console** screen appears.

Figure 52: Attendant Console Button Layout

```
                          ATTENDANT CONSOLE 2

        Type: 302           Name: operator
   Extension:               Group: 1             Auto Answer: none
Console Type: night-only       TN: 1             Data Module? n
        Port:                 COR: 1        Disp Client Redir? n
                              COS: 1         Display Language: english

DIRECT TRUNK GROUP SELECT BUTTON ASSIGNMENTS (Trunk Access Codes)
    Local Remote            Local Remote              Local Remote
 1:                      5:                         9:
 2:                      6:                        10:
 3:                      7:                        11:
 4:                      8:                        12:

HUNDREDS SELECT BUTTON ASSIGNMENTS
 1:           5:            9:           13:           17:
 2:           6:           10:           14:           18:
 3:           7:           11:           15:           19:
 4:           8:           12:           16:           20:
```

2. In the **Type** field, enter **302**.

   This is the type of attendant console.

3. If you want this attendant to have its own extension, enter one in the **Extension** field.

   Tip:

   If you assign an extension to the console, the class of restriction (COR) and class of service (COS) that you assign on this Console screen override the COR and COS you assigned on the **Console Parameters** screen. To avoid unexpected behavior, you should assign the same COR and same COS on both screens.

If you give your attendants an individual extension, users can call the attendant directly by dialing the extension.

Individual attendant extensions also allow attendants to use features that an attendant group cannot use — for example, you can assign them to hunt groups.

4. In the **Console Type** field, enter **night-only**.

   This indicates how this console is used in your organization—as a principal, day only, night only, or day/night console. You can have only one night-time console (night only or day/night) in the system.

5. In the **Port** field, enter the port address for this console.
6. Type a name to associate with this console in the **Name** field.
7. In the **Direct Trunk Group Select Button Assignments** fields, enter trunk access codes for the trunks you want the attendant to be able to select with just one button.
8. If you are using the **Enhanced Selector** console, assign the **Hundreds Select Buttons** that you want this console to have.

   If you want this console to be able to access extensions in the range 3500 to 3999, you need to assign them 5 **Hundreds Select Buttons**: **35** for extensions 3500 to 3599, **36**, **37**, **38**, and **39**.

9. Assign the Feature Buttons that you want the 302 console to have.

   To determine which buttons you can assign to a console, see Attendant Console Feature Buttons

   Tip:

   Feature buttons are not numbered top-to-bottom on the attendant console, as you might expect. Button numbers map to physical positions on the console as shown in Figure 50.

10. Press **Enter** to save your changes.

Attendant Console Feature Buttons

The following table lists the feature buttons that you can assign to an attendant console.

Table 3: Attendant console feature buttons *1 of 7*

Feature or Function	Recommended Button Label	Name Entered on Station Screen	Maximum Allowed	Notes
Abbreviated Dialing	AD	abrv-dial (List:___ DC:___)	1 per List/ DC	1
Administered Connection [status lamp]	AC Alarm	ac-alarm	1	
Automatic Call Distribution (ACD)	After Call Work	after-call (Grp. No.__)	N	2
	Assist	assist (Grp. No:__)	1 per split group	2
	Auto In	auto-in (Grp. No.__)	1 per split group	2
	Auxiliary Work	aux-work (Grp. No.__)	1 per split group	2
	Manual-In	manual-in (Grp. No.__)	1 per split group	2
	Release	release	1	
	Work Code	work-code	1	
	Stroke (0-9)	stroke-cnt (Code:_)	1	3
Attendant Console (Calls Waiting)	CW Aud Off	cw-ringoff	1	
Attendant Control of Trunk Group Access (Activate)	Cont Act	act-tr-grp	1	
Attendant Control of Trunk Group Access (Deactivate)	Cont Deact	deact-tr-g	1	

1 of 7

Table 3: Attendant console feature buttons *2 of 7*

Feature or Function	Recommended Button Label	Name Entered on Station Screen	Maximum Allowed	Notes
Attendant Direct Trunk Group Select	Local TG Remote TG	local-tgs (TAC:__) remote-tgs (LT:__) (RT:__)	12	4
Attendant Crisis Alert	Crisis Alert	crss-alert	1	
Attendant Display [display buttons]	Date/Time	date-time	1	
	Inspect Mode	inspect	1	
	Normal Mode	normal	1	
	Stored Number	stored-num	1	
Attendant Hundreds Group Select	Group Select _	hundrd-sel (Grp:__)	20 per console	5
Attendant Room Status	Occupied Rooms Status	occ-rooms	1	6
	Maid Status	maid-stat	1	6
Attendant Override	Override	override	1	
Automatic Circuit Assurance	ACA	aca-halt	1 per system	
Automatic Wakeup (Hospitality)	Auto Wakeup	auto-wkup	1	
Busy Verification	Busy Verify	verify	1	
Call Coverage	Cover Cback	cov-cback	1	
	Consult	consult	1	
	Go To Cover	goto-cover	1	
Call Coverage [display button]	Cover Msg Rt	cov-msg-rt	1	
Call Offer (Intrusion)	Intrusion	intrusion	1	

2 of 7

Table 3: Attendant console feature buttons *3 of 7*

Feature or Function	Recommended Button Label	Name Entered on Station Screen	Maximum Allowed	Notes
Call Prompting [display button]	Caller Info	callr-info	1	
Call Type	Call Type	type-disp	1	
Centralized Attendant Service	CAS-Backup	cas-backup	1	
Check In/Out (Hospitality) [display buttons]	Check In	check-in	1	
	Check Out	check-out	1	
Class of Restriction [display button]	COR	class-rstr	1	
Conference Display [display button]	Conference Display	conf-dsp	1	
Demand Print	Print Msgs	print-msgs	1	
DID View	DID View	did-view	1	
Do Not Disturb (Hospitality)	Do Not Disturb	dn-dst	1	
Do Not Disturb (Hospitality) [display buttons]	Do Not Disturb Ext	ext-dn-dst	1	
	Do Not Disturb Grp	grp-dn-dst	1	
Don’t Split	Don’t Split	dont-split	1	
Emergency Access To the Attendant	Emerg. Access To Attd	em-acc-att	1	
Facility Busy Indication [status lamp]	Busy (trunk or extension#)	busy-ind (TAC/Ext: _)	1 per TAC/ Ext.	[7]
Facility Test Calls [status lamp]	FTC Alarm	trk-ac-alm	1	
Far End Mute [display button]	Far End Mute for Conf	fe-mute	1	

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Table 3: Attendant console feature buttons *4 of 7*

Feature or Function	Recommended Button Label	Name Entered on Station Screen	Maximum Allowed	Notes
Group Display	Group Display	group-disp	1	
Group Select	Group Select	group-sel	1	
Hardware Failure [status lamps]	Major Hdwe Failure	major-alrm	10 per system	
	Auto Wakeup	pr-awu-alm	1	
	DS1 (facility)	ds1-alarm	10 per system	
	PMS Failure	pms-alarm	1	
	PMS Ptr Alm	pr-pms-alm	1	
	CDR 1 Failure	cdr1-alrm	1	
	CDR 2 Failure	cdr2-alrm	1	
	Sys Ptr Alm	pr-sys-alm	1	
Hold	Hold	hold	1	
Integrated Directory [display button]	Integrtd Directory	directory	1	
Incoming Call Identification	Coverage (Group number, type, name, or ext.#)	in-call-id	N	
Intrusion (Call Offer)	Intrusion	intrusion	1	
Leave Word Calling	Cancel LWC	lwc-cancel	1	
	LWC	lwc-store	1	
Leave Word Calling [display buttons]	Delete Msg	delete-msg	1	
	Next	next	1	
	Call Display	call-disp	1	

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Table 3: Attendant console feature buttons *5 of 7*

Feature or Function	Recommended Button Label	Name Entered on Station Screen	Maximum Allowed	Notes
Leave Word Calling (Remote Message Waiting) [status lamp]	Msg (name or extension #)	aut-msg-wt (Ext:___)	N	
Link Failure	Link Failure (Link No.__)	link-alarm (Link No.__)	1 per Link #	8
Login Security Violation	lsvn-halt	lsvn-halt	1 per system	
Message Waiting	Message Waiting Act.	mwn-act	1 per system	
	Message Waiting Deact.	mwn-deact	1 per system	
Night Service	Trunk Grp. NS	trunk-ns (Grp. No.__)	1 per trunk group	9
No Answer Alert	noans-altr	noans-altr	1 per group	
Off Board Alarm	off-bd-alm	off-bd-alm	1 per group	
PMS Interface [display buttons]	PMS display			
Priority Calling	Prior Call	priority	N	
Position Busy	Position Busy	pos-busy	1	
Queue Status Indications (ACD) [display buttons]	AQC	atd-qcalls	1	
	AQT	atd-qtime		
Queue Status Indications (ACD) [status lamps]	NQC	q-calls (Grp:_)	1	10
	OQT	q-time Grp:_)	1 per hunt group	10
Remote Access Security Violation	rsvn-halt	rsvn-halt	1 per system	
Ringing	In Aud Off	in-ringoff	1	
Security Violation Notification Halt	ssvn-halt	ssvn-halt	1 per system	
Serial Call	Serial Call	serial-cal	1	

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Table 3: Attendant console feature buttons *6 of 7*

Feature or Function	Recommended Button Label	Name Entered on Station Screen	Maximum Allowed	Notes
Split/Swap	Split-swap	split-swap	1	11
System Reset Alert	System Reset Alert [status lamp]	rs-alert	1	
Station Security Code Notification Halt	ssvn-halt	ssvn-halt	1 per system	
Night Service (ACD)	Hunt Group	hunt-ns (Grp. No.__)	3 per hunt group	12
Time of Day Routing [display buttons]	Immediate Override	man-ovrid	1	
	Clocked Override	clk-overid	1	
Timed Reminder	RC Aud Off	re-ringoff	1	
Timer	Timer	timer	1	
Trunk Identification [display button]	Trunk-ID	trk-id	1	
Trunk Group Name [display button]	Trunk-Name	trunk-name	1	
Visually Impaired Service (VIAS)	VIS	vis	1	
	Console Status	con-stat	1	
	Display	display	1	
	DTGS Status	dtgs-stat	1	
	Last Message	last-mess	1	
	Last Operation	last-op	1	
VDN of Origin Announcement Repeat	VOA Repeat	voa-repeat	1	12

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Table 3: Attendant console feature buttons *7 of 7*

Feature or Function	Recommended Button Label	Name Entered on Station Screen	Maximum Allowed	Notes
VuStats	VuStats	vu-display	1	

N = any number of buttons on the telephone can be assigned to this feature. For telephone feature button descriptions, see Telephone feature Buttons on page 123.

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1. List: List number 1 to 3 where the destination number is stored.
   DC: Dial codes of destination number.
2. Grp: The split group number for ACD.
3. Code: Enter a stroke code (0 through 9).
4. TAC: local-tgs — TAC of local TG
   remote-tgs — (L-TAC) TAC of TG to remote PBX
   remote-tgs — (R-TAC) TAC of TG on remote PBX
   The combination of local-tgs/remote-tgs per console must not exceed 12 (maximum).
   Label associated button appropriately so as to easily identify the trunk group.
5. Grp: Enter a hundreds group number (1 through 20).
6. **Enhanced Hospitality** must be enabled on the **System-Parameters Customer-Options** screen.
7. Ext: May be a VDN extension.
8. Link: A link number — 1 to 8 for multi-carrier cabinets, 1 to 4 for single-carrier cabinets.
9. Grp: A trunk group number.
10. Grp: Group number of the hunt group.
11. Allows the attendant to alternate between active and split calls.
12. VDN of Origin must be enabled.

Setting Console Parameters

You can define system-wide console settings on the **Console Parameters** screen.

For example, if you want to warn your attendants when there are more than 3 calls in queue or if a call waits for more than 20 seconds, complete the following steps:

1. Type **change console-parameters** and press **Enter**.

   The **Console Parameters** screen appears.

Figure 53: Console Parameters screen

```
change console-parameters                                    Page   1 of 4
                            CONSOLE PARAMETERS
         Attendant Group Name: OPERATOR
                          COS: 0                                   COR: 0
       Calls in Queue Warning: 5                    Attendant Lockout? y
        Ext Alert Port (TAAS):
                          CAS: none
                                              Night Service Act. Ext.:
               IAS (Branch)? n                IAS Tie Trunk Group No.:
         IAS Att. Access Code:                  Alternate FRL Station:
              Backup Alerting? n         DID-LDN Only to LDN Night Ext? n
     Attendant Vectoring VDN:
```

2. In the **Calls in Queue Warning** field, enter **3**.

   The system lights the console's second call waiting lamp if the number of calls waiting in the attendant queue exceeds 3 calls. Click **Next** to display page 2.

3. In the **Time in Queue Warning** field, enter **20**.

   The system issues a reminder tone if a call waits in the attendant queue for more than 20 seconds.

4. Press **Enter** to save changes.

Note that some of the settings on the individual **Attendant** screens can override your system-wide settings.

Removing an Attendant Console

Before you physically remove an attendant from your system, check the attendant's status, remove it from any group or usage lists, and then delete it from the system's memory.

For example, to remove attendant 3, which also is assigned extension 4345:

1. Type **status attendant 3** and press **Enter**.

   The **Attendant Status** screen appears.

2. Make sure that the attendant:

   - is plugged into the jack
   - is idle (not making or receiving calls)

3. Type **list usage extension 4345** and press **Enter**.

   The **Usage** screen shows where the extension is used in the system.

4. Press **Cancel**.

5. If the attendant extension appears on the **Usage** screen, access the appropriate feature screen and delete the extension.

   For example, if extension 1234 belongs to hunt group 2, type **change hunt group 2** and delete the extension from the list.

6. Type **remove attendant 3** and press **Enter**.

   The system displays the **Attendant Console** screen so you can verify that you are removing the correct attendant.

7. If this is the correct attendant, press **Enter**.

   If the system responds with an error message, the attendant is busy or still belongs to a group. Press **Cancel** to stop the request, correct the problem, and type **remove attendant 3** again.

8. Remove the extension from voice mail service if the extension has a voice mailbox.

9. Type **save translations** and press **Enter** to save your changes.

Note that you do not need to delete the extension from coverage paths. The system automatically adjusts coverage paths to eliminate the extension.

Now you can unplug the console from the jack and store it for future use. You do not need to disconnect the wiring at the cross-connect field. The extension and port address remain available for assignment at a later date.

Providing Backup for an Attendant

Avaya Communication Manager allows you to configure your system so that you have backup positions for your attendant. Attendant Backup Alerting notifies backup telephones that the attendant need assistance in handling calls. The backup telephones are alerted when the attendant queue reaches the queue warning level or when the console is in night service.

Once a backup telephone receives an alert, the user can dial the Trunk Answer Any Station (TAAS) feature access code (FAC) to answer the alerting attendant calls.

Tip:

You can find more information about attendant backup in the *GuestWorks Technician Handbook.*

Before you start

- You can assign the attendant backup alerting only to multiappearance telephones that have a client room class of service (COS) set to No. For more information, see Class of Service on page 830.
- If you have not yet defined a Trunk Answer Any Station (TAAS) feature access code, you need to define one and provide the feature access code to each of the attendant backup users. For more information, see Feature Access Code on page 956.

Instructions

To enable your system to alert backup stations, you need to administer the **Console Parameters** screen for backup alerting. You also need to give the backup phones an attendant queue calls feature button and train your backup users how to answer the attendant calls.

To configure the system to provide backup alerts and to setup extension 4345 to receive these alerts, complete the following steps:

1. Type **`change console-parameters`** and press **Enter**.

   The **Console Parameters** screen appears.

2. In the **Backup Alerting** field, enter **y**.
3. Press **Enter** to save changes.

   The system will now notify anyone with an attendant queue calls button when the attendant queue reaches the warning level or when the console is in night service.

4. Type **change station 4345** and press **Enter**.

   The **Station** screen appears.

5. In one of the Button Assignment fields, enter **atd-qcalls**.

   The atd-qcalls button provides the visual alerting for this telephone. When this button is dark (idle state), there are no calls in the attendant queue. When the button shows a steady light (busy state), there are calls in the attendant queue. When button shows a flashing light (warning state), the number of calls in the attendant queue exceeds the queue warning. The backup-telephone user also hears an alerting signal every 10 seconds.

6. Press **Enter** to save changes.

Now you need to train the user how to interpret the backup alerting and give them the TAAS feature access code so that they can answer the attendant calls.

Chapter 6: Managing Displays

Displaying Caller Information

This chapter provides information on the messages that appear on the read-out screen on display phones.

Your system uses automatic incoming call display to provide information about incoming calls to a display telephone that is in use, or active on a call. The information is displayed for 30 seconds on all phones except for CALLMASTER phones, where the display goes blank after 30 seconds. However, the information for each new call overrides the existing message.

Call information appears on the display only if the call terminates at the telephone. For example, if the call is forwarded to another extension, no call information appears.

For more information on the buttons and languages you can set up for the messages that appear on the display, see the "Telephone Displays" feature description in the *Feature Description and Implementation for Avaya Communication Manager,* 555-245-505.

Displaying ANI Calling Party Information

Calling party information may consist of either a billing number that sometimes is referred to as Automatic Number Identification (ANI), or a calling party number. Your telephone may display the calling party number and name, or the incoming trunk group name.

Instructions

We will set up tie trunk group 10 to receive calling party information and display the calling party number on the telephone of the person called.

1. Type **`change trunk group 10`**.

   The **Trunk Group** screen for trunk group 10 appears.

Figure 54: Trunk Group screen

```
                                 TRUNK GROUP

 Group Number: 10_                 Group Type: tie_____      CDR Reports: r
   Group Name: Node1 to Node3      ________ COR: 10       TN: ___      TAC: *10__
    Direction: two-way        Outgoing Display? n    Trunk Signaling Type: ____
 Dial Access? n                 Busy Threshold: 99_         Night Service: _____
 Queue Length: ___                                  Incoming Destination: 2000_
    Comm Type: voice                 Auth Code? n
                                   Trunk Flash? n
          BCC: _                            ITC? ____
 TRUNK PARAMETERS
    Trunk Type (in/out): auto/auto-incorrect Incoming Rotary Timeout(sec): 5
     Outgoing Dial Type: _________                  Incoming Dial Type: tone
                                                Disconnect Timing(msec): 500
        Digit Treatment: __________                          Digits: ____
                                                    Sig Bit Inversion: none
     Incoming Dial Tone? y

Disconnect Supervision - In? y  Out? n
  Answer Supervision Timeout: 0__           Receive Answer Supervision? y
```

2. Type **tone** in the **Incoming Dial Type** field.
3. Press **Next Page** and type ***ANI*DNIS** in the **Incoming Tone (DTMF) ANI** field.
4. Press **Enter** to save your changes.

Displaying ICLID Information

Communication Manager collects the calling party name and number (Incoming Call Line Identification, or ICLID) received from the central office (CO) on analog trunks.

Before you start

Be sure the **Analog Trunk Incoming Call ID** field is set to **y** on the **System-Parameters Customer-Options** screen.

See the *Hardware Guide* for *Avaya Communication Manager (*555-245-207*)* for information on the required circuit pack.

Instructions

We will set up the analog diod trunk group 1 to receive calling party information and display the calling party number on the telephone of the person called.

1. Type **`change trunk group 1`**.

   The **Trunk Group** screen for trunk group 1 appears. The **Group Type** field is already set to diod.

2. Press **Next Page** to display the **Trunk Features** screen.

Figure 55: Trunk Features screen

```
change trunk-group 1                                              Page 2 of x
                                TRUNK GROUP
TRUNK FEATURES
          ACA Assignment? n           Measured: none
                                                      Maintenance Tests? y
                             Data Restriction? n
   Suppress # Outpulsing? n
                                  Receive Analog Incoming Call ID: Bellcore
Incoming Tone (DTMF) ANI: no
```

3. Type **Bellcore** in the **Receive Analog Incoming Call ID** field.
4. Press **Next Page** to display the **Administrable Timers** screen.

Figure 56: Administrable Timers screen

```
change trunk-group 1                                              Page 3 of x
                                TRUNK GROUP
ADMINISTRABLE TIMERS
   Incoming Disconnect(msec): 500
   Incoming Dial Guard(msec): 70
                                              Incoming Seizure(msec): 120
          Flash Length(msec): 540  Incoming Incomplete Dial Alarm (sec): 255
END TO END SIGNALING
    Tone (msec): 350   Pause (msec): 150
```

5. Type **120** in the **Incoming Seizure (msec)** field.
6. Press **Enter** to save your changes.

Changing the Display Language

This section explains how to set or change the display language.

Changing the display language

To set or change the display language:

1. Type **change attendant *n***, where ***n*** is the number of the attendant console that you want to change. Press **Enter**.

   The system displays the **Attendant Console** screen.

Figure 57: Attendant Console screen

```
change attendant 1                                        Page   1 of   3
                             ATTENDANT CONSOLE 1

        Type: console           Name: Gunther
   Extension: 1000             Group: 1               Auto Answer: none
Console Type: principal           TN: 1               Data Module? n
        Port: 01C1106             COR: 1          Disp Client Redir? n
                                  COS: 1           Display Language: english


DIRECT TRUNK GROUP SELECT BUTTON ASSIGNMENTS (Trunk Access Codes)
    Local Remote           Local Remote              Local Remote
 1: 9                   5:                         9:
 2: 82                  6:                        10:
 3:                     7:                        11:
 4:                     8:                        12:

HUNDREDS SELECT BUTTON ASSIGNMENTS
 1:           5:            9:           13:           17:
 2:           6:           10:           14:           18:
 3:           7:           11:           15:           19:
 4:           8:           12:           16:           20:
```

2. In the **Display Language** field, enter the display language you want to use.
3. Press **Enter** to save your changes.

Entering translations for a user-defined language

To enter translations for a user-defined language:

1. Type **change attendant *n***, where ***n*** is the number of the attendant console you want to change. Press **Enter**.

   The system displays the **Attendant Console** screen.

2. In the **Display Language** field, enter **user-defined**.
3. Press **Enter** to save your changes.
4. Type **change display-messages *n***, where ***n*** is the message for which you want to translate the display language. Click **help** to view the messages that you can choose to translate. Press **Enter**.

   The system displays the **Language Translations** screen for the type of message that you want to translate. This example shows the screen for message type "Transfer-conference."

Figure 58: Language Translations screen

```
change display-messages transfer-conference                Page 1 of 3

                                Language Translations

  1.     English: Transfer completed.
     Translation: abtretung abgeschlossen
```

5. In the **Translation** field, type the translation of the message in the user-defined language. In this example, the translation is **abtretung abgeschlossen**.
6. Press **Enter** to save your changes.

Administering Unicode display

To use Unicode display languages, you must have the appropriate Avaya Unicode Message files loaded on Communication Manager. These files are named *avaya_unicode.txt* and *custom_unicode.txt*. As of the January 2005 2.2 Release of Communication Manager, the following languages are available using Unicode display: English, Japanese, Chinese, Russian, German, Hungarian, Dutch, and Czech. The Unicode Message files for each of these languages are available for download on the Avaya support Web site (http://www.avaya.com/support). Additional languages are periodically becoming available, so check this site often for the most up-to-date message files.

Note:

Message files for each of these languages are contained in a downloadable ZIP file that also contains the following two white papers: *CM2.0WhitePaper.pdf* and *UnicodeMessageFileWhitePaper.pdf*. Refer to these white papers for details on the following procedures:

To administer Unicode display:

1. Download the appropriate Unicode message file to your PC. For an existing translation, download the desired language. To create a new translation, download the English version.
2. If necessary, create a new translation, or modify an existing translation. Use the Avaya Excel Translation Editor to create or modify translations.
3. Use the AvayaL10n Tool to prepare the Unicode Message file for installation. This tool is included in the Unicode Message file ZIP file. The Unicode Message file will *not* install if you do not first process the file with this tool.
4. Transfer the Unicode Message file to an Avaya media server that is running Communication Manager 2.0 or later. You can use the Avaya Web pages, the Avaya Installation Wizard, or ftp to transfer the Unicode Message file.
5. Install the Unicode message file. You can install Unicode Message files from either the Install Web page or the Avaya Installation Wizard. Note that the Installation Wizard is the same wizard that you use to transfer Unicode Message files to an Avaya media server that is running Communication Manager.
6. Load the Unicode message files. After you install the Unicode Message files, you must do a reset 4 to load the Unicode Message files in the Avaya media server that is running Communication Manager 2.0 or later. Starting with Communication Manager 2.1, a reset 4 is no longer required to load the Unicode Message file *custom_unicode.txt*.

Note:

Unicode display is only available for Unicode-supported telephones. Currently, only the 4620SW and 4610SW support Unicode display.

Unicode "Native Name" support

Communication Manager supports Unicode for the "Name" associated with Vector Directory Numbers (VDNs), trunk groups, hunt groups, and stations. The Unicode Name (also referred to as *Native Name* and *Name 2)* fields are hidden fields that are associated with the name fields you administer on the respective screens for each. These fields can only be administered using Avaya Site Administration (ASA) or MultiSite Administrator (MSA).

- The Unicode VDN Name is associated with the name administered in the **Name** field on the **Vector Directory** screen. You must use MSA.
- The Unicode Trunk Group Name is associated with the name administered in the **Group Name** field on the **Trunk Group** screen. You must use MSA.

- The Unicode Hunt Group Name is associated with the name administered in the **Group Name** field on the **Hunt Group** screen. You must use MSA.
- The Unicode Station Name is associated with the name administered in the **Name** field on the **Station** screen. You must use ASA or MSA.

Station General Status screen

The **General Status** screen for stations contains three Unicode script-related fields. To access the **General Status** screen, type **`status station`** ***xxxx***, where ***xxxx*** is the extension of the station. The **General Status** screen appears. Click **Next** to display page 2 of the screen:

Figure 59: General Status screen

```
 status station xxxx                                          page 2 of x

                                GENERAL STATUS

CONNECTED STATION INFORMATION
               Part ID Number: unavailable
                Serial Number: unavailable
          Station Lock Active? No

UNICODE DISPLAY INFORMATION
          Native Name Scripts: 0x00000007:Latn;Lat1;LatA
     Display Messages Scripts: 0x04000007:Latn;Lat1;LatA;Jpan
    Station Supported Scripts: 0x7c000007:Latn;Lat1;LatA;Jpan;Chis;Chit;Korn
```

"Scripts" are a collection of symbols used to represent text in one or more writing systems. The three script fields shown in the **UNICODE DISPLAY INFORMATION** section are as follows:

Native Name Scripts: Scripts supported in the Unicode station name.

Display Messages Scripts: The scripts used in the Unicode Display Language.

Station Supported Scripts: The scripts supported in the IP station that is registered to an extension.

The following table defines the script tags and spells out the script abbreviations.

Script	Script tag (hex)	Start .. End Codes	Script or block name	SAT screen name (note 1)
1	00000001	0000..007F	Basic Latin	Latn
2	00000002	0080..00FF	Latin-1 Supplement	Lat1
3	00000004	0100..017F	Latin Extended-A	LatA

1 of 6

Script	Script tag (hex)	Start .. End Codes	Script or block name	SAT screen name (note 1)
4	00000008	0180..024F	Latin Extended-B	LatB
		0250..02AF	IPA Extensions	
		02B0..02FF	Spacing Modifier Letters	
		0300..036F	Combining Diacritical Marks	
5	00000010	0370..03FF	Greek and Coptic	Grek
6	00000020	0400..04FF	Cyrillic	Cyrl
	00000020	0500..052F	Cyrillic Supplementary	
7	00000040	0530..058F	Armenian	Armn
8	00000080	0590..05FF	Hebrew	Hebr
9	00000100	0600..06FF	Arabic	Arab
		0700..074F	Syriac	
		0780..07Bf	Thaana	
10	00000200	0900..097F	Devanagari	Deva
11	00000400	0980..09FF	Bengali	Beng
12	00000800	0A00..0A7F	Gurmukhi	Guru
13	00001000	0A80..0AFF	Gujarati	Gujr
14	00002000	0B00..0B7F	Oriya	Orya
15	00004000	0B80..0BFF	Tamil	Taml
16	00008000	0C00..0C7F	Telugu	Telu
17	00010000	0C80..0CFF	Kannada	Knda
18	00020000	0D00..0D7F	Malayalam	Mlym
19	00040000	0D80..0DFF	Sinhala	Sinh
20	00080000	0E00..0E7F	Thai	Thai
21	00100000	0E80..0EFF	Lao	Laoo
		0F00..0FFF	Tibetan	
22	00200000	1000..109F	Myanmar	Mymr

2 of 6

Script	Script tag (hex)	Start .. End Codes	Script or block name	SAT screen name (note 1)
23	00400000	10A0..10FF	Georgian	Geor
	80000000	1100..11FF	Hangul Jamo	
		1200..137F	Ethiopic	
		13A0..13FF	Cherokee	
		1400..167F	Unified Canadian Aboriginal Syllabics	
		1680..169F	Ogham	
		16A0..16FF	Runic	
24	00800000	1700..171F	Tagalog	Tglg
		1720..173F	Hanunoo	
		1740..175F	Buhid	
		1760..177F	Tagbanwa	
25	01000000	1780..17FF	Khmer	Khmr
		1800..18AF	Mongolian	
		1E00..1EFF	Latin Extended Additional	
		1F00..1FFF	Greek Extended	
		2000..206F	General Punctuation	
		2070..209F	Superscripts and Subscripts	
		20A0..20CF	Currency Symbols	
		20D0..20FF	Combining Diacritical Marks for Symbols	
		2100..214F	Letterlike Symbols	
		2150.218F	Number Forms	
		2190..21FF	Arrows	
		2200..22FF	Mathematical Operators	
		2300..23FF	Miscellaneous Technical	
		2400..243F	Control Pictures	

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Script	Script tag (hex)	Start .. End Codes	Script or block name	SAT screen name (note 1)
		2440..245F	Optical Character Recognition	
		2460..24FF	Enclosed Alphanumerics	
		2500..257F	Box Drawing	
		2580..259F	Geometric Shapes	
		2600..26FF	Miscellanious Symbols	
		2700..27BF	Dingbats	
		27C0..27EF	Miscellaneous Mathematical Symbols-A	
		27F0..27FF	Supplemental Arrows-A	
26	02000000	2800..28FF	Braille Patterns	Bral
		2900..297F	Supplemental Arrows-B	
		2980..29FF	Miscellaneous Mathematical Symbols-B	
		2A00..2AFF	Supplemental Mathematical Operators	
	04000000 08000000 10000000 20000000 40000000	2E80..2EFF	CJKV Radicals Supplement	
	04000000 08000000 10000000 20000000 40000000	2F00..2FDF	Kangxi Radicals	
		2FF0..2FFF	Idographic Description Characters	
	04000000 08000000 10000000 20000000 40000000	3000..303F	CJKV Symbols and Punctuation	
	04000000	3040..309F	Hiragana	

Script	Script tag (hex)	Start .. End Codes	Script or block name	SAT screen name (note 1)
	04000000	30A0..31BF	Katakana	
	10000000	3100..312F	Bopomofo	
	80000000	3130..318F	Hangul Compatibility Jamo	
		31‘90..319F	Kanbun	
	10000000	31A0..31BF	Bopomofo Extended	
	04000000	31F0..31FF	Katakana Phonetic Extensions	
	04000000 08000000 10000000 20000000 40000000	3200..32FF	Enclosed CJK Letters and Months	
	04000000 08000000 10000000 20000000 40000000	3300..33FF	CJKV Compatibility	
	04000000 08000000 10000000 20000000 40000000	3400..4DBF	CJKV Unified Ideographs Extension A	
27	04000000	4E00..9FFF	CJKV Unified Ideographs (Japanese)	Jpan
28	08000000	4E00..9FFF	CJKV Unified Ideographs (Simplified Chinese)	ChiS
29	10000000	4E00..9FFF	CJKV Unified Ideographs (Traditional Chinese)	ChiT
30	20000000	4E00..9FFF	CJKV Unified Ideographs (Korean-Hanja)	Korn
31	40000000	4E00..9FFF	CJKV Unified Ideographs (Vietnamese)	Viet
		A000..A48F	Yi Syllables	
		A490..A4Cf	Yi Radicals	

Script	Script tag (hex)	Start .. End Codes	Script or block name	SAT screen name (note 1)
32	80000000	AC00..D7AF	Hangul Syllables	Hang
		D800..DB7F	High Surrogates	
		DB80..DBFF	High Private Use Surrogates	
		DC00..DFFF	Low Surrogates	
		E000..F8FF	Private Use Area	
	04000000 08000000 10000000 20000000 40000000	F900..FAFF	CJK Compatibility Ideographs	
		FB00..FBF4	Alphabetic Presentation Forms	
	00000100	FB50..FDFF	Arabic Presentation Forms-A	
		FE00..FE0F	Variation Selectors	
		FE20..FE2F	Combining Half Marks	
	04000000 08000000 10000000 20000000 40000000	FE30..FE4F	CJK Compatibility Forms	
		FE50..FE6F	Small Form Variants	
	00000100	FE70..FEFF	Arabic Presentation Forms-B	
	04000000 08000000 10000000 20000000 40000000 80000000	FF00..FFEF	Halfwidth and Fullwidth Forms	
		FFF0..FFFF	Specials	

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Related topics

See the "Telephone Displays" and the "Administrable Display Languages" feature descriptions in the *Feature Description and Implementation for Avay Communication Manager* for more information about choosing the language for messages on your display phones and for mapping US English to Cyrillic (for Russian), Katakana (for Japanese), European, or Ukrainian characters.

Fixing problems

Symptom	Cause and Solution
Characters that display are not what you thought you entered.	This feature is case sensitive. Check the table to make sure that you entered the right case.
You entered "~c", and "*" appears on the display instead.	Lower-case "c" has a specific meaning in Avaya Communication Manager, and therefore cannot be mapped to any other character. An asterisk "*" appears in its place.
You entered "~->" or "~<-" and nothing appears on the display.	These characters do not exist as single keys on the standard US-English keyboard. Therefore the system is not programmed to handle them.
Enhanced display characters appear in fields that you did not update.	If an existing display field contains a tilde (~) followed by Roman characters, and you update and submit that screen after this feature is activated, that field will display the enhanced character set.
Nothing displays on the terminal at all.	Some unsupported terminals do not display anything if a special character is presented. Check the model of display terminal that you are using.
You entered a character with a descender and part of it appears cut off in the display.	Some of the unused characters in Group2a have descenders that do not appear entirely within the display area. These characters are not included in the character map. For these characters (g,j,p,q,y), use Group1 equivalents.

Setting up Directory Buttons

Your Communication Manager directory contains the names and extensions that are assigned on each station screen. Display-telephone users can use a telephone button to access the directory, use the touch-tone buttons to key in a name, and retrieve an extension from the directory.

Instructions

We will assign directory telephone buttons for extension 2000. Our button assignment plan is set up so that telephone buttons 6, 7, and 8 are used for the directory. Remember, the name you type in the Name field on the first page of the station screen is the name that appears when the directory is accessed on a telephone display.

1. Type **change station 2000**.

   The **Station** screen for extension 2000 appears.

2. Move to the **Button Assignments** section.

Figure 60: Station screen

```
                                                          Page 3 of X
                                STATION
     SITE DATA
          Room: _______                                    Headset? n
          Jack: ___                                        Speaker? n
         Cable: ___                                       Mounting: d
         Floor: _______                                 Cord Length: 0_
      Building: _______                                   Set Color: _____

ABBREVIATED DIALING
     List1: ________          List2: _________          List3: _________

BUTTON ASSIGNMENTS
 1: call-appr                          5:
 2: call-appr                          6: directory
 3: call-appr                          7: next
 4: call-appr                          8: call-display
```

3. In **Button Assignment** field 6, type **directory**.
4. In **Button Assignment** field 7, type **next**.
5. In **Button Assignment** field 8, type **call-display**.
6. Press **Enter** to save your changes.

Chapter 7: Handling Incoming Calls

Setting up Basic Call Coverage

This chapter shows you how to set up call coverage for incoming calls to be sure that incoming calls are answered when the called party is not available.

Basic incoming call coverage:

- provides for automatic redirection of calls to alternate destinations when the called party is not available or not accepting calls
- provides the order in which Avaya Communication Manager redirects calls to alternate phones or terminals
- establishes up to 6 alternate termination points for an incoming call
- establishes redirection criteria that govern when a call redirects
- redirects calls to a local telephone number (extension) or an off-switch telephone number (public network)

Call coverage allows an incoming call to redirect from its original destination to an extension, hunt group, attendant group, uniform call distribution (UCD) group, direct department calling (DDC) group, automatic call distribution (ACD) split, coverage answer group, Audio Information Exhange (AUDIX), or vector for a station not accepting calls.

Administering system-wide call coverage characteristics

This section shows you how to set up system-wide call coverage characteristics that govern how coverage is handled.

Instructions

The **System Parameters Call Coverage/Call Forwarding** screen sets up the global parameters which direct Communication Manager how to act in certain situations.

1. Leave all default settings as they are set for your system.
2. If you desire to customize your system, carefully read and understand each field description before you make any changes.

   See the **System Parameters Call Coverage/Call Forwarding** screen for more information.

Related topics

See Covering calls redirected to an off-site location on page 195 for more information on redirecting calls.

Creating coverage paths

This section explains how to administer various types of call coverage. In general, call coverage refers to what happens to incoming calls. You can administer paths to cover all incoming calls, or define paths for certain types of calls, such as calls to busy phones. You can define where incoming calls go if they are not answered and in what order they reroute to other locations. For example, you can define coverage to ring the called telephone, then move to a receptionist if the call is not answered, and finally access a voice mailbox if the receptionist is not available.

With call coverage, the system redirects a call to alternate answering extensions when no one answers at the first extension. An extension can have up to 6 alternate answering points. The system checks each extension in sequence until the call connects. This sequence of alternate extensions is called a coverage path.

The system redirects calls based on certain criteria. For example, you can have a call redirect to coverage without ever ringing on the principal set, or after a certain number of rings, or when one or all call appearances (lines) are busy. You can set coverage differently for internal (inside) and external (outside) calls, and you can define coverage individually for different criteria. For example, you can decide that external calls to busy phones can use the same coverage as internal calls to phones with Do Not Disturb active.

Note:

If a call with a coverage path is redirected to a coverage point that is not available, the call proceeds to the next coverage point regardless of the type of coverage administered in the point that was unavailable. For example, if the unavailable coverage point has a hunt group coverage path administered, the hunt group coverage path would not be used by a call coming into the hunt group through the higher-level coverage path. The hunt group coverage path would be used only for calls coming directly into the hunt group extension.

Instructions

To create a coverage path:

1. Type **`add coverage path next`** and press **Enter**.

   The **Coverage Path** screen appears.

Figure 61: Coverage path screen

```
                                COVERAGE PATH

                  Coverage Path Number: 2
                                                  Hunt After Coverage: n
                      Next Path Number: ___       Linkage: ___  ___

COVERAGE CRITERIA
    Station/Group Status   Inside Call   Outside Call
              Active?            n             n
                Busy?            y             y
       Don't Answer?             y             y Number of Rings: 3
                 All?            n             n
 DND/SAC/Goto Cover?             y             y

COVERAGE POINTS

 Terminate to Coverage Pts. with Bridged Appearance? n

   Point1: __________ Point2: __________ Point3: __________
   Point4: __________ Point5: __________ Point6: __________
```

The system displays the next undefined coverage path in the sequence of coverage paths. Our example shows coverage path number 2.

2. Type a coverage path number in the **Next Path** field.

   The next path is optional. It is the coverage path to which calls are redirected if the current path's coverage criteria does not match the call status. If the next path's criteria matches the call status, it is used to redirect the call; no other path is searched.

3. Fill in the **Coverage Criteria** fields.

   You can see that the default sets identical criteria for inside and outside calls. The system sets coverage to take place from a busy telephone, if there is no answer after a certain number of rings, or if the **DND** (do not disturb), **SAC** (send all calls), or **Go to Cover** button has been pressed or corresponding feature-access codes dialed.

4. Fill in the **Point** fields with the extensions, hunt group number, or coverage answer group number you want for coverage points.

   Each coverage point can be an extension, hunt group, coverage answer group, remote number, or attendant.

5. Press **Enter** to save your changes.

Tip:

If you want to see which extensions or groups use a specific coverage path, type **`display coverage sender group n`**, where ***n*** is the coverage path number. For example, you should determine which extensions use a coverage path before you make any changes to it.

Assigning a coverage path to users

Now assign the new coverage path to a user. For example, we will assign this new coverage path to extension 2054.

Note:

A coverage path can be used for more than one extension.

Instructions

To assign a coverage path:

1. Type **`change station 2054`** and press **Enter**.

   The **Station** screen for extension 2054 appears.

Figure 62: Station screen

```
                                STATION

Extension: 2054                          Lock Messages? n       BCC: 0
     Type: 7406D                         Security Code: ______   TN: 1
     Port: _____                       Coverage Path 1: 2__     COR: 1
     Name: ____________________________  Coverage Path 2: ___    COS: 1
                                       Hunt-to-Station: _______

STATION OPTIONS
             Data Module? n            Personalized Ringing Pattern: 1
          Display Module? n                        Message Lamp Ext: ________


                                                 Media Complex Ext: ________
                                                      IP Softphone? ________
                                               Remote Office Phone? ________
```

2. Type **2** in the **Coverage Path 1** field.

   To give extension 2054 another coverage path, you can type a coverage path number in the **Coverage Path 2** field.

3. Press **Enter** to save your changes.

Related information

Assigning Coverage Options on page 377

Setting up Advanced Call Coverage

Advanced incoming call coverage:

- redirects calls based on time-of-day
- allows coverage of calls that are redirected to sites not on the local server running Communication Manager
- allows users to change back and forth between two coverage choices (either specific lead coverage paths or time-of-day tables).

Covering calls redirected to an off-site location

You can provide coverage for calls that have been redirected to an off-site location (for example, your home). This capability, called Coverage of Calls Redirected Off-Net (CCRON) allows you to redirect calls onto the public network and bring back unanswered calls for further coverage processing.

Before you start

- On the **System-Parameters Customer-Options** screen, verify the **Coverage of Calls Redirected Off-Net Enabled** field is **y**. If not, contact your Avaya representative.
- You need call classifier ports for all situations except ISDN end-to-end signaling, in which case the ISDN protocol does the call classification. For all other cases, use one of the following:
  - Tone Clock with Call Classifier - Tone Detector circuit pack. See the *Hardware Guide for Avaya Communication Manager* for more information on the circuit pack.
  - Call Classifier - Detector circuit pack.

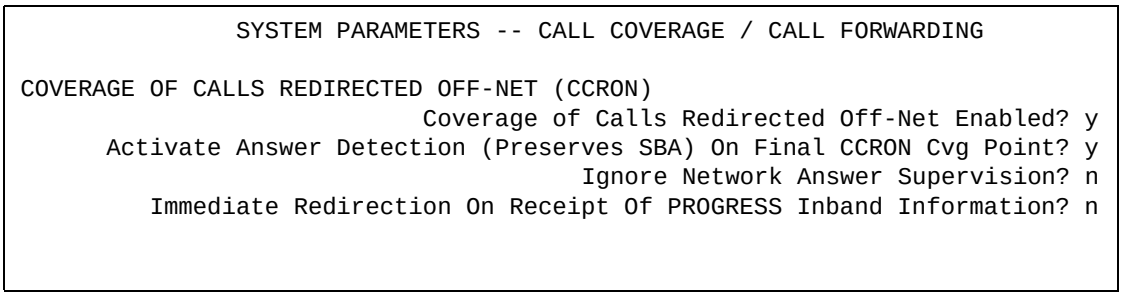
Instructions

To provide coverage of calls redirected to an off-site location:

1. Type **`change system-parameters coverage-forwarding`** and press **Enter**.

   The **System Parameters Call Coverage/Call Forwarding** screen appears.

Figure 63: System Parameters - Call Coverage/Call Forwarding screen



2. In the **Coverage of Calls Redirected Off-Net Enabled** field, type **y**.

   This instructs Avaya Communication Manager to monitor the progress of an off-net coverage or off-net forwarded call and provide further coverage treatment for unanswered calls.

3. In the **Activate Answer Detection (Preserves SBA) On Final CCRON Cvg Point** field, leave the default as **y**.
4. In the **Ignore Network Answer Supervision** field, leave the default as **n**.
5. In the **Immediate Redirection On Receipt Of PROGRESS Inband Information** field, leave the default as **n**.
6. Press **Enter** to save your changes.

Defining coverage for calls redirected to external numbers

You can administer the system to allow calls in coverage to redirect to off-net (external) or public-network numbers.

Standard remote coverage to an external number allows you to send a call to an external telephone, but does not monitor the call once it leaves your system. Therefore, if the call is busy or not answered at the external number, the call cannot be pulled back to the system. With standard remote call coverage, make the external number the last coverage point in a path.

With newer systems, you may have the option to use the Coverage of Calls Redirected Off-Net feature. If this feature is active and you use an external number in a coverage path, the system can monitor the call to determine whether the external number is busy or does not answer. If necessary, the system can redirect a call to coverage points that follow the external number. With this feature, you can have a call follow a coverage path that starts at the user's extension, redirects to the user's home telephone, and if not answered at home, returns to redirect to their voice mail box.

The call will not return to the system if the external number is the last point in the coverage path.

To use a remote telephone number as a coverage point, you need to define the number in the Remote Call Coverage Table and then use the remote code in the coverage path.

Instructions

For example, to add an external number (303-538-1000) to coverage path 2:

1. Type **change coverage remote** and press **Enter**.

   The **Remote Call Coverage Table** screen appears.

Figure 64: Remote Call Coverage Table screen

```
change coverage remote

                      REMOTE CALL COVERAGE TABLE
                      ENTRIES FROM 1  TO 1000

01: 93035381000_____     16: _________________     31: _________________
02: _________________     17: _________________     32: _________________
03: _________________     18: _________________     33: _________________
04: _________________     19: _________________     34: _________________
05: _________________     20: _________________     35: _________________
06: _________________     21: _________________     36: _________________
07: _________________     22: _________________     37: _________________
08: _________________     23: _________________     38: _________________
09: _________________     24: _________________     39: _________________
10: _________________     25: _________________     40: _________________
11: _________________     26: _________________     41: _________________
12: _________________     27: _________________     42: _________________
13: _________________     28: _________________     43: _________________
14: _________________     29: _________________     44: _________________
15: _________________    30: _________________     45: _________________
```

2. Type **93035381000** in one of the remote code fields.

   If you use a digit to get outside of your network, you need to add the digit before the external number. In this example, the system requires a ‘9’ to place outside calls.

3. Be sure to record the remote code number you use for the external number.

   In this example, the remote code is r01.

4. Press **Enter** to save your changes.

5. Type **change coverage path 2** and press **Enter**.

   The **Coverage Path** screen appears.

   Tip:

   Before making changes, you can use **display coverage sender group 2** to determine which extensions or groups use path 2.

Figure 65: Coverage Path screen

```
                    COVERAGE PATH

                 Coverage Path Number: _
                                          Hunt After Coverage: y
                     Next Path Number: ___    Linkage: ___  ___

COVERAGE CRITERIA
    Station/Group Status   Inside Call   Outside Call
             Active?            n             n
               Busy?            y             y
       Don't Answer?            y             y Number of Rings: 2
                All?            n             n
 DND/SAC/Goto Cover?            y             y

COVERAGE POINTS

 Terminate to Coverage Pts. with Bridged Appearance? n

              Point1: 4101_____ Point2: r1_______ Point3: h77_____
              Point4: __________ Point5: __________ Point6: __________
```

6. Type **r1** in a coverage **Point** field.

   In this example, the coverage rings at extension 4101, then redirects to the external number. If you administer Coverage of Calls Redirected Off-Net and the external number is not answered or is busy, the call redirects to the next coverage point. In this example, the next point is Point 3 (h77 or hunt group 77).

   If you do not have the Coverage of Calls Redirected Off-Net feature, the system cannot monitor the call once it leaves the network. The call ends at the remote coverage point.

7. Press **Enter** to save your changes.

Related topics

For more information on coverage, see "Call Coverage" in *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205.

Defining time-of-day coverage

The Time of Day Coverage Table on your system lets you redirect calls to coverage paths according to the time of day and day of the week when the call arrives. You need to define the coverage paths you want to use before you define the time of day coverage plan.

For example, let us say you want to administer the system so that incoming calls to extension 2054 redirect to a coworker in the office from 8:00 a.m. to 5:30 p.m., and to a home office from 5:30 p.m. to 8:00 p.m. on weekdays. You want to redirect the calls to voice mail after 8:00 p.m. weekdays and on weekends.

Instructions

To set up a time-of-day coverage plan that redirects calls for our example above:

1. Type **`add coverage time-of-day next`** and press **Enter**.

   The **Time of Day Coverage Table** screen appears and the selects the next undefined table number in the sequence of time-of-day table numbers. If this is the first time-of-day coverage plan in your system, the table number is 1.

Figure 66: Time of Day Coverage Table screen

```
                    TIME OF DAY COVERAGE TABLE 1__

          Act   CVG     Act   CVG     Act   CVG     Act   CVG     Act   CVG
          Time  PATH    Time  PATH    Time  PATH    Time  PATH    Time  PATH

Sun       00:00 3___    __:__ ____    __:__ ____    __:__ ____    __:__ ____
Mon       00:00 3___    08:00 1       17:30 2       20:00 3       __:__ ____
Tue       00:00 3___    08:00 1       17:30 2       20:00 3       __:__ ____
Wed       00:00 3___    08:00 1       17:30 2       20:00 3       __:__ ____
Thu       00:00 3___    08:00 1       17:30 2       20:00 3       __:__ ____
Fri       00:00 3___    08:00 1       17:30 2       20:00 3       __:__ ____
Sat       00:00 3___    __:__ ____    __:__ ____    __:__ ____    __:__ ____
```

   Record the table number so that you can assign it to extensions later.

2. To define your coverage plan, enter the time of day and path number for each day of the week and period of time.

   Enter time in a 24-hour format from the earliest to the latest. For this example, assume that coverage path 1 goes to the coworker, path 2 to the home, and path 3 to voice mail.

   Define your path for the full 24 hours (from 00:01 to 23:59) in a day. If you do not list a coverage path for a period of time, the system does not provide coverage for that time.

3. Press **Enter** to save your changes.

Now assign the time-of-day coverage to a user. For example, we use extension 2054:

1. Type **change station 2054** and press **Enter**.

   The **Station** screen for extension 2054 appears.

Figure 67: Station screen

```
                                    STATION

Extension: 2054                        Lock Messages? n       BCC: 0
     Type: 7406D                       Security Code: 4196__   TN: 1
     Port: _____                    Coverage Path 1: t1_     COR: 1
     Name: ____________________________ Coverage Path 2: ___    COS: 1
                                    Hunt-to-Station: _______

STATION OPTIONS
             Data Module? n             Personalized Ringing Pattern: 1
          Display Module? n                       Message Lamp Ext: ________

                                                 Media Complex Ext: ________
                                                      IP Softphone? ________
                                               Remote Office Phone? ________
```

2. Move your cursors to Coverage Path 1 and type **t** plus the number of the Time of Day Coverage Table.
3. Press **Enter** to save your changes.

Now calls to extension 2054 redirect to coverage depending on the day and time that each call arrives.

Creating coverage answer groups

You can create a coverage answer group so that up to 8 phones simultaneously ring when calls cover to the group. Anyone in the answer group can answer the incoming call.

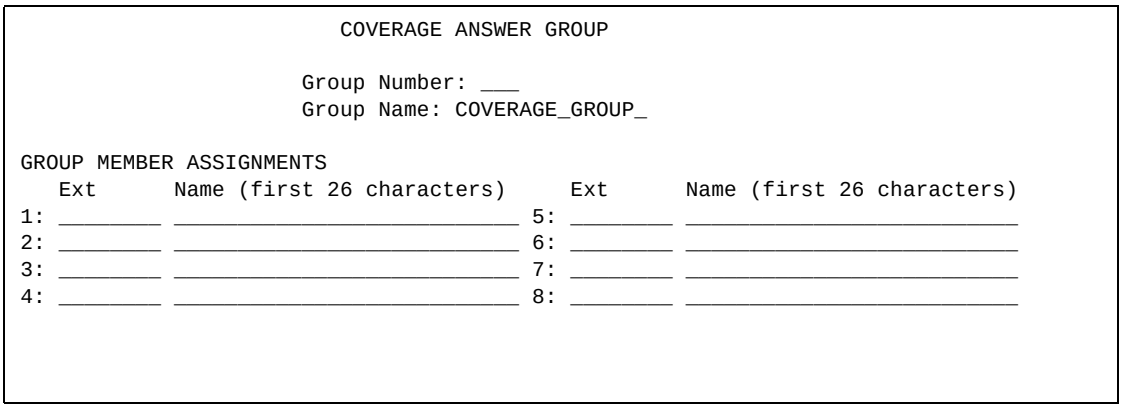
Instructions

To add a coverage answer group:

1. Type **add coverage answer-group next a**nd press **Enter**.

   The **Coverage Answer Group** screen appears.

Figure 68: Coverage Answer Group screen



```
                        COVERAGE ANSWER GROUP

                     Group Number: ___
                     Group Name: COVERAGE_GROUP_

GROUP MEMBER ASSIGNMENTS
   Ext      Name (first 26 characters)     Ext      Name (first 26 characters)
1: ________ ___________________________ 5: ________ ___________________________
2: ________ ___________________________ 6: ________ ___________________________
3: ________ ___________________________ 7: ________ ___________________________
4: ________ ___________________________ 8: ________ ___________________________
```

2. In the **Group Name** field, enter a name to identify the coverage group.
3. In the **Ext** field, type the extension of each group member.
4. Press **Enter** to save your new group list.

   The system automatically completes the **Name** field when you press **Enter**.

Related topics

See Assigning a coverage path to users on page 194 for instructions on assigning a coverage path.

Setting up Call Forwarding

This section explains how to administer various types of automatic call forwarding. To provide call forwarding to your users, assign each extension a class of service (COS) that allows call forwarding. Then assign call-forwarding buttons to the user phones (or give them the feature access code (FAC) for call forwarding) so that they can easily forward calls. You use the Station screen to assign the COS and any call-forwarding buttons.

Within each class of service, you can determine whether the users in that COS have the following call forwarding features:

- Call Forwarding All Calls — allows users to redirect all incoming calls to an extension, attendant, or external telephone number.
- Call Forwarding Busy/Don't Answer — allows users to redirect calls only if their extensions are busy or they do not answer.
- Restrict Call Fwd-Off Net — prevents users from forwarding calls to numbers that are outside your system network.

As the administrator, you can administer system-wide call-forwarding parameters to control when calls are forwarded. Use the **System Parameters Call Coverage/Call Forwarding** screen to set the number of times an extension rings before the system redirects the call because the user did not answer (CFWD No Answer Interval). For example, if you want calls to ring 4 times at an extension and, if the call is not answered, redirect to the forwarding number, set this parameter to 4.

You also can use the **System Parameters Call Coverage/Call Forwarding** screen to determine whether the forwarded-to telephone can override call forwarding to allow calls to the forwarded-from telephone (Call Forward Override). For example, if an executive forwards incoming calls to an attendant and the attendant needs to call the executive, the call can be made only if the **Call Forwarding Override** field is set to **y**.

Instructions

To determine which extensions have call forwarding activated:

1. Type **`list call-forwarding`** and press **Enter**.

   This command lists all the extensions that are forwarded along with each forwarding number.

   Note:

   If you have a V1, V2, or V3 system, you can see if a specific extension is forwarded only by typing **`status station`** ***nnnn***, where ***nnnn*** is the specific extension.

Related topics

See "Call Forwarding" in *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205.

Setting up call forwarding for users

This section shows you how to give your users access to call forwarding.

Instructions

We will change a call forwarding access code from a local telephone with a Class of Service of 1:

1. Type **change feature-access-codes** and press **Enter**.

   The **Feature Access Code** screen appears.

Figure 69: Feature Access Code screen

```
                                FEATURE ACCESS CODE (FAC)
         Abbreviated Dialing List1 Access Code: ____
         Abbreviated Dialing List2 Access Code: ____
         Abbreviated Dialing List3 Access Code: ____
Abbreviated Dial - Prgm Group List Access Code: ____
                      Announcement Access Code: ____
                       Answer Back Access Code: ____
      Auto Alternate Routing (AAR) Access Code: ____
      Auto Route Selection (ARS) Access Code 1: ____   Access Code 2: ____
                Automatic Callback Activation: ____    Deactivation: ____
Call Forwarding Activation Busy/DA: *70    All: *71     Deactivation: #72
                         Call Park Access Code: ____
                       Call Pickup Access Code: ____
CAS Remote Hold/Answer Hold-Unhold Access Code: ____
                  CDR Account Code Access Code: ____
                   Change Coverage Access Code: ____
                  Data Origination Access Code: ____
                      Data Privacy Access Code: ____
              Directed Call Pickup Access Code: ____
     Emergency Access to Attendant Access Code: ____
 Extended Call Fwd Activate Busy D/A: ____ All: ____    Deactivation: ____
```

2. In the **Call Forwarding Activation Busy/DA** field, type ***70**.

   The ***70** feature access code activates the call forwarding option so incoming calls forward when your telephone is busy or does not answer.

3. In the **Call Forwarding Activation All** field, type ***71**.

   The ***71** feature access code forwards all calls.

4. In the **Call Forwarding Deactivation** field, type **#72**.

   The **#72** feature access code deactivates the call forwarding option.

5. Press **Enter** to save your changes.

6. Type **change cos** and press **Enter**.

   The **Class of Service** screen appears.

Figure 70: Class of Service screen

```
                          CLASS OF SERVICE

                                  0 1 2 3 4 5 6 7 8 9 10 11 12 13 14 15

Auto Callback                     n y y n y n y n y n  y  n  y  n  y  n
Call Fwd-All Calls                n y n y y n n y y n  n  y  y  n  n  y
Data Privacy                      n y n n n y y y y n  n  n  n  y  y  y
Priority Calling                  n y n n n n n n n y  y  y  y  y  y  y
Console Permissions               n y n n n n n n n n  n  n  n  n  n  n
Off-hook Alert                    n n n n n n n n n n  n  n  n  n  n  n
Client Room                       n n n n n n n n n n  n  n  n  n  n  n
Restrict Call Fwd-Off Net         n y y y y y y y y y  y  y  y  y  y  y
Call Forward Busy/DA              n y n n n n n n n n  n  n  n  n  n  n
Personal Station Access           n n n n n n n n n n  n  n  n  n  n  n
Extended Forwarding All           n n n n n n n n n n  n  n  n  n  n  n
Extended Forwarding B/DA          n n n n n n n n n n  n  n  n  n  n  n
Trk-to-Trk Restriction Override n n n n n n n n n n  n  n  n  n  n  n
QSIG Call Offer Originations      n n n n n n n n n n  n  n  n  n  n  n
```

7. On the **Call Fwd-All Calls** line, in the **1** column, type **y**.

   This allows the user with this Class of Service to forward their calls. The **1** column is for phones with a Class of Service of 1.

8. On the **Console Permissions** line, in the **1** column, type **y**. This allows the user to define call forwarding on any station, not just the dialing station.

9. On the **Restrict Call Fwd-Off Net** line, in the **1** column, type **y**.

   This restricts your users from forwarding calls off-site. If you want your users to be able to call off-site, leave this field as **n**.

10. On the **Call Forward Busy/DA** line, in the **1** column, type **y**.

    This forwards a user's calls when the telephone is busy or doesn't answer after a programmed number of rings.

11. Press **1** to save your changes.

Allowing users to specify a forwarding destination

Now that you have set up system-wide call forwarding, have your users use this procedure if they want to change their call forwarding destination from their work (local) station.

1. They dial either their Call Forwarding Activation Busy/DA or Call Forwarding Activation All feature access code. If your users have buttons assigned, they press those buttons, listen for dial tone, and dial the digits.

   Note:

   Both Call Forwarding Activation Busy/DA or the Call Forwarding Activation All cannot be active for the same telephone at the same time.

   In this example, enter ***71** for **Call Forwarding Activation All**.

2. They dial their "forwarding-to" off-site or on-site number.

   In this example, enter **2081**. This is a local number; for off-site forwarding, include the AAR/ARS feature access code.

3. When they hear the 3-beep confirmation tone, they hang up.

Changing the forwarding destination remotely

Now that you have set up all of the required system administration for call forwarding, have your users use this procedure if they want to change their call forwarding destination from a telecommuting (off-site) telephone.

1. They dial their telecommuting extension.

   See Telecommuting Access on page 1503 for more information. In this example, enter **555-9126**.

2. When they get dial tone, they dial either their Extended Call Forward Activate Busy/DA or the Extended Call Forward Activate All feature access code.

   In this example, enter ***61** for the **Extended Call Forward Activate All** number.

3. When they get dial tone, they dial their extension number and press the ‘#’ key.

   In this example, enter **1014**, then #.

4. Even though there is no dial tone, they dial their security code and press the ‘#’ key.

   In this example, enter **4196**, then #.

5. When they get dial tone, they dial their "forwarding-to" off-site or on-site number.

   In this example, enter **9-555-2081**.

6. When they hear the 3-beep confirmation tone, they hang up.

Allowing users to change coverage remotely

This section shows you how to allow users to change their call coverage path from a local or telecommuting (off-site) telephone.

Instructions

To change call coverage from off-site:

1. Type **change feature-access-codes** and press **Enter**.

   The **Feature Access Code** screen appears.

2. In the **Change Coverage Access Code** field, type ***85**.

   Use the ***85** feature access code to change a coverage path from a telephone or remote station.

3. Press **Enter** to save your changes.

4. Type **change cor** and press **Enter**.

   The **Class of Restriction** screen appears.

5. In the **Can Change Coverage** field, type **y**.

   This permits users to select one of two previously administered coverage paths.

6. Press **Enter** to save your changes.

7. Type **change station 1014** and press **Enter**.

   The **Station** screen for extension 1014 appears.

8. In the **Security Code** field, type **4196**.

   In this example, this is your security code. See Security-Related System Parameters on page 1378 for more information on setting the length of your security code.

9. In the **Coverage Path 1** and **Coverage Path 2** fields, verify that both are defined enabling your user to move from one coverage path to another.

   The **t1** and **t2** are the numbers of the Time of Day Coverage Tables.

10. Press **Enter** to save your changes.

Setting up Night Service

You can use night service to direct calls to an alternate location when the primary answering group is not available. For example, you can administer night service so that anyone in your marketing department can answer incoming calls when the attendant is at lunch or has left for the day.

Once you administer night service to route calls, your end-users merely press a button on the console or a feature button on their phones to toggle between normal coverage and night service.

There are five types of night service:

- Night Console Night Service — directs all attendant calls to a night or day/night console
- Night Station Night Service — directs all incoming trunk or attendant calls to a night service destination
- Trunk Answer from Any Station (TAAS) — directs incoming attendant calls and signals a bell or buzzer to alert other employees that they can answer the calls
- Trunk Group Night Service — directs incoming calls to individual trunk groups to a night service destination
- Hunt Group Night Service — directs hunt group calls to a night service destination

Setting up night station service to voice mail

The night station service (also known as Listed Directory Number (LDN) Night Service) sends calls directed to an LDN to voice mail when the system is in night service.

Instructions

What is described below is a common setup; however, you can use a regular extension in this field, but it will not follow coverage.

Note:

You can use a dummy hunt group (one with no members) or an exported station with a coverage path. The instructions below use a hunt group.

To set up a night station service to voice mail:

1. Type **`add hunt-group next`** and press **Enter**.

   The **Hunt Group** screen appears.

Figure 71: Hunt Group screen

```
change hunt-group x                                          Page 1 of X
                                    HUNT GROUP

            Group Number: 5                                    ACD? n
              Group Name: ldn nights_____________________    Queue? n
         Group Extension: 51002                             Vector? n
              Group Type: _________                  Coverage Path: ____
                     TN: ___              Night Service Destination:
_____
                    COR: _                            MM Early Answer? _
           Security Code: ____
     ISDN Caller Display: _________
```

The **Group Number** field fills automatically with the next hunt group number.

2. In the **Group Name** field, type the name of the group.

   In our example, type **ldn nights**. There should be no members in this hunt group.

3. Press **Enter** to save your changes.

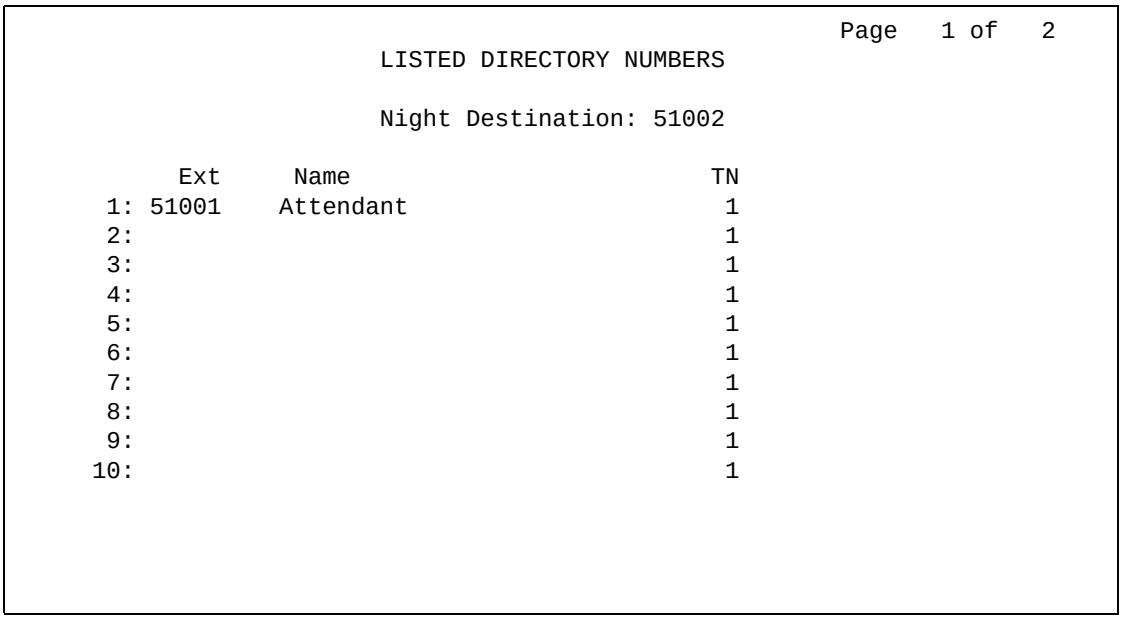
   Note:

   If you are using tenant partitioning, the command for the next step will be **change tenant x**. If you are using tenant partitioning, the **Night Destination** field does not appear on the **Listed Directory Numbers** screen. Instead, it is on the **Tenant** screen.

4. Type **change listed-directory-numbers** and press **Enter**.

   The **Listed Directory Numbers** screen appears.

Figure 72: Listed Directory Number screen



```
                                                        Page   1 of   2
                     LISTED DIRECTORY NUMBERS

                     Night Destination: 51002

      Ext     Name                          TN
 1: 51001    Attendant                      1
 2:                                         1
 3:                                         1
 4:                                         1
 5:                                         1
 6:                                         1
 7:                                         1
 8:                                         1
 9:                                         1
10:                                         1
```

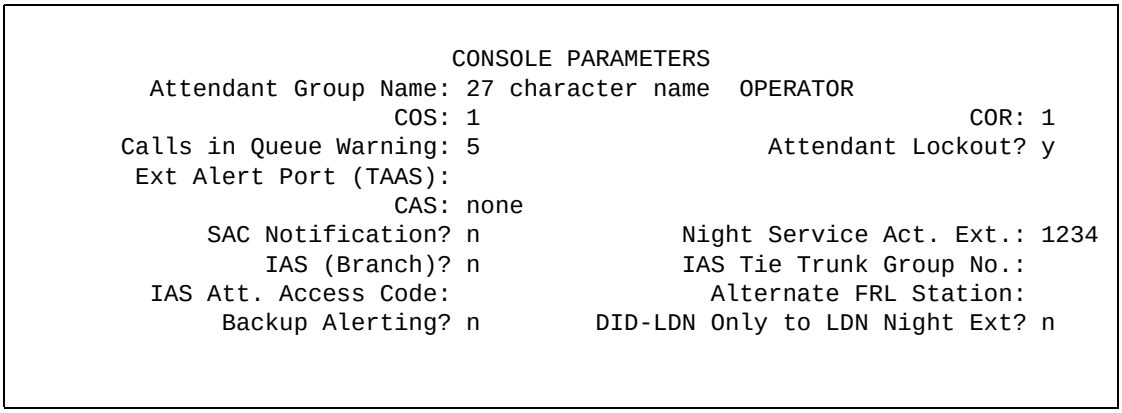
5. In the **Night Destination** field, add the night destination on the listed directory telephone.

   In our example, type **51002**.

6. Press **Enter** to save your changes.
7. Type **`change console-parameters`** and press **Enter**.

   The **Console Parameters** screen appears.

Figure 73: Console Parameters screen



```
                        CONSOLE PARAMETERS
  Attendant Group Name: 27 character name  OPERATOR
                   COS: 1                                  COR: 1
Calls in Queue Warning: 5                    Attendant Lockout? y
 Ext Alert Port (TAAS):
                   CAS: none
      SAC Notification? n              Night Service Act. Ext.: 1234
          IAS (Branch)? n              IAS Tie Trunk Group No.:
 IAS Att. Access Code:                   Alternate FRL Station:
       Backup Alerting? n        DID-LDN Only to LDN Night Ext? n
```

8. In the **DID-LDN Only** to **LDN Night Ext** field, type **n**.
9. Press **Enter** to save your changes.

10. From a telephone with console permissions, dial the call forwarding feature access code, then the hunt group's extension, followed by the main number of AUDIX.

    In our example, dial 51002.

    Note:

    You should receive the confirmation tone (3 beeps). This step is very important as calls to the LDN night service extension do not follow coverage.

11. In voice mail, build your auto attendant with the extension of the Listed Directory Number, not the hunt group.

    The originally dialed number was the LDN. That is what Communication Manager passes to the voice mail. In the case of the INTUITY and newer embedded AUDIX Voice Mail systems, you can use the Auto Attendant routing table to send the calls to a common Auto Attendant mailbox.

Setting up night console service

This section shows you how to set up night console service.

Night Console Service directs all calls for primary and daytime attendant consoles to a night console. When a user activates Night Console Service, the Night Service button for each attendant lights and all attendant-seeking calls (and calls waiting) in the queue are directed to the night console.

Note:

Activating night console service also puts trunk groups into night service, except those for which a night service button has been administered. See Setting up trunk answer from any stationSetting up trunk answer from any station on page 213 for more information.

To activate and deactivate Night Console Service, the attendant typically presses the **Night** button on the principal attendant console or designated console.

Only the principal console can activate night service. In the absence of any console, a telephone can activate night service.

Instructions

We will put the attendant console (attendant 2) in a night service mode.

To set up Night Console Service:

1. Type **`change attendant 2`** and press **Enter**.

   The **Attendant Console** screen appears.

Figure 74: Attendant Console screen

```
                          ATTENDANT CONSOLE 2

        Type: console           Name: 27 character attd cons name
   Extension: 1000             Group: 1                Auto Answer: none
Console Type: principal           TN: 1                Data Module? n
        Port: 01C1106            COR: 1          Disp Client Redir? n
                                 COS: 1           Display Language: english


DIRECT TRUNK GROUP SELECT BUTTON ASSIGNMENTS (Trunk Access Codes)
    Local Remote           Local Remote             Local Remote
 1: 9                   5:                       9:
 2: 82                  6:                      10:
 3:                     7:                      11:
 4:                     8:                      12:

HUNDREDS SELECT BUTTON ASSIGNMENTS
 1:           5:            9:           13:           17:
 2:           6:           10:           14:           18:
 3:           7:           11:           15:           19:
 4:           8:           12:           16:           20:
```

2. In the **Console Type** field, type **principal**.

   There can be only one night-only or one day/night console in the system unless you administer Tenant Partitioning. Night Service is activated from the principal console or from the one station set per-system that has a nite-serv button.

3. Press **Enter** to save your changes.

Setting up night station service

You can use night station service if you want to direct incoming trunks calls, DID-LDN (direct inward dialing-listed directory number) calls, or internal calls to the attendant (dialed ‘O’ calls) to a night service destination.

Let us say your attendant, who answers extension (LDN) 8100, usually goes home at 6:00 p.m. When customers call extension 8100 after hours, you would like them to hear an announcement that asks them to try their call again in the morning.

To set up night station service, you need to record the announcement (in our example, it is recorded at announcement extension 1234). See Managing Announcements on page 457 for information on setting up the announcement.

Tip:

All trunk groups that are routed through the attendant direct to this night service destination provided they already do not have a night service destination and, on the **Console Parameters** screen, the **DID-LDN Only to DID-LDN Night Ext** field is **n**. See Setting up trunk answer from any station on page 213.

Instructions

To set up night station service:

1. Type **change listed-directory-numbers** and press **Enter**.

   The **Listed Directory Numbers** screen appears.

Figure 75: Listed Directory Number screen

```
                        LISTED DIRECTORY NUMBERS

                      Night Destination: 1234

      Ext     Name                          TN
 1: 8100     attendant 8100
 2:
 3:
 4:
 5:
 6:
 7:
 8:
 9:
10:
```

2. Enter **1234** in the **Night Destination** field.

   The destination can be an extension, a recorded announcement extension, a vector directory number, or a hunt group extension.

3. Press **Enter** to save your changes.

4. Type **change console-parameters** and press **Enter**.

   The **Console Parameters** screen appears.

Figure 76: Console Parameters screen

```
                        CONSOLE PARAMETERS
    Attendant Group Name: 27 character name  OPERATOR
                     COS: 1                                  COR: 1
 Calls in Queue Warning: 5                     Attendant Lockout? y
  Ext Alert Port (TAAS):
                     CAS: none
       SAC Notification? n               Night Service Act. Ext.: 1234
           IAS (Branch)? n               IAS Tie Trunk Group No.:
   IAS Att. Access Code:                   Alternate FRL Station:
        Backup Alerting? n         DID-LDN Only to LDN Night Ext? n
Attendant Vectoring VDN:
```

5. In the **DID-LDN Only to LDN Night Extension** field, type **n**.
6. Press **Enter** to save your changes.

   After you set up night station service, have the attendant use the night console button to activate and deactivate night service.

Setting up trunk answer from any station

There may be situations where you want everyone to be able to answer calls when the attendant is away. Use trunk answer any station (TAAS) to configure the system so that it notifies everyone when calls are ringing. Then, you can give users the trunk answer any station feature access code so they can answer these calls.

When the system is in night service mode, attendant calls redirect to an alerting device such as a bell or a buzzer. This lets other people in the office know when they should answer the telephone.

Note:

If no one answers the call, the call will not redirect to night service.

We will define a feature access code (we'll use 71) and configure the alerting device for trunk answer any station.

Before you start

You need a ringing device and 1 port on an analog line circuit pack. See the *Hardware Guide for Avaya Communication Manager* for more information on the circuit pack.

Instructions

To set the feature access code for TAAS:

1. Type **change feature-access-codes** and press **Enter**.

   The **Feature Access Code** screen appears.

Figure 77: Feature Access Code (FAC) screen

```
change feature-access-codes                                  Page 3 of x
                           FEATURE ACCESS CODE (FAC)

                 Priority Calling Access Code:
                          Program Access Code:
       Refresh Terminal Parameters Access Code:
              Remote Send All Calls Activation:        Deactivation:
              Self Station Display Access Code:
                     Send All Calls Activation:        Deactivation:
         Station Firmware Download Access Code:
                       Station Lock Activation:        Deactivation:
     Station Security Code Change Access Code:
              Station User Admin of FBI Assign:              Remove:
  Station User Button Ring Control Access Code:
                         Telephone Activation: #*
             Terminal Dial-up Test Access Code:
Terminal Translation Initialization Merge Code:     Separation Code:
            Transfer to Voice Mail Access Code:
          Trunk Answer Any Station Access Code: 71
              User Control Restrict Activation:       Deactivation:
  Voice Coverage Message Retrieval Access Code:
 Voice Principal Message Retrieval Access Code: *80
```

2. In the **Trunk Answer Any Station Access Code** field, type **71**.
3. Press **Enter** to save your changes.

Once you set the feature access code, determine where the external alerting device is connected to the Communication Manager server (we'll use port 01A0702).

To set up external alerting:

1. Type **change console-parameters** and press **Enter**.

   The **Console Parameters** screen appears.

Figure 78: Console Parameters screen

```
change console-parameters                                    Page   1 of x
                              CONSOLE PARAMETERS
         Attendant Group Name: OPERATOR
                          COS: 0                                   COR: 0
       Calls in Queue Warning: 5                    Attendant Lockout? y
        Ext Alert Port (TAAS): 01A0702
                          CAS: none
                                                Night Service Act. Ext.:
                 IAS (Branch)? n                IAS Tie Trunk Group No.:
          IAS Att. Access Code:                   Alternate FRL Station:
               Backup Alerting? n         DID-LDN Only to LDN Night Ext? n
     Attendant Vectoring VDN:
```

2. In the **EXT Alert Port (TAAS)** field, type **01A0702**.

   Use the port address assigned to the external alerting device.

3. Press **Enter** to save your changes.

Setting up external alerting night service

Calls redirected to the attendant via Call Forwarding or Call Coverage will not go to the LDN Night Station. If there is no night station specified, and the TAAS bell is being used, these calls ring the TAAS bell. A call following the coverage path rings the TAAS bell for the number of times indicated in the Coverage Don't Answer Interval for Subsequent Redirection (Rings) field. If not answered, the call proceeds to the next point in the station's coverage path. If the call was sent to the Attendant by Call Forwarding, it continues to ring the TAAS bell.

When night service is enabled, and there is a night service destination on the LDN screen, calls covering to the attendant attempt to ring the night destination instead of the attendant position even if the handset is plugged in.

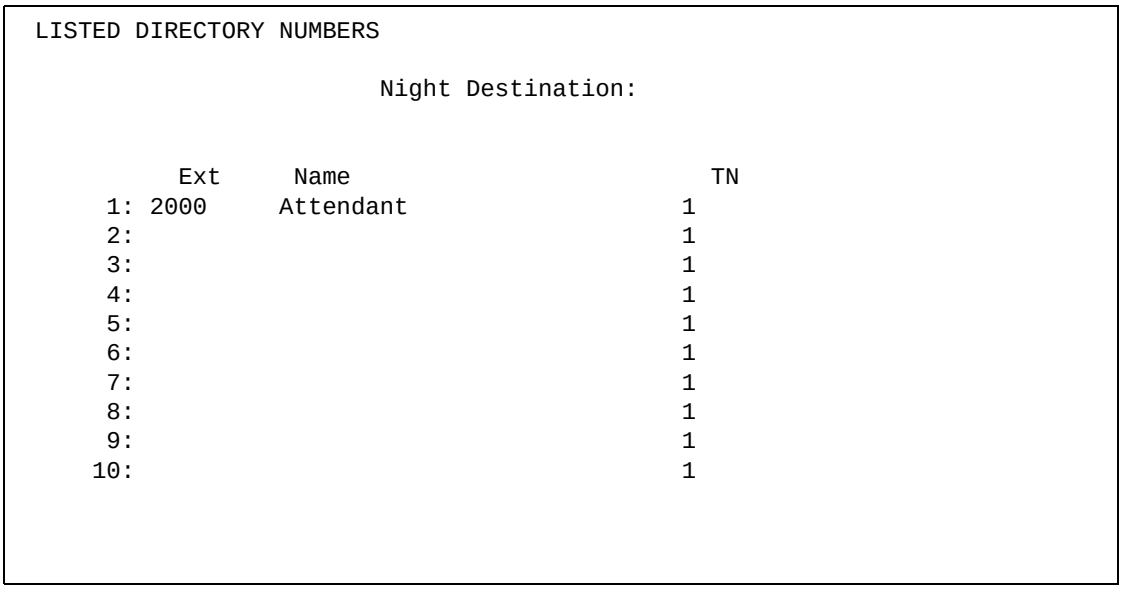
Instructions

To send LDN calls to the attendant during the day and to a guard's desk at night:

1. Type **`change listed-directory-numbers`** and press **Enter**.

   The **Listed Directory Numbers** screen appears.

Figure 79: Listed Directory Number screen

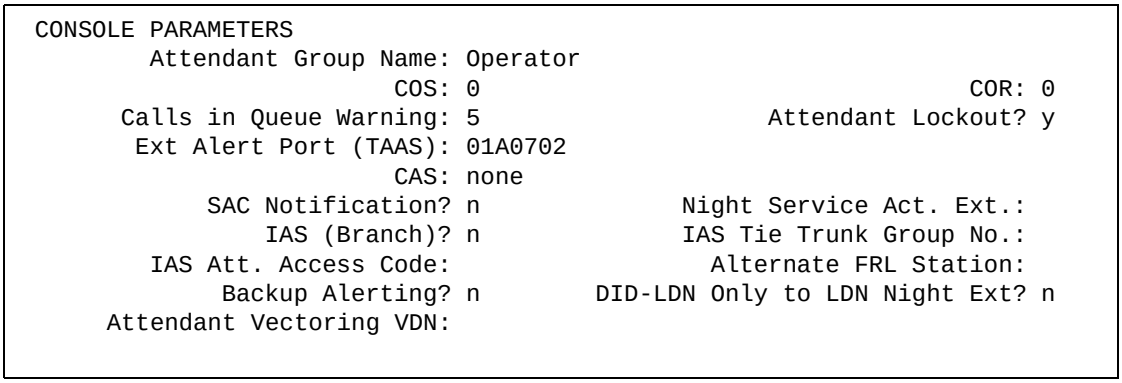


```
LISTED DIRECTORY NUMBERS

                        Night Destination:


          Ext     Name                          TN
     1: 2000     Attendant                    1
     2:                                       1
     3:                                       1
     4:                                       1
     5:                                       1
     6:                                       1
     7:                                       1
     8:                                       1
     9:                                       1
    10:                                       1
```

2. In the **Night Destination** field, verify this field is blank.
3. Press **Enter** to save your changes.
4. Type **change console-parameters** and press **Enter**.

   The **Console Parameters** screen appears.

Figure 80: Console Parameters screen



```
CONSOLE PARAMETERS
        Attendant Group Name: Operator
                         COS: 0                                  COR: 0
      Calls in Queue Warning: 5                    Attendant Lockout? y
       Ext Alert Port (TAAS): 01A0702
                         CAS: none
             SAC Notification? n              Night Service Act. Ext.:
                 IAS (Branch)? n              IAS Tie Trunk Group No.:
        IAS Att. Access Code:                   Alternate FRL Station:
             Backup Alerting? n        DID-LDN Only to LDN Night Ext? n
     Attendant Vectoring VDN:
```

5. In the **EXT Alert Port (TAAS)** field, type **01A0702**.

   This is the port address assigned to the external alerting device.

6. Press **Enter** to save your changes.

   The system is in Night Service.

   Any calls to extension 2000 now go to extension 3000 (the guard's desk).

   Any "0" seeking calls go to extension 3000 (the guard's desk).

To send LDN calls to the attendant during the day and to the TAAS bell at night:

1. Type **change console-parameters** and press **Enter**.

   The **Console Parameters** screen appears.

Figure 81: Console Parameters screen

```
                        CONSOLE PARAMETERS
     Attendant Group Name: Operator
                      COS: 0                                  COR: 0
  Calls in Queue Warning: 5                     Attendant Lockout? y
   Ext Alert Port (TAAS): 01A0702
                      CAS: none
          SAC Notification? n               Night Service Act. Ext.:
              IAS (Branch)? n               IAS Tie Trunk Group No.:
    IAS Att. Access Code:                     Alternate FRL Station:
           Backup Alerting? n          DID-LDN Only to LDN Night Ext? y
Attendant Vectoring VDN:
```

2. In the **DID-LDN Only to Night Ext**. field, type **y**.

   This allows only listed directory number calls (LDN) to go to the listed directory night service number extension.

3. In the **Ext Alert Port (TAAS)** field, type **01A070**.

   This is the port address assigned to the external alerting device.

4. Press **Enter** to save your changes.

   The system is in night service.

   Any DNIS extension 2000 calls now go to the TAAS bell.

   Any "0" seeking calls now go to the TAAS bell.

Setting up trunk group night service

You can use trunk group night service if you want to direct individual trunk groups to night service. The system redirects calls from the trunk group to the group's night service destination.

Trunk group night service overrides night station service. For example, we will say you activate trunk group night service, and then your attendant activates night station service. In this case, calls to the trunk group use the trunk night service destination, rather than the station night service destination.

Instructions

We will direct night calls for trunk group 2 to extension 1245.

To set up trunk group night service:

1. Type **`change trunk-group 2`** and press **Enter**.

   The **Trunk Group** screen appears.

Figure 82: Trunk Group screen

```
                               TRUNK GROUP

 Group Number: 2                    Group Type: co           CDR Reports: y
   Group Name: outside calls   COR: 1_       TN: 1__     TAC: ____
    Direction: two-way_       Outgoing Display? n
  Dial Access? n                Busy Threshold: 99      Night Service: 1245
 Queue Length: 0                  Country: 1_    Incoming Destination: ________
    Comm Type: voice              Auth Code? n  Digit Absorption List: _
     Prefix-1? y                   Trunk Flash? n     Toll Restricted? y
          BCC: _
 TRUNK PARAMETERS
             Trunk Type: loop-start
     Outgoing Dial Type: tone                              Cut-Through? n
      Trunk Termination: rc                    Disconnect Timing(msec): 500_

             Auto Guard? n    Call Still Held? n     Sig Bit Inversion: none
      Analog Loss Group: ___                        Digital Loss Group: ___
                                  Trunk Gain: high
             Bit Rate: 1200          Synchronization: _____    Duplex: ____
Disconnect Supervision - In? y  Out? n
  Answer Supervision Timeout: 10            Receive Answer Supervision? n
```

2. Type **1245** in the **Night Service** field.

   The destination can be a station extension, a recorded announcement extension, a vector directory number, a hunt group extension, a terminating extension group, or attd if you want to direct the call to the attendant.

3. Press **Enter** to save your changes.

Setting up night service for hunt groups

You can administer hunt group night service if you want to direct hunt group calls to a night service destination.

Let us say your helpline on hunt group 3 does not answer calls after 6:00 p.m. When customers call after hours, you would like them to hear an announcement that asks them to try their call again in the morning.

Instructions

To set up night service for your helpline, you need to record the announcement (in our example, the announcement is on extension 1234) and then modify the hunt group to send calls to this extension.

To administer the hunt group for night service:

1. Type **change hunt-group 3** and press **Enter**.

   The **Hunt Group** screen appears for hunt group 3.

Figure 83: Hunt Group screen

```
                                HUNT GROUP

           Group Number: 3                                ACD? n
             Group Name: Accounting                     Queue? y
        Group Extension: 2011                         Vector? n
             Group Type: ucd-mia                  Coverage Path: 1
                     TN: 1              Night Service Destination: 1234____
                    COR: 1                          MM Early Answer? n
          Security Code: ____
       ISDN Caller Disp: ________

           Queue Length: 4
Calls Warning Threshold: ___   Port: x      Extension: _______
 Time Warning Threshold: ___   Port: x      Extension: _______
```

2. In the **Night Service Destination** field, type **1234**.

   The destination can be an extension, a recorded announcement extension, a vector directory number, a hunt group extension, or attd if you want to direct calls to the attendant.

   Calls to hunt group 3 will follow the coverage path assigned to extension 1234.

3. Press **Enter** to save your changes.

4. Now you need to program a night service button.

   See Adding Feature Buttons on page 121 for more information.

Related topics

Managing Hunt Groups on page 228.

How do night service types interact?

Here is an example of how several types of night service might be used in one company.

Assume that you already administered the following night service settings:

- Night station night service redirects to extension 3000 and DID-LDN only to LDN Night Ext is set to **n**
- EXT Alert Port (TAAS) field is not defined
- Trunk group 4 redirects to extension 2000

Let us look at how calls for this company are directed after hours:

call type	directs to
An LDN call on a DID trunk	extension 3000
A call on trunk group 4	extension 2000
An internal call to ‘0’	extension 3000
A call that redirects to the attendant through a coverage path	the attendant queue

Adding Call Pickup

To give your users the ability to pickup other users' calls, you may want to use Call Pickup. To do this, you need to define a call pickup group.

Users may want to be able to pick up a call that is ringing at a nearby desk. Call Pickup provides 3 ways to pick up calls ringing at another telephone:

- With Call Pickup, you create a call pickup group. All group members can answer a call ringing at another telephone in the group from their own telephone. If more than one telephone is ringing, the one that has been ringing the longest is picked up.
- With Directed Call Pickup, users specify which ringing telephone they want to answer from their own telephone. A call pickup group is not required.
- With Group Call Pickup, users within an "extended" group can answer calls outside of their immediate group by entering a feature access code (FAC) followed by the 1- or 2-digit pickup (index) number.

Creating pickup groups

A pickup group is a list of phones where each member of the group can answer another member's calls. For example, if you want everyone in the payroll department to be able to answer calls to any payroll extension (in case someone is away from their desk), create a pickup group that contains all of the payroll extensions. Members of a pickup group should be located in the same area so that they can hear when the other extensions in the group ring.

Note that each extension may belong to only one pickup group. Also, the maximum number of pickup groups may be limited by your system configuration.

To create a pickup group:

1. Type **`add pickup-group next`** and press **Enter**.

   The **Pickup Group** screen appears. The system selects the next Group Number for the new pickup group.

Figure 84: Pickup Group screen

```
                                PICKUP GROUP

                   Group Number:          Extended Group Number: ____

GROUP MEMBER ASSIGNMENTS

   Ext      Name (first 26 characters)      Ext      Name (first 26 char.)
 1:                                        14:
 2:                                        15:
 3:                                        16:
 4:                                        17:
 5:                                        18:
 6:                                        19:
 7:                                        20:
 8:                                        21:
 9:                                        22:
10:                                        23:
11:                                        24:
12:                                        25:
13:
```

2. Enter the extension of each group member.

   Up to 50 extensions can belong to one group.

3. Press **Enter** to save your new group list.

   The system automatically completes the name field when you press **Enter** to save your changes.

Once you define a pickup group, you can assign call-pickup buttons for each telephone in the group or you can give each member the call-pickup feature-access code. Use the **Station** screen to assign call-pickup buttons.

To allow users to answer calls that are not in their pickup group, you may be able to use Directed Call Pickup.

Setting up directed call pickup

To set up a telephone so that the user can pick up calls with Directed Call Pickup, you need to determine if directed call pickup is enabled on your system and make sure that the user's telephone has a COR that allows directed call pickup.

To determine if Directed Call Pickup is enabled on your system:

1. Type **change system-parameters features** and press **Enter**.

   The **Feature-Related System Parameters** screen appears.

2. Move to the **Directed Call Pickup?** field and enter **y**.
3. Press **Enter** to save the changes.

Now we will modify extension 4444 to allow directed call pickup. For this example, assume that the **Can Use Directed Call Pickup** field for COR 5 is set to Y.

1. Type **change station 4444** and press **Enter**.

   The **Station** screen appears.

2. In the **COR** field, enter **5**.
3. Press **Enter** to save your changes.

Setting up "simple" extended group pickup

Let us add a pickup group to an existing extended group where all members of the extended group pick up each other's calls. We will add pickup group 6 to the existing extended group 56.

To create a simple extended pickup group:

1. Type **change system-parameters features** and press **Enter**.

   The **Feature-Related System Parameters** screen appears.

2. In the **Extended Group Call Pickup** field, type **simple**.

   Permits feature access codes to be administered.

3. Press **Enter** to save your changes.
4. Type **change feature-access-codes** and press **Enter**.

   The **Feature Access Code** screen appears.

5. In the **Extended Group Call Pickup Access Code** field, type the desired FAC.

   See the dial plan to enter the correct sequence and number of digits.

6. Press **Enter** to save your changes.
7. Type **add pickup-group next** and press **Enter**.

   The **Pickup Group** screen appears.

Figure 85: Pickup Group screen

```
                                PICKUP GROUP

                  Group Number: 1        Extended Group Number: ____

GROUP MEMBER ASSIGNMENTS

   Ext      Name (first 26 characters)         Ext     Name (first 26 char.)
 1:51001    station  51001                  14:51001    station  51002
 2:                                         15:
 3:                                         16:
 4:                                         17:
 5:                                         18:
 6:                                         19:
 7:                                         20:
 8:                                         21:
 9:                                         22:
10:                                         23:
11:                                         24:
12:                                         25:
13:
```

8. Enter the primary extensions of the users desired in the pickup group. Administer all the pickup groups that will be grouped together in the same extended pickup group.
9. Type **change extended-pickup-group *n***, where ***n*** is the number of the extended pickup group to change. Press **Enter**.

   The **Extended Pickup Group** screen appears.

Figure 86: Extended Pickup Group screen

```
change extended-pickup-group 1                                Page   1 of   2
                              EXTENDED PICKUP GROUP

                            Extended Group Number: 56

Pickup   Pickup Group                          Pickup   Pickup Group
Number     Number                              Number     Number
 0:        345                                 13:      _____
 1:         25                                 14:      _____
 2:         2                                  15:      _____
 3:       _____                                16:      _____
 4:       _____                                17:      _____
 5:       _____                                18:      _____
 6:       _____                                19:      _____
 7:       _____                                20:      _____
 8:       _____                                21:      _____
 9:       _____                                22:      _____
10:       _____                                23:      _____
11:       _____                                24:      _____
12:       _____                                25:      _____
```

10. In the **Pickup Group Number** column, enter the pickup group number of the pickup groups that belong to the extended group being administered.

    The Pickup Number associated with the entered Pickup Group Number is the number users will enter following the FAC when pickup group. In the above listed example, a user in Pickup Group 2 will enter "0" following the FAC to pick up a call from pickup group 345.

    To minimize the number of digits entered by users, the Pickup Numbers should be the lowest available Pickup Numbers.

11. Press **Enter** to save your changes.

Setting up "flexible" extended group pickup

The **Extended Pickup Group** screen groups together the pickup groups that can have their calls picked up by the pickup groups with identical extended pickup group numbers. The pickup groups numbers are listed using the lowest available Pickup Numbers. A unique Extended Pickup Group number can be assigned to each pickup group on this screen. The entries in the Extended Group are the groups that can have calls picked up by the group with the matching Extended Group number in the Pickup Group screen. Thus, a particular group's calls may be picked up by members of many extended pickup groups.

Note:

On the **Feature-Related System Parameters** screen, if the **Extended Group Call Pickup** field is **flexible**, you can administer a pickup group in multiple extended pickup groups.

To create a flexible extended pickup group:

1. Type **change system-parameters features** and press **Enter**.

   The **Feature-Related System Parameters** screen appears.

2. In the **Extended Group Call Pickup** field, type **flexible**.
3. Press **Enter** to save your changes.
4. Type **change feature-access-codes** and press **Enter**.

   The **Feature Access Code** screen appears.

5. In the **Extended Group Call Pickup Access Code** field, type the desired FAC.

   See the dial plan to enter the correct sequence and number of digits.

6. Press **Enter** to save your changes.
7. Type **add pickup-group next** and press **Enter**.

   The **Pickup Group** screen appears.

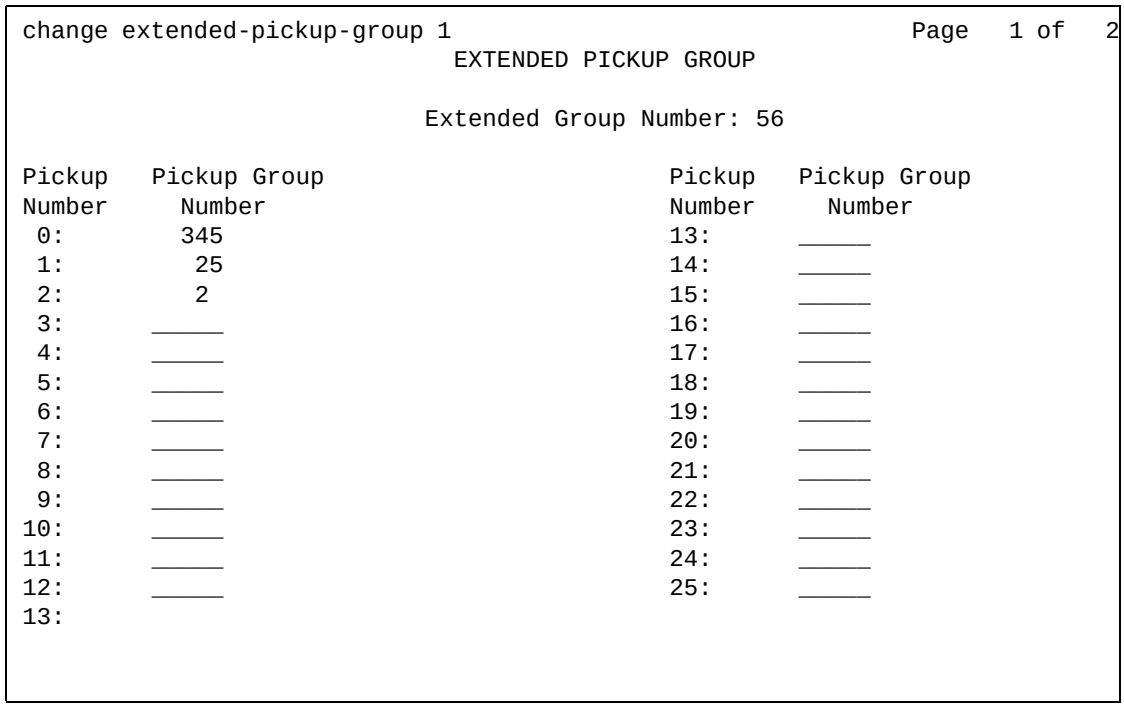
8. In the **Extended Group Number** field, type the number of the extended pickup group that can answer calls to this group.

   For each administered pickup group, indicate an extended group by entering the extended pickup group number. This extended pickup group number identifies the other pickup groups that can have calls picked up by the pickup group.

9. Add extended members as for other pickup features.
10. Type **change extended-pickup-group *n*** , where ***n*** is the number of the extended pickup group to change, and press **Enter**.

    The **Extended Pickup Group** screen appears.

Figure 87: Extended Pickup Group screen



```
change extended-pickup-group 1                                Page   1 of   2
                              EXTENDED PICKUP GROUP

                            Extended Group Number: 56

Pickup   Pickup Group                          Pickup   Pickup Group
Number     Number                              Number     Number
 0:        345                                 13:      _____
 1:         25                                 14:      _____
 2:         2                                  15:      _____
 3:       _____                                16:      _____
 4:       _____                                17:      _____
 5:       _____                                18:      _____
 6:       _____                                19:      _____
 7:       _____                                20:      _____
 8:       _____                                21:      _____
 9:       _____                                22:      _____
10:       _____                                23:      _____
11:       _____                                24:      _____
12:       _____                                25:      _____
13:
```

11. In the **Pickup Group Number** column, type the pickup group numbers of the other pickup groups that can answer each groups's calls.
12. Press **Enter** to save your changes.

Managing Hunt Groups

This section shows you how to set up hunt groups. It explains how calls to a hunt group are handled and shows you different call distribution methods.

What are hunt groups?

A hunt group is a group of extensions that receive calls according to the call distribution method you choose. When a call is made to a certain telephone number, the system connects the call to an extension in the group.

Use hunt groups when you want more than one person to be able to answer calls to the same number. For example, set up a hunt group for:

- a benefits department within your company
- a travel reservations service

Setting up hunt groups

Let us set up a hunt group for an internal helpline. Before making changes to Communication Manager, we'll decide:

- the telephone number for the hunt group
- the number of people answering calls
- the way calls are answered

Our dial plan allows 4-digit internal numbers that begin with 1. The number 1200 is not in use. So, we'll set up a helpline hunt group so anyone within the company can call extension 1200 for help with a telephone.

We will assign 3 people (agents) and their extensions to our helpline. We want calls to go to the first available person.

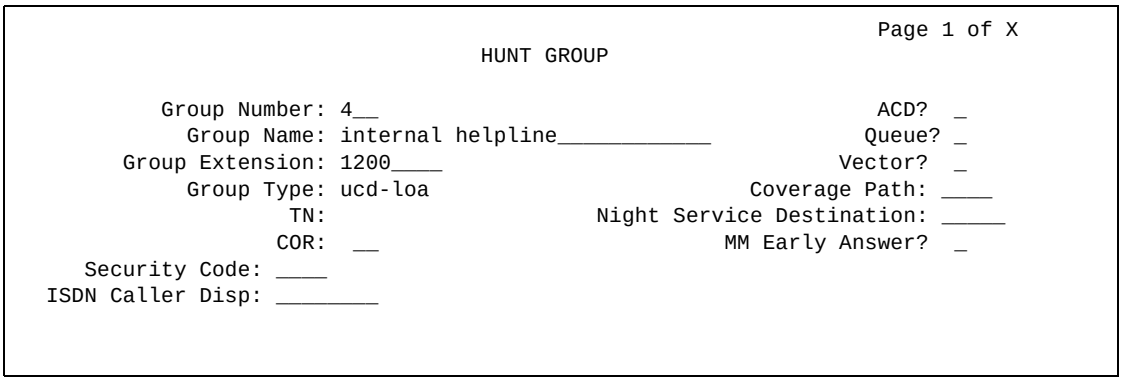
Instructions

To set up our helpline hunt group:

1. Type **`add hunt-group next`** and press **Enter**.

   The **Hunt Group** screen appears. The **Group Number** field is automatically filled in with the next hunt group number.

Figure 88: Hunt Group screen



2. In the **Group Name** field, type the name of the group.

   In our example, type **internal helpline**.

3. In the **Group Extension** field, type the telephone number.

   We'll type **1200**.

4. In the **Group Type** field, type the code for the call distribution method you choose.

   We'll type **ucd-loa** so a call goes to the agent with the lowest percentage of work time since login.

   Note:

   The COS for all hunt groups defaults to 1. Therefore, any changes to COS 1 on the **Class of Service** screen changes the COS for all your hunt groups. A **COS** field does not appear on the **Hunt Group** screen.

5. Press **Next Page** to find the **Group Member Assignments** screen.

Figure 89: Group Member Assignments screen

```
 HUNT GROUP
          Group Number: 4 Group Extension: 3001        Group Type: ucd
  Member Range Allowed: 1 - 1500 Administered Members (min/max): 1  /9
                                                Total Administered Members: 9
GROUP MEMBER ASSIGNMENTS
      Ext      Name (24 characters)            Ext       Name (24 characters)

 1 :1011                                    14 :
 2 :1012                                    15 :
 3 :1013                                    16 :
 4 :                                        17 :
 5 :                                        18 :
 6 :                                        19 :
 7 :                                        20 :
 8 :                                        21 :
 9 :                                        22 :
10 :                                        23 :
11 :                                        24 :
12 :                                        25 :
13 :                                        26 :

More Members Exist
```

6. In the **Ext** field, type the extensions of the agents you want in the hunt group.

   We'll type **1011**, **1012**, and **1013**.

   Tip:

   For a ddc group type (also known as "hot seat" selection), the call is sent to the extension listed in the first **Ext** field. The system uses this screen to determine the hunting sequence.

7. Press **Enter** to save your changes.

   The **Name** fields are display-only and do not appear until the next time you access this hunt group.

Dynamic hunt group queue slot allocation

The dynamic hunt group queue slot allocation feature eliminates the need to preallocate queue slots for hunt groups. The system dynamically allocates the queue slots from a common pool on an as-needed basis. All possible calls can be queued. There is no additional administration needed. This feature expands the capacities of your system by eliminating the potential of missed calls due to a full queue.

When the **Queue?** field on the **Hunt Group** screen is set to **y**, this feature applies to all uses of hunt groups:

- Automatic Call Distribution (ACD) non-vector/vector splits and skills
- Non-ACD hunt group
- Voice mail

Related topics

See Hunt Group on page 1077 for more information on an ACD and non-ACD hunt group.

Changing a hunt group

To make changes to a hunt group:

1. Type **change hunt-group *n***, where ***n*** is the number of the hunt group, and press **Enter**.
2. Change the necessary fields.
3. Press **Enter** to save your changes.

Setting up a queue

You can tell your server running Communication Manager how to handle a hunt-group call when it cannot be answered right away. The call waits in "queue."

We will tell Communication Manager that as many as 10 calls can wait in the queue, but that you want to be notified if a call waits for more than 30 seconds.

You also want Communication Manager to send a warning when 5 or more calls are waiting in the queue. This warning flashes queue-status buttons on phones that have a status button for this hunt group. When the buttons flash, everyone answering these calls can see that the help-line calls need more attention.

Instructions

To set up our helpline queue:

1. Type **change hunt-group *n***, where ***n*** is the number of the hunt group to change. Press **Enter**.

   In our example, type **change hunt-group 5**. The **Hunt Group** screen appears.

Figure 90: Hunt Group screen

```
                                                          Page 1 of X
                                   HUNT GROUP

           Group Number: 4__                                 ACD?
             Group Name: internal helpline_____________      Queue?
        Group Extension: 1200____                          Vector?
             Group Type: ucd-loa                    Coverage Path: ____
                     TN:                  Night Service Destination: _____
                    COR:                            MM Early Answer?
     Security Code: ____
  ISDN Caller Disp: ________


                 Measured: ________          Supervisor: Extension: ____

      Controlling Adjunct: ____
   Multiple Call Handling: ___________
                Objective: ____

           Queue Length: 10
Calls Warning Threshold: 5            Calls Warning Port: __
 Time Warning Threshold: 30            Time Warning Port: __

                  Redirect on No Answer (rings): _  Redirect to VDN: _
              Forced Entry of Stroke Counts or Call Work Codes? _
```

2. In the **Queue** field, type **y**.
3. In the **Queue Length** field, type the maximum number of calls that you want to wait in the queue.

   In our example, type **10**.
4. In the **Calls Waiting Threshold** field, type the maximum number of calls that can be in the queue before the system flashes the queue status buttons.

   In our example, type **5**.
5. In the **Time Warning Threshold** field, type the maximum number of seconds you want a call to wait in the queue before the system flashes the queue status buttons.

   In our example, type **30**.
6. Press **Enter** to save your changes.

Setting up hunt groups for TTY callers

Several laws, such as the Americans with Disabilities Act (ADA) of 1990 and Section 255 of the Telecommunications Act of 1996, require that "reasonable accommodation" be provided for people with disabilities. For this reason, your company may choose to offer support for callers who use TTYs. (These devices are also known as TDDs -- "Telecommunication Device for the Deaf" -- but the term TTY is generally preferred, in part because many users of these devices are hearing-impaired, but not deaf.)

TTY callers can be accommodated by creating a hunt group that includes TTY-equipped agents. The TTY itself looks a little like a laptop computer, except that it has a one- or two-line alphanumeric display instead of a computer screen. The cost of a typical TTY is approximately three hundred dollars. Although many TTYs can connect directly with the telephone network via analog RJ-11 jacks, Avaya recommends that agents be equipped with TTYs that include an acoustic coupler that can accommodate a standard telephone handset. One reason for this recommendation is that a large proportion of TTY users are hearing impaired, but still speak clearly. These individuals often prefer to receive calls on their TTYs and then speak in response. This requires the call center agent to alternate between listening on the telephone and then typing on the TTY, a process made considerably easier with an acoustically coupled configuration.

Although TTY-emulation software packages are available for PCs, most of these do not have the ability to intermix voice and TTY on the same call.

For a TTY hunt group, you can record TTY announcements and use them for the hunt group queue. To record announcements for TTY, simply follow the same steps as with voice recordings from your telephone (see Managing Announcements on page 457). However, instead of speaking into your telephone to record, you type the announcement with the TTY device.

Note:

For an alternative to simply creating a TTY hunt group, you can use vectors to process TTY calls. With vectors, you can allow TTY callers and voice callers to use the same telephone number. In this case, you can also record a single announcement that contains both TTY signaling and a voice recording.

Adding hunt group announcements

You can add recorded announcements to a hunt group queue. Use announcements to encourage callers to stay on the line or to provide callers with information. You can define how long a call remains in the queue before the caller hears an announcement.

See Recording Announcements on page 461 for information on how to record an announcement.

Let us add an announcement to our internal helpline. We want the caller to hear an announcement after 20 seconds in the queue, or after approximately 4 or 5 rings. Our announcement is already recorded and assigned to extension 1234.

Tip:

You can use **`display announcements`** to find the extensions of your recorded announcements.

Instructions

To add an announcement to our helpline queue:

1. Type **`change hunt-group`** ***n*** and press **Enter**, where n is the number of the hunt group to change.

   In our example, type **`change hunt-group 5`**.

   The **Hunt Group** screen appears.

2. Press **Next Page** to find the **First Announcement Extension** field.

Figure 91: Hunt Group screen

```
                              HUNT GROUP

                          Message Center: _________
                         AUDIX Extension: _________
                Message Center AUDIX Name: _______
                                  Primary? _
     Calling Party Number to INTUITY AUDIX? _
                            LWC Reception: ________
                               AUDIX Name: ________
                    Messaging Server Name: ________

             First Announcement Extension: 1234 ___ Delay (sec): 20
            Second Announcement Extension: _________ Delay (sec): __ Recurring? _
```

3. In the **First Announcement Extension** field, type the extension of the announcement you want callers to hear.

   In this example, type **1234**.

4. In the **First Announcement Delay (sec)** field, type the number of seconds you want the caller to wait before hearing the first announcement.

   In our example, type **20**.

   Tip:

   If you set the delay announcement interval to 0, callers automatically hear the announcement before anything else. This is called a "forced first announcement."

5. Press **Enter** to save your changes.

You can use the same announcement for more than one hunt group.

Managing Vectors and VDNs

This section provides an introduction to vectors and Vector Directory Numbers (VDN). It gives you basic instructions for writing simple vectors.

SECURITY ALERT:

Vector fraud is one of the most common types of toll fraud because vectors route calls based on the Class of Restriction (COR) assigned to the VDN. See *BCS Products Security Handbook* for more information.

This section references announcements, hunt groups, queues, splits, and skills, which are covered in detail in other sections of this book. You can also find information about these topics in the *Avaya Communication Manager Contact Center Call Vectoring and Expert Agent Selection (EAS) Guide*.

Note:

The **Client Room** field on the **Class of Service** screen will affect VDN displays. If a local station that has a COS with the **Client Room** field set to **y** calls a local VDN, the agent's display that receives the call will look as if it is a direct station call rather than the expected VDN display of "station name to vdn name."

What are vectors?

A vector is a series of commands that you design to tell the system how to handle incoming calls. A vector can contain up to 32 steps and allows customized and personalized call routing and treatment. Use call vectoring to:

- play multiple announcements
- route calls to internal and external destinations
- collect and respond to dialed information

Tip:

The vector follows the commands in each step in order. The vector "reads" the step and follows the command if the conditions are correct. If the command cannot be followed, the vector skips the step and reads the next step.

Your system can handle calls based on a number of conditions, including the number of calls in a queue, how long a call has been waiting, the time of day, day of the week, and changes in call traffic or staffing conditions.

Writing vectors

Writing vectors is easy, but Avaya recommends that you set up and test your vectors before you use them across the system.

We'll write a vector to handle calls to our main number. It is the first vector so we'll use number 1.

Tip:

Use **`list vector`** to see a list of existing vectors.

Before you start

- On the **System-Parameters Customer-Options** screen, ensure that the **Basic Call Vectoring** field is **y**. If not, contact your Avaya representative.
- To provide announcements, you need an Announcement circuit pack. For more information on the circuit pack, see the *Hardware Guide for Avaya Communication Manager,* 555-245-207.

- Use one of the following:
  - Tone Clock with Call Classifier - Tone Detector circuit pack.
  - Call Classifier - Detector circuit pack.
- Note on adding a Meet-Me Conference vectors: If the vector for Meet-Me conferencing allows a new party to join a conference immediately, and that party is joining as an H.323 ip trunk user, the caller may not have talkpath with the others in the conference. To prevent this, include in the vector a short delay before a new party joins the Meet-Me conference, such as a step to collect digits, a 1-second delay, or play an announcement. Since Meet-Me vectors are always configured with announcements and digit collections, this should rarely be an issue.

Instructions

To write a vector:

1. Type **`change vector 1`** and press **Enter**.

   The **Call Vector** screen appears.

Figure 92: Call Vector screen

```
                              CALL VECTOR
  Number:  1       Name: main number calls
             Attendant Vectoring? n     Meet-me Conf? y           Lock? n
    Basic? y  EAS? n   G3V4 Enhanced? y    ANI/II-Digits? y   ASAI Routing? n
Prompting? y  LAI? n  G3V4 Adv Route? y    CINFO? y   BSR? n      Holidays? n

 01 _____________
 02 _____________
 03 _____________
 04 _____________
 05 _____________
 06 _____________
 07 _____________
 08 _____________
 09 _____________
 10 _____________
 11 _____________
```

The vector **Number** field on the left side of the screen is filled in automatically.

2. In the **Name** field, type a description for the vector.

   In our example, type **main number calls**.

   Tip:

   The information in the heading of the **Call Vector** screen is display only. Use the **`display system-parameters customer-options`** command to see the features that are turned on for your Communication Manager server.

3. Type your vector steps in the numbered column on the left of the screen.

   Tip:

   When you type in your vector steps, Communication Manager automatically completes some of the vector step information for you. For example, if you type "q" in a vector step field, it fills in "queue-to." Also, additional fields appear when you complete a field and press **Tab**. This makes it very easy to type in your vector steps.

Now that vector 1 is set up, we will add a vector step to it to tell Communication Manager how to handle the calls to our main number.

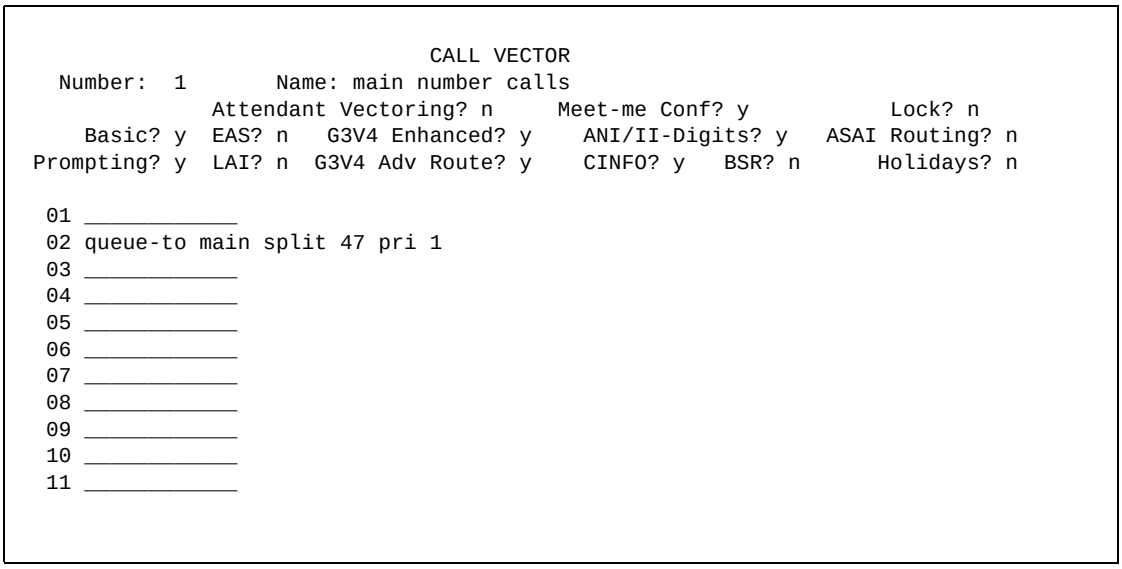
Putting a call in a queue

Write a vector so that calls that come into the main business number redirect to a queue.

We will use a vector-controlled hunt group for the main number queue. This hunt group was set up as main split 47. When calls first arrive, all calls to our main number should be queued as "pri 1" for low priority.

To queue calls, write the following vector (step 2). (Please note, we started our example on step 2 because step 1 is used later in this chapter.)

Figure 93: Call Vector screen



```
                              CALL VECTOR
  Number:  1       Name: main number calls
             Attendant Vectoring? n     Meet-me Conf? y           Lock? n
    Basic? y  EAS? n   G3V4 Enhanced? y    ANI/II-Digits? y   ASAI Routing? n
Prompting? y  LAI? n  G3V4 Adv Route? y    CINFO? y   BSR? n      Holidays? n

 01 _____________
 02 queue-to main split 47 pri 1
 03 _____________
 04 _____________
 05 _____________
 06 _____________
 07 _____________
 08 _____________
 09 _____________
 10 _____________
 11 _____________
```

Tip:

Remember, Communication Manager automatically fills in some of the information when you type your vector step and press **Tab**.

Playing an announcement

Write a vector to play an announcement for callers in a queue. Use the announcement to ask callers to wait. You need to record the announcement before the vector can use it.

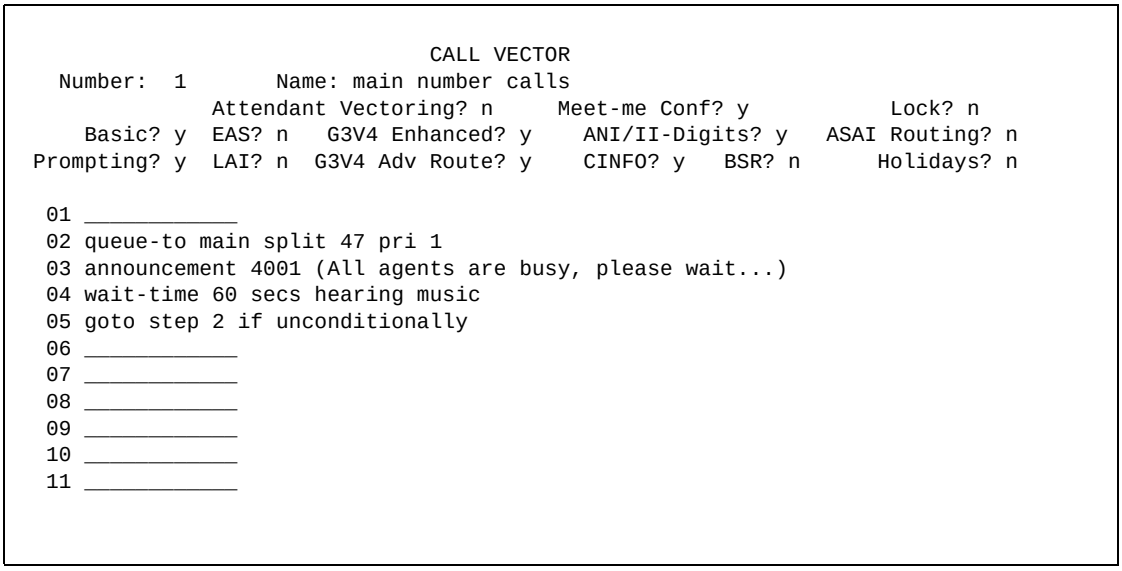
Let us play our announcement 4001, asking the caller to wait, then play music for 60 seconds, then repeat the announcement and music until the call is answered. The **goto** command creates the loop to repeat the announcement and the music. **Unconditionally** means under all conditions.

Tip:

Rather than loop your vectors directly back to the announcement step, go to the previous queue-to step. This way, if for some reason the call does not queue the first time, Communication Manager can attempt to queue the call again. If the call successfully queued the first time though, it merely skips the queue-to step and plays the announcement. The system cannot queue a call more than once in the exact same priority level.

To play and repeat an announcement, write this vector (steps 3-5):

Figure 94: Call Vector screen



```
                              CALL VECTOR
  Number:  1       Name: main number calls
             Attendant Vectoring? n     Meet-me Conf? y           Lock? n
    Basic? y  EAS? n   G3V4 Enhanced? y    ANI/II-Digits? y   ASAI Routing? n
Prompting? y  LAI? n  G3V4 Adv Route? y    CINFO? y   BSR? n      Holidays? n

 01 _____________
 02 queue-to main split 47 pri 1
 03 announcement 4001 (All agents are busy, please wait...)
 04 wait-time 60 secs hearing music
 05 goto step 2 if unconditionally
 06 _____________
 07 _____________
 08 _____________
 09 _____________
 10 _____________
 11 _____________
```

Routing based on time of day

Write a vector for calls that come in after your office closes.

Assume that your business is open 7 days a week, from 8:00 a.m. to 5:00 p.m. When calls come in after business hours, you want to play your announcement 4002, which states that the office is closed and asks callers to call back during normal hours. Write the vector so the call disconnects after the announcement is played.

For after hours treatment, write this vector (steps 1, 6, and 7):

Figure 95: Call Vector screen

```
                              CALL VECTOR
  Number:  1       Name: main number calls
             Attendant Vectoring? n     Meet-me Conf? y           Lock? n
    Basic? y  EAS? n   G3V4 Enhanced? y    ANI/II-Digits? y   ASAI Routing? n
Prompting? y  LAI? n  G3V4 Adv Route? y    CINFO? y   BSR? n      Holidays? n

 01 goto step 7 if time-of-day is all 17:00 to all 8:00
 02 queue-to main split 47 pri 1
 03 announcement 4001 (All agents are busy, please wait...)
 04 wait-time 60 secs hearing music
 05 goto step 2 if unconditionally
 06 stop
 07 disconnect after announcement 4002 ("We're sorry, our office is closed...")
 08 _____________
 09 _____________
 10 _____________
 11 _____________
```

If the **goto** command in step 5 fails, Communication Manager goes to the next step. The **stop** in step 6 prevents callers from incorrectly hearing the "office is closed" announcement in step 7. **Stop** keeps the call in the state it was in before the command failed. In this case, if step 5 fails, the call remains in step 4 and the caller continues to hear music.

CAUTION:

Add a stop vector step only after calls are routed to a queue. If a stop vector is executed for a call not in queue, the call drops.

Allowing callers to leave a message

Write a vector that allows callers to leave messages. This type of vector uses a hunt group called a messaging split. For our example, we send after-hours calls to the voice mailbox at extension 2000 and use messaging split 99.

Once the vector routes a call to the mailbox, the caller hears a greeting (that was recorded with the voice mail for mailbox 2000) that tells them they can leave a message.

To let callers leave messages, write this vector (step 7):

Figure 96: Call Vector screen

```
                              CALL VECTOR
  Number:  1       Name: main number calls
             Attendant Vectoring? n     Meet-me Conf? y           Lock? n
    Basic? y  EAS? n   G3V4 Enhanced? y    ANI/II-Digits? y   ASAI Routing? n
Prompting? y  LAI? n  G3V4 Adv Route? y    CINFO? y   BSR? n      Holidays? n

 01 goto step 7 if time-of-day is all 17:00 to all 8:00
 02 queue-to main split 47 pri 1
 03 announcement 4001 (All agents are busy, please wait...)
 04 wait-time 60 secs hearing music
 05 goto step 2 if unconditionally
 06 stop
 07 messaging split 99 for extension 2000
 08 _____________
 09 _____________
 10 _____________
 11 _____________
```

Redirecting calls during an emergency or holiday

You can provide a quick way for a supervisor or agent to redirect calls during an emergency or holiday. Use a special mailbox where you can easily change announcements. This vector is also an alternative to making sure all agents log out before leaving their phones.

In our example, no agents are normally logged in to split 10. We'll use split 10 for an emergency. We preset buttons on our agents' phones so people with these phones can log in at the touch of a button.

To quickly redirect calls:

1. Create a special mailbox with the appropriate announcement such as "We are unable to answer your call at this time" or "Today is a holiday, please call back tomorrow."

   In our example, we recorded the mailbox greeting for extension 2001.

2. Insert the following bold steps (steps 1, 10, and 11).

   See Inserting a step on page 244 for more information.

Figure 97: Call Vector screen

```
                              CALL VECTOR
  Number:  1       Name: main number calls
             Attendant Vectoring? n     Meet-me Conf? y           Lock? n
    Basic? y  EAS? n   G3V4 Enhanced? y    ANI/II-Digits? y   ASAI Routing? n
Prompting? y  LAI? n  G3V4 Adv Route? y    CINFO? y   BSR? n      Holidays? n

 01 goto step 10 if staff agents split 10 > 0
 02 goto step 8 if time-of-day is all 17:00 to all 8:00
 03 queue-to main split 47 pri 1
 04 announcement 4001 (All agents are busy, please wait...)
 05 wait-time 60 secs hearing music
 06 goto step 2 if unconditionally
 07 stop
 08 messaging split 99 for extension 2000
 09 stop
 10 messaging split 99 for extension 2001
 11 stop
```

When there is an emergency, fire drill, or holiday, the supervisor or agent logs into this split.

When an agent logs into split 10, the system looks at vector step 1, sees that more than 0 people are logged into split 10, and sends calls to step 10 (which sends to messaging split 99).

When your business returns to normal and the agent logs out of split 10, call handling returns to normal.

Giving callers additional choices

You can give your callers a list of options when they call. Your vector tells Communication Manager to play an announcement that contains the choices. Communication Manager collects the digits the caller dials in response to the announcement and routes the call accordingly.

We'll create a vector that plays an announcement, then lets callers dial an extension or wait in the queue for an attendant.

Please note, the following example of this "auto attendant" vector is a new vector and is not built on the vector we used in the previous example.

To let callers connect to an extension, write this kind of vector:

Figure 98: Call Vector screen

```
                              CALL VECTOR
  Number:  1       Name: main number calls
             Attendant Vectoring? n     Meet-me Conf? y           Lock? n
    Basic? y  EAS? n   G3V4 Enhanced? y    ANI/II-Digits? y   ASAI Routing? n
Prompting? y  LAI? n  G3V4 Adv Route? y    CINFO? y   BSR? n      Holidays? n

 01 wait-time 0 seconds hearing music
 02 collect 4 digits after announcement 4004 (You have reached our company.
       Please dial a 4-digit extension or wait for the attendant.)
 03 route-to digits with coverage y
 04 route-to number 0 with cov n if unconditionally
 05 stop
 06 _____________
 07 _____________
 08 _____________
 09 _____________
 10 _____________
 11 _____________
```

Inserting a step

It is easy to change a vector step and not have to retype the entire vector. We will add announcement 4005 between step 3 and step 4 in vector 20.

To insert a new vector step in vector 20:

1. Type **change vector 20** and press **Enter**.

   The **Call Vector** screen appears.

2. Press **Edit**.
3. Type **i** followed by a space and the number of the step you want to add.

   In our example, type i 4.

4. Type the new vector step.

   We will type **announcement 4005 (Please wait...)**.

5. Press **Enter** to save your changes.

Tip:

When you insert a new vector step, the system automatically renumbers the rest of the vector steps and all references to the vector steps. Communication Manager inserts a "*" when the numbering needs more attention.

Deleting a step

To delete vector step 5 from vector 20:

1. Type **change vector 20** and press **Enter**.

   The **Call Vector** screen appears.

2. Press **Edit**.

3. Type **d** followed by a space and the number of the step you want to delete.

   In our example, type **d 5**.

   Tip:

   You can delete a range of vector steps. For example, to delete steps 2 through 5, type **d 2-5** and press **Enter**.

4. Press **Enter** to save your changes.

   Tip:

   When you delete a vector step, the system automatically renumbers the rest of the vector steps and all references to the vector steps. An "*" is inserted when the numbering needs more attention.

Handling TTY calls with vectors

Unlike fax machines and computer modems, a Tele-typwriter device (TTY) has no handshake tone and no carrier tone. A TTY is silent when not transmitting. This is why systems cannot identify TTY callers automatically. However, the absence of these special tones also means that voice and TTY tones can be intermixed in pre-recorded announcements. The ability to provide a hybrid voice-and-TTY announcement, when combined with the auto-attendant vectoring capability, can permit a single telephone number to accommodate both voice and TTY callers.

The sample vector that follows allows TTY callers to access a TTY agent. It begins with a step that plays a TTY announcement combined with a voice announcement. The announcement tells the TTY caller to enter a digit that will direct them to a TTY support person. The vector then processes the digit entered to connect the TTY caller to the TTY split (or hunt group). For more information on recording TTY announcements, see Managing Announcements on page 457.

In the following example, split 47 (hunt group 47) has already been established and consists of TTY-enabled agents.

If a TTY caller calls the number that connects to vector 33, the following occurs:

1. After a short burst of ringing, a quick burst of TTY tones is sent to the caller telling the caller to hold, "HD". Then, a voice announcement follows for callers using a normal telephone connection. The announcement tells them to stay on the line. Finally, another burst of TTY tones is sent to the TTY caller which displays on the caller's TTY device as, "Dial 1."

   The TTY caller won't hear the voice announcement, but because the step collects digits, it allows the caller to enter **1** on his or her touchtone telephone.

   Note:

   For voice callers, the burst of TTY tones lasts about one second and sounds like a bird chirping.

2. In vector step 3, since the TTY caller entered **1** in vector step 2, the TTY caller is sent to vector step 8, at which point the caller is put in queue for a TTY-enabled agent in split 47.

   Note:

   The voice caller is sent to vector step 3 also, but a voice caller does not go to vector step 8 because the caller did not enter **1** at vector step 2. Instead, voice callers continue on to vector step 4, where they connect to split 48.

3. While the TTY caller waits in queue, he or she hears silence from vector step 9, then the announcement in vector step 10, and is then looped back to wait with silence by vector step 11.

More information

See the *Avaya Communication Manager Contact Center Call Vectoring and Expert Agent Selection (EAS) Guide* for more information.

Automated Attendant competes with several features for ports on the Call Classifier — Detector circuit pack or equivalent. See the *Hardware Guide for Avaya Communication Manager* for more information on the circuit pack.

Fixing vector problems

If there is a problem with a vector, Communication Manager records the error as a vector event. Vector events occur for a number of reasons including problems with a trunk, full queue slots, or the vector reaching the maximum 1000 steps allowed.

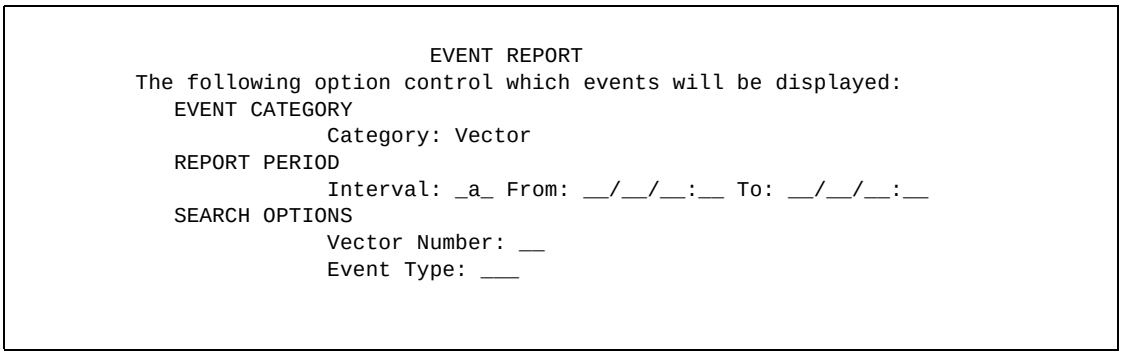
Use **display events** to access the **Event Report** screen and see the event record. Use the event record to see why the vector failed.

To view the **Event Report**:

1. Type **`display events`** and press **Enter**.

   The **Event Report** screen appears.

Figure 99: Event Report screen



```
                          EVENT REPORT
The following option control which events will be displayed:
   EVENT CATEGORY
               Category: Vector
   REPORT PERIOD
               Interval: _a_ From: __/__/__:__ To: __/__/__:__
   SEARCH OPTIONS
               Vector Number: __
               Event Type: ___
```

2. To see all current vector events, press **Enter**.

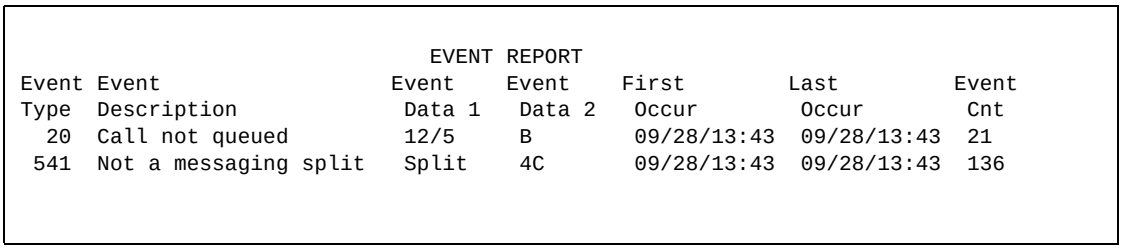
   OR

   Indicate the events that you want to see by completing the **Report Period** and **Search Option** fields. See the *Avaya Communication Manager Contact Center Call Vectoring and Expert Agent Selection (EAS) Guide* for more information.

3. Press **Enter** to view the report.

   The **Event Report (detail)** screen appears.

Figure 100: Event Report screen



```
                              EVENT REPORT
Event Event                  Event    Event    First         Last          Event
Type  Description             Data 1   Data 2   Occur         Occur         Cnt
  20  Call not queued         12/5     B        09/28/13:43   09/28/13:43   21
 541  Not a messaging split   Split    4C       09/28/13:43   09/28/13:43   136
```

Look at the information in the **Event Data** field to diagnose the vector event. In this example, there was a problem with:

- Vector 12, step 5
- Split 89

Vector directory numbers

A VDN is an extension that directs an incoming call to a specific vector. This number is a "soft" extension number not assigned to an equipment location. VDNs must follow your dial plan.

We will create VDN 5011 for our sales department. A call into 5011 routes to vector 11. This vector plays an announcement and queues calls to the sales department.

SECURITY ALERT:

Vector fraud is one of the most common types of toll fraud because vectors route calls based on the class of restriction (COR) assigned to the VDN. See the *Avaya Toll Fraud and Security Handbook* for more information.

Adding a vector directory number

To add a vector directory number:

1. Type **add VDN 5011** and press **Enter**.

   You enter the VDN extension you want to add. The **Vector Directory Number** screen appears.

Figure 101: Vector Directory Number screen

```
                        VECTOR DIRECTORY NUMBER

                          Extension: 5011
                               Name: Sales Department
                 Allow VDN Override? n
                                COR: 1
                                 TN: 1
                       Vector Number: 11
                          AUDIX Name:
                Messaging Server Name:
                           Measured: both
    Acceptable Service Level (sec):
     VDN of Origin Annc. Extension: 301
                          1st Skill:
                          2nd Skill:
                          3rd Skill:

                   Return Destination:
              VDN Timed ACW Interval:
                     BSR Application:
      BSR Available Agent Strategy: 1st-found
```

2. Type a description for this VDN in the Name field.

   In our example, type **Sales Department**.

   The information in the VDN Name field appears on a display telephone. This allows the agent to recognize the nature of the call and respond accordingly.

   Tip:

   The **VDN Override** on the Vector Directory Number screen controls the operation of the display.

3. Enter the vector number.

   In our example, type **11**.

4. In the **Measured** field, indicate how you want to measure calls to his VDN.

   In our example, type **both** (for both CMS and BCMS).

   Tip:

   BCMS must be enabled to use **both**. Use **`display system-parameters customer-options`** to see if BCMS is enabled.

5. Press **Enter** to save your changes.

Viewing vector directory numbers

To see the VDNs already associated with your vectors:

1. Type **`list VDN`** and press **Enter**.

   The Vector Directory Number screen appears.

Figure 102: Vector Directory Number screen

```
 VECTOR DIRECTORY NUMBER
                                                              Event
                                  VDN          Vec        Orig   Notif  Skills
Name                        Ext   Ovrd  COR TN Num  Meas  Annc   Adj 1st 2nd 3rd
Tech Support                5000  y     59  1  234  none  301
Customer Serv.              5001  n     1   1    1  none  302
New Orders                  5002  y     23  1    5  none  303
```

2. Each VDN maps to one vector. Several VDNs can map to the same vector.

Understanding Automatic Call Distribution

Automatic Call Distribution (ACD) is an Avaya Communication Manager feature used in many contact centers. ACD gives you greater flexibility to control call flow and to measure the performance of agents.

ACD systems operate differently from non-ACD systems, and they can be much more complex. ACD systems can also be more powerful because they allow you to use features and products that are not available in non-ACD systems. See the *Avaya Communication Manager Contact Center Guide to ACD Contact Centers* for more information on ACD call centers.

Enhancing an ACD system

First, all call center management systems (such as Avaya's Basic Call Management System (BCMS), BCMSVu, and the sophisticated Avaya IP Agent Call Management System) require ACD. These management systems give you the ability to measure more aspects of your center's operation, and in more detail, than is possible with standard Avaya Communication Manager reports.

Call vectoring greatly enhances the flexibility of a call center, and most vectoring functions require ACD. Vectoring is a simple programming language that allows you to custom design every aspect of call processing. See What are vectors? on page 236 for more information.

Together, ACD and vectoring allow you to use Expert Agent Selection (EAS) For a variety of reasons, you may want certain agents to handle specific types of calls. For example, you may want only your most experienced agents to handle your most important customers. You may have multilingual agents who can serve callers in a variety of languages.

EAS allows you to classify agents according to their specific skills and then to rank them by ability or experience within each skill. Avaya Communication Manager uses these classifications to match each call with the best available agent. See the *Avaya Communication Manager Contact Center Call Vectoring and Expert Agent Selection (EAS) Guide* for more information on call vectoring and EAS.

Assigning a Terminating Extension Group

A Terminating Extension Group (TEG) allows an incoming call to ring as many as 4 phones at one time. Any user in the group can answer the call.

Once a member of the TEG has answered a group call, the TEG is considered busy. If a second call is directed to the group, it follows a coverage path if one has been assigned.

Instructions

Now assign a terminating extension group to the advertising department. For example, we will assign this TEG to extension 6725.

1. Type **`add term-ext-group next`** and press **Enter**.

   The **Terminating Extension Group** screen appears.

Figure 103: Terminating Extension Group screen

```
                    TERMINATING EXTENSION GROUP

    Group Number: 1                          Group Extension: 6725
      Group Name: advertising                  Coverage Path: 5
   Security Code:                                       COR: 1
                                                         TN: 1
ISDN Caller Disp: mbr-name                     LWC Reception: none
      AUDIX Name:                        Messaging Server Name:

GROUP MEMBER ASSIGNMENTS

    Ext     Name (first 26 characters)      Ext     Name (first 26 characters)
 1: 5101    26 character name sta 5101   3:
 2:                                      4: 5102    26 character name sta 5102
```

2. In the **Group Extension** field, type **6725**.

   This is the extension for the advertising group.

3. In the **Group Name** field, type **advertising**.

   This is the name of the group.

4. In the **Coverage Path** field, type **5**.

   This is the number of the call coverage path for this group.

Chapter 8: Routing Outgoing Calls

World Class Routing

Your system uses Automatic Alternate Routing (AAR) and Automatic Route Selection (ARS) to direct outgoing calls.

- AAR routes calls within your company over your own private network.
- ARS routes calls that go outside your company over public networks. ARS also routes calls to remote company locations if you do not have a private network.

Automatic routing begins when a user dials a feature access code (FAC) followed by the number the user wants to call. Avaya Communication Manager analyzes the digits dialed, selects the route for the call, deletes and inserts digits if necessary, and routes the call over the trunks you specify in your routing tables. ARS and AAR can access the same trunk groups and share the same route patterns and other routing information. ARS calls can be converted to AAR calls and vice-versa.

The FAC for AAR is usually the digit 8. The FAC for ARS is usually the digit 9 in the US and 0 outside of the US. Your Avaya technician or business partner sets up AAR on your server running Communication Manager and usually assigns the AAR FAC at the same time. You can administer your own ARS FAC.

This section describes only ARS call routing.

Managing Calling Privileges

Each time you set up a telephone, you use the **Station** screen to assign a class of restriction (COR). You can create different CORs for different groups of users. For example, you may want executives in your company to have different calling privileges than receptionists.

When you set up a COR, you specify a facility restriction level (FRL) on the Class of Restriction screen. The FRL determines the calling privileges of the user. Facility Restriction Levels are ranked from 0 to 7, where 7 has the highest level of privileges.

You also assign an FRL to each route pattern preference in the **Route Pattern** screen. When a user makes a call, the system checks the user's COR. The call is allowed if the caller's FRL is higher than or equal to the route pattern preference's FRL.

Instructions

Let us say we are setting up a new telephone for an executive. The current translations assign COR 1, with outward restrictions and an FRL 0, which is the lowest permission level available. We want to assign a COR with the highest level of permissions, FRL 7, to station 1234.

To change station 1234 from COR 1 to COR 7:

1. Type **`change station 1234`** and press **Enter**.

   The **Station** screen appears.

2. In the **COR** field, type **7** and press **Enter** to save your changes.

3. To change from FRL 0 to FRL 7, type **`change cor 7`** and press **Enter**.

   The **Class of Restriction** screen appears.

4. In the **FRL** field, type **7** and press **Enter** to save your changes.

Now all users with COR 7 will have the highest level of calling permissions.

Assigning ARS FAC

Be sure the ARS feature access code (FAC) is set up on your system. In the U.S., 9 is usually the ARS FAC. Users dial 9 to make an outgoing call.

When a user dials 9 to access ARS and make an outgoing call, the ARS access code 9 is dropped before digit analysis takes place. will not be part of the digit analysis.

Instructions

To assign the ARS FAC:

1. Type **`change dialplan`** and press **Enter**.

   The **DCS to QSIG TSC Gateway screen** appears.

2. Move to the 9 row and type **fac** in the first column. Press **Enter** to save your changes.
3. Type **`change features`** and press **Enter**.

   The **Feature Access Code** screen appears.

4. Type **9** in the **ARS - access code** field.
5. Press **Enter** to save your changes.

Location ARS FAC

The **Location ARS FAC** allows users in different locations to use the same "culturally significant" FAC they are accustomed to, such as dialing 9 for an outside line, and access the same feature. The Location ARS FAC is only accessible for calling numbers at locations administered with that ARS FAC (for details on setting up Location ARS FAC, see the **Locations** screen). If an attempt is made to use an ARS FAC at a location for which it is not valid, the attempt is denied. The ARS access code on the **Feature Access Code** screen continues to be used when a location ARS does not exist. If a location ARS FAC exists, then the ARS access code on the **Feature Access Code** screen is prohibited/denied from that location.

By using a local ARS code, the ability to administer two ARS codes on the **Feature Access Code** screen is lost.

Displaying ARS Analysis Information

Instructions

You will want to become familiar with how your system currently routes outgoing calls. To display the ARS Digit Analysis Table that controls how the system routes calls that begin with 1:

1. Type **`display ars analysis 1`** and press **Enter**.

   The ARS Digit Analysis Table for dialed strings that begin with 1 appears. Note that Communication Manager displays only as many dialed strings as can fit on one screen at a time.

To see all the dialed strings that are defined for your system, run an ARS Digit Analysis report:

1. Type **`list ars analysis`** and press **Enter**.

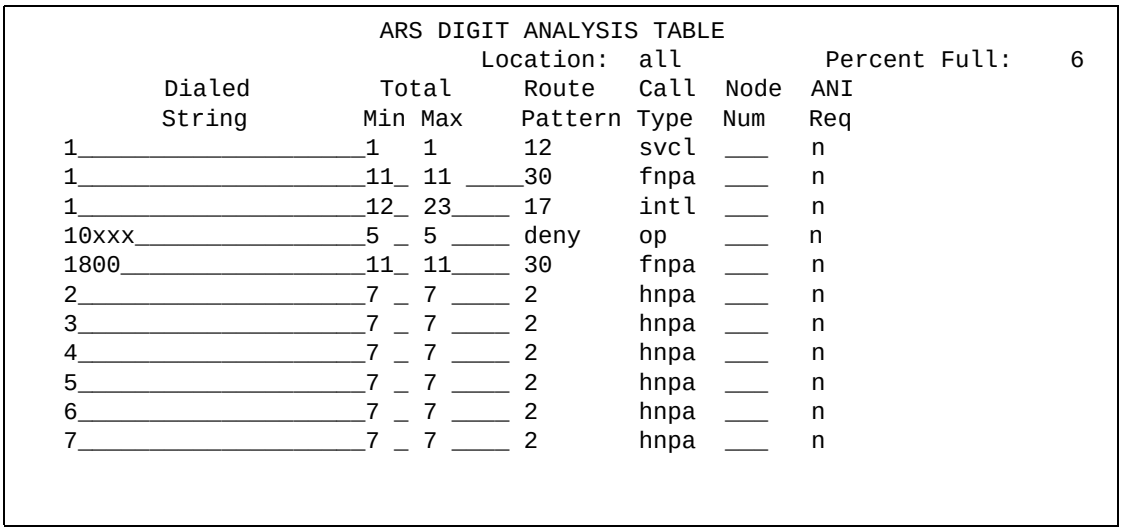
The **ARS Digit Analysis Report** appears. You may want to print this report to keep in your paper records.

Understanding ARS Analysis

With ARS, Communication Manager checks the digits in the number called against the ARS Digit Analysis Table to determine how to handle the dialed digits. Communication Manager also uses Class of Restriction (COR) and Facility Restriction Level (FRL) to determine the calling privileges.

Let us look at a very simple AAR and ARS digit analysis table. Your system likely has more defined dialed strings than this example.

Figure 104: ARS Digit Analysis Table



ARS DIGIT ANALYSIS TABLE
Location: all Percent Full: 6

Dialed String	Total Min	Total Max	Route Pattern	Call Type	Node Num	ANI Req
1	1	1	12	svcl		n
1	11	11	30	fnpa		n
1	12	23	17	intl		n
10xxx	5	5	deny	op		n
1800	11	11	30	fnpa		n
2	7	7	2	hnpa		n
3	7	7	2	hnpa		n
4	7	7	2	hnpa		n
5	7	7	2	hnpa		n
6	7	7	2	hnpa		n
7	7	7	2	hnpa		n

The far-left column of the **ARS Digit Analysis Table** lists the first digits in the dialed string. When a user makes an outgoing call, the system analyzes the digits, looks for a match in the table, and uses the information in the matching row to determine how to route the call.

Let us say a caller places a call to 1-303-233-1000. Communication Manager matches the dialed digits with those in the first column of the table. In this example, the dialed string matches the ‘1’. Then Communication Manager matches the length of the entire dialed string (11 digits) to the minimum and maximum length columns. In our example, the 11-digit call that started with 1 follows route pattern 30 as an fnpa call.

Tip:

The first dialed digit for an external call is often an access code. If ‘9’ is defined as the ARS access code, Communication Manager drops this digit and analyzes the remaining digits with the ARS Analysis Table.

The Route Pattern points to the route that handles the calls that match this dial string.

Call Type tells what kind of call is made with this dial string. Call type helps Communication Manager decide how to handle the dialed string.

Examples of digit conversion

Your system uses the AAR or ARS Digit Conversion Table to change a dialed number for more efficient routing. Digits may be inserted or deleted from the dialed number. For instance, you can tell Communication Manager to delete a 1 and an area code on calls to one of your locations, and avoid long-distance charges by routing the call over your private network.

The table below reflects these values:

- ARS feature access code = 9
  AAR feature access code = 8
  Private Network Office Code (also known as Home RNX) = 222
  Prefix 1 is required on all long-distance DDD calls
  Dashes (-) are for readability only

Communication Manager maps the dialed digits to the matching pattern that most closely matches the dialed number. Example: If the dialed string is 957-1234 and matching patterns 957-1 and 957-123 are in the table, the match is on pattern 957-123.

Table 4: ARS digit conversion examples *1 of 2*

Operation	Actual Digits Dialed	Matching Pattern	Replacement String	Modified Address	Notes
DDD call to ETN	9-1-303-538-1345	1-303-538	362	362-1345	Call routes via AAR for RNX 362
Long-distance call to specified carrier	9-10222+DDD	10222	(blank)	(blank)	Call routes as dialed with DDD # over private network
Terminating a local DDD call to an internal station	9-1-201-957-5567 or 9-957-5567	1-201-957-5 or 957-5	222-5	222-5567.	Call goes to home RNX 222, ext. 5567
Unauthorized call to intercept treatment	9-1-212-976-1616	1-XXX-976	#	(blank)	"#" means end of dialing. ARS ignores digits dialed after 976. User gets intercept treatment.

1 of 2

Table 4: ARS digit conversion examples *2 of 2*

Operation	Actual Digits Dialed	Matching Pattern	Replacement String	Modified Address	Notes
International calls to an attendant	9-011-91-6725 30	011-91	222-0111#	222-0111	Call routes to local server (RNX 222), then to attendant (222-0111).
International call to announcement (This method may also be used to block unauthorized IDDD calls)	9-011-91-6725 30	011-91	222-1234#	222.1234-	Call routes to local server (RNX 222), then to announcement extension (222-1234).
International call from certain European countries needing dial tone detection	0-00-XXXXXX XX	00	+00+	00+XXXX	The first 0 denotes ARS, the second pair of 0s denotes an international call, the pluses denote "wait" for dial tone detection.

2 of 2

Defining operator assisted calls

Here is an example of how Communication Manager routes an ARS call that begins with 0 and requires operator assistance. The user dials 9 to access ARS, then a 0, then the rest of the number.

To see how Communication Manager handles a call to an operator:

1. Type **`display ars analysis 0`** and press **Enter**.

   The **AAR and ARS Digit Analysis Table** screen starting with 0 appears.

Figure 105: ARS Digit Analysis Table screen

```
                      ARS DIGIT ANALYSIS TABLE
                             Location:  all          Percent Full:    6

       Dialed          Total     Route   Call  Node  ANI
       String        Min Max    Pattern Type  Num   Req
0_____________________1   1      1       svcl  ___   n
0_____________________8__ 8 ____ 1       op    ___   n
0_____________________11_ 11____ 1       op    ___   n
00____________________2 _ 2 ____ 1       op    ___   n
01____________________10_ 23 ___ deny    op    ___   n
011___________________10_ 23 ___ deny    iop   ___   n
1_____________________11_ 11 ___ 3       intl  ___   n
```

The table in this example shows 6 translations for calls that begin with 0.

Instructions

We will use the ARS digit analysis table shown above and follow the routing for an operator assisted a call to NJ.

- A user dials 9 0 908 956 1234.
- Communication Manager drops the ARS FAC (9 in our example), looks at the **ARS Digit Analysis Table** for 0, and analyzes the number. Then it:
  - determines that more than 1 digit was dialed
  - rules out the plan for 00, 01, and 011
  - determines that 11 digits were dialed
- Communication Manager routes the call to route pattern 1 as an operator assisted call.

Defining Inter-exchange carrier calls

Here is an example of how Communication Manager routes an ARS call to an inter-exchange (long-distance) carrier (IXC). IXC numbers directly access your long-distance carrier lines.

IXC numbers begin with 1010, followed by three digits, plus the number as it is normally dialed including 0, 00, or 1+ 10 digits. These numbers are set up on your default translations.

Remember, the user dials 9 to access ARS, then the rest of the number.

Instructions

To see how Communication Manager handles a call to an IXC:

1. Type **`display ars analysis 1`** and press **Enter**.

   The **AAR and ARS Digit Analysis Table** screen starting with 1 appears.

Figure 106: ARS Digital Analysis Table screen

```
                      ARS DIGIT ANALYSIS TABLE
                             Location:  all          Percent Full:    6
       Dialed          Total     Route   Call  Node  ANI
       String        Min Max    Pattern Type  Num   Req
1______________________11_ 11 ___ 3       intl  ___   n
1010xxx________________7 _ 7 ____ 5       op    ___   n
1010xxx0_______________8 _ 8 ____ 5       op    ___   n
1010xxx0_______________18_ 18 ___ 5       op    ___   n
1010xxx00______________9 _ 9 ____ 5       op    ___   n
1010xxx01______________17_ 25 ___ 3       iop   ___   n
2______________________7 _ 7 ____ 2       hnpa  ___   n
3______________________7 _ 7 ____ 2       hnpa  ___   n
4______________________7 _ 7 ____ 2       hnpa  ___   n
```

This table shows 5 translations for IXC calls.

When you use x in the **Dialed String** field, Communication Manager recognizes x as a wildcard. The x represents any digit, 0 - 9. If I dial 1010, the next 3 digits will always match the x wild cards in the dialed string.

Use the ARS digit analysis table shown above and follow the routing for an IXC call to AT&T. 1010288 is the carrier access code for AT&T.

- A user dials 9 1010288 plus a public network number.
- Communication Manager drops the ARS FAC (9 in our example), looks at the **ARS Digit Analysis Table** for 1010, and analyzes the number.
- Then it matches 288 with xxx and sends the call over route pattern 5.

Restricted area codes and prefixes

Certain area code numbers are set aside in the North American Numbering Plan. These numbers are 200, 300, 400, 500, 600, 700, 800, 877, 888, 900. You need to specifically deny calls made to area codes 200 through 900 (except 800 and 888).

You can also deny access to the 976 prefix, which is set aside in each area code for pay-per call services, if you do not want to incur charges. You can block 976 or any other prefix in all NPAs with a single entry in the digit analysis table. See Using wild cards on page 263 for more information.

Instructions

Set the 200 area code apart from other area codes 201 through 209. We use the digit analysis table 120 because it defines long distance calls that begin with 1 and all area codes from 200 through 209.

To deny long distance calls to the 200 area code:

1. Type **`change ars analysis 120`** and press **Enter**.

   The **AAR and ARS Digit Analysis Table** screen beginning with 120 appears.

Figure 107: ARS Digit Analysis Table screen

```
                        ARS DIGIT ANALYSIS TABLE
                                Location:  all          Percent Full:    6
         Dialed          Total     Route   Call  Node  ANI
         String         Min Max    Pattern Type  Num   Req
  120_____________________11  11     4       fnpa  ___   n
  1200____________________11_ 11____ deny    fnpa  ___   n
```

The table in this example shows 2 translations for calls that begin with 120.

First, follow the routing for a long-distance call that begins with 120 and is allowed. The 120 translation handles all dial strings 1-201 through 1-209, and there are many matches.

- A user dials 9 120 plus 8 digits (the first of the 8 digits is not 0).
- Communication Manager drops the ARS FAC (9 in our example), looks at the **ARS Digit Analysis Table** for 120, and analyzes the number. It determines the call is long-distance and sends the call over route pattern 4.

Now we will follow a call that begins with the restricted area code 200. Only one string matches this translation.

- A user dials 9 1200 plus 7 digits.
- Communication Manager drops the ARS FAC (9), and looks at the **ARS Digit Analysis Table** for 1200. It determines that the call type is deny, and the call does not go through.

Using wild cards

You can use wild cards to help separate out calls to certain numbers. Remember, when you use the wild card x in the **Dialed String** field, Communication Manager recognizes x as any digit, 0 - 9. For example, you can restrict users from making calls to a 555 information operator where you might incur charges.

Instructions

To prevent callers from placing calls to long-distance 555 information numbers:

1. Type **change ars analysis 1** and press **Enter**.

   The **AAR and ARS Digit Analysis Table** screen beginning with 1 appears.

Figure 108: AAR and ARS Digit Analysis Table screen

```
                        ARS DIGIT ANALYSIS TABLE
                                Location:  all          Percent Full:    6
        Dialed          Total     Route   Call  Node  ANI
        String        Min Max    Pattern Type  Num   Req
1_____________________11_ 11 ___ 1       intl  ___   n
1xxx555_______________11_ 11 ___ 1       intl  ___   n
1010xxx_______________7 _ 7 ____ 1       op    ___   n
```

2. Use the arrow keys to move to a blank **Dialed String** field.
3. Enter **1xxx555** in the **Dialed String** field.
4. Enter **11** in the **Total Min** and **11** in **Total Max** fields.
5. Enter **deny** (denied) in the **Route Pattern** field.
6. Enter **fnhp** in the **Call Type** field.
7. Press **Enter** to save your changes.

Defining local information calls

You can set up Communication Manager to allow calls to local information, or in this example, 411.

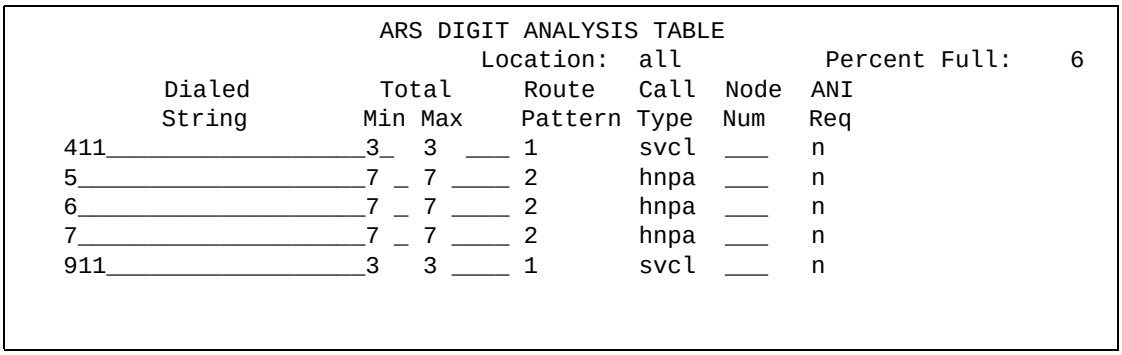
Instructions

To allow 411 service calls:

1. Type **change ars analysis 4** and press **Enter**.

   The **AAR and ARS Digit Analysis Table** screen beginning with 4 appears.

Figure 109: AAR and ARS Digit Analysis Table screen



```
                        ARS DIGIT ANALYSIS TABLE
                                Location:  all          Percent Full:    6
        Dialed          Total     Route   Call  Node  ANI
        String        Min Max    Pattern Type  Num   Req
411_____________________3_  3  ___ 1       svcl  ___   n
5_______________________7 _ 7 ____ 2       hnpa  ___   n
6_______________________7 _ 7 ____ 2       hnpa  ___   n
7_______________________7 _ 7 ____ 2       hnpa  ___   n
911_____________________3   3 ____ 1       svcl  ___   n
```

2. Use the arrow keys to move to a blank **Dialed String** field.
3. Enter **411** in the **Dialed String** field.
4. Enter **3** in the **Total Min** and **3** in **Total Max** fields.
5. Enter **1** in the **Route Pattern** field.
6. Enter **svcl** (service call) in the **Call Type** field.

Press **Enter** to save your changes.

Setting up Multiple Locations

You can define a location number for:

- Cabinets in DEFINITY Server R
- Remote Offices
- Media gateways
- IP network regions, used by IP stations and IP trunks

You can create numbering plans and time zone and daylight savings plans that are specific for each location. Choose your main location, and offset the local time for each location relative to the system clock time. The main location is typically set to have offset 0.

Before you start

Ensure that the **Multiple Locations** field on the **System-Parameters Customer-Options** screen is set to **y**. If this field is set to **n**, contact your Avaya representative for more information. If you are setting up locations across international borders, you must that the **Multinational Locations** field on the **System-Parameters Customer-Options** screen is also set to **y**.

Be sure your daylight savings rules are administered. Daylight Savings Rule numbers are located on the **Daylight Savings Rules** screen.

Each cabinet in a server or switch and each port network in the cabinet must be assigned a location number. See the **`add-cabinet`** and **`change-cabinet`** commands in *Maintenance Commands Reference,* 555-245-101.

Instructions

For example, we will set up multiple locations for Communication Manager server with cabinets in Chicago and New York. Location 1 is assigned to the cabinet in Chicago, our main office, so Central Standard Time is used for our main location. Location 2 is assigned to the cabinet in New York. We'll define the numbering plan area (NPA) for the Chicago and New York locations, and set the time zone offset for NY to show the difference in time between Eastern Standard Time and Central Standard Time.

Tip:

Type **`list cabinets`** to see the cabinet screen and a list of cabinets and their locations.

To define locations for cabinets in Chicago and New York:

1. Type **change locations** and press **Enter**.

   The **Locations** screen appears.

Figure 110: Locations screen

```
change locations                                          Page   1 of   1
                                  LOCATIONS

                ARS Prefix 1 Required For 10-Digit NANP Calls? y

 Loc. Name            Timezone  Daylight-  Number ARS Attd Loc.   Pre- Foreign
 No                    Offset   Savings    Plan   FAC FAC  Parms. fix  Domain
                                Rule       Area                      Route
                                           Code                      Pattern
   1  Main            + 00:00   0
 xxx ________________ _ __:__   __         ___    ____ ____ __    ____ _______
 xxx ________________ _ __:__   __         ___    ____ ____ __    ____ _______
 xxx ________________ _ __:__   __         ___    ____ ____ __    ____ _______
 xxx ________________ _ __:__   __         ___    ____ ____ __    ____ _______
```

2. Type **y** in the **ARS Prefix 1 required for 10-digit NANP calls** field.

   Our dial plan requires users to dial a 1 before all 10-digit (long distance) NANP calls.

3. Type **Chicago** in the **Name** field in the Number **1** row.

   Use this field to identify the location.

4. Type **+00:00** in the **TimeZone Offset** field in the Number **1** row.

   In our example, the system time and the Chicago location time are the same.

5. Type **1** in the **Daylight Savings Rule** field in the Number **1** row.

   In our example, daylight savings rule 1 applies to U.S. daylight savings time.

   Tip:

   Use the **display daylight-savings-rules** command to see what rules have been administered on Communication Manager.

6. Type **312** in the **Number Plan Area Code** field in the Number **1** row.

   In our example, 312 is the local area code for Chicago, location 1.

7. Type **New York** in the **Name** field in the Number **2** row.

8. Type **-01:00** in the **TimeZone Offset** field in the Number **2** row.

   In our example, subtract one hour from the system clock in Chicago to provide the correct time for the location in New York.

9. Type **1** in the **Daylight Savings Rule** field in the Number **2** row.

   In our example, daylight savings rule 1 applies to U.S. daylight savings time, and both locations use the same rule.

10. Type **212** in the **NANP** field in the Number **2** row.

    In our example, 212 is the local area code for New York, location 2.

11. Press **Enter** to save your changes.

Related topics

See *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205 for more information on the Multiple Locations feature.

See Setting the System Date and Time on page 41 for more information about how to set your system clock and specify the daylight savings rule for the location.

See Establishing Daylight Savings Rules on page 31 for more information about how to specify the dates and times of daylight savings rules.

Routing with Multiple Locations

When you set up multiple locations, you can define call routing that covers all locations as well as call routing specific to each individual location. Use your routing tables to define local routing for 911, service operators, local operator access, and all local calls for each location. Leave long-distance and international numbers that apply across all locations on the routing tables with **Location** field set to **all**.

Before you start

Be sure the **Multiple Locations** field on the **System-Parameters Customer-Options** screen is set to **y**. If this field is set to **n**, contact your Avaya representative for more information.

AAR or ARS must be administered.

- For AAR, verify that either the **Private Networking** field or the **Uniform Dialing Plan** field is **y** on the **System-Parameters Customer-Options** screen.
- For ARS, verify that the **ARS** field is **y** on the **System-Parameters Customer-Options** screen.

You can define a location number for:

- Cabinets in DEFINITY Server R
- Remote Offices
- Media gateways
- IP network regions, used by IP stations and IP trunks

Instructions

For example, we will use ARS to set up local call routing for two Communication Manager server locations. Our Chicago server is assigned to location 1, and our New York server is assigned to location 2.

Our example shows a simple local dialing plan. Each location already contains location-specific routing tables. We'll use route pattern 1 for local service calls and route pattern 2 for local HNPA calls in the Chicago location.

Tip:

Create location-specific routing by assigning different route patterns for each location.

To define local calls for servers in Chicago and New York:

1. Type **`change ars analysis location 1`** and press **Enter**.

   The **AAR and ARS Digit Analysis Table** screen for location 1 appears.

Figure 111: ARS Digital Analysis Table screen

```
change ars analysis                                          Page 1 of
X

                          ARS DIGIT ANALYSIS TABLE
                                  Location: 1___      Percent Full:  ___

     Dialed          Total    Route    Call   Node  ANI
     String        Min Max  Pattern   Type   Num   Reqd
2_________________ 7   7     2____    hnpa   ___    n
3_________________ 7   7     2____    hnpa   ___    n
4_________________ 7   7     2____    hnpa   ___    n
411_______________ 3   3     1____    svcl   ___    n
5_________________ 7   7     2____    hnpa   ___    n
555_______________ 7   7     deny_    hnpa   ___    n
6_________________ 7   7     2____    hnpa   ___    n
611_______________ 3   3     1____    svcl   ___    n
7_________________ 7   7     2____    hnpa   ___    n
8_________________ 7   7     2____    hnpa   ___    n
811_______________ 3   3     1____    svcl   ___    n
9_________________ 7   7     2____    hnpa   ___    n
```

2. Type the information for local dialed strings and service calls in each row on the screen.

   In our example, for location 1 (Chicago) local HNPA calls:

   - Type the appropriate digit in the **Dialed String** field.
   - Type **7** in the **Total Min** field.
   - Type **7** in the **Total Max** field.
   - Type **2** in the **Route Pattern** field.
   - Type **hnpa** in the **Call Type** field.

   In our example, for location 1 (Chicago) local service calls:

   - Type the appropriate digits in the **Dialed String** field.
   - Type **3** in the **Total Min** field.
   - Type **3** in the **Total Max** field.
   - Type **1** in the **Route Pattern** field.
   - Type **svcl** in the **Call Type** field.

3. Press **Enter** to save your changes.
4. Type **`change ars analysis 4 location 2`** and press **Enter**.

   The **ARS Digit Analysis Table** for location 2 appears.

5. Type in the local HNPA and service call routing information for New York.
6. Press **Enter** to save your changes.

Related topics

See "Automatic Routing" in *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205 for more information on ARS.

See AAR and ARS Digit Analysis Table on page 709, AAR and ARS Digit Conversion Table on page 715, and Toll Analysis on page 1525 for general information on ARS administration. You can define location specific entries in addition to the global entries on these screens.

See "Multiple Locations" in *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205 for more information on the Multiple Locations feature.

Modifying Call Routing

If your system uses ARS Digit Analysis to analyze dialed strings and select the best route for a call, you must change the digit analysis table to modify call routing. For example, you'll need to update this table to add new area codes or to restrict users from calling specific areas or countries.

Adding a new area code or prefix

A common task for system administrators is to configure their system to recognize new area codes or prefixes.

When you want to add a new area code or prefix, you look up the settings for the old area code or prefix and enter the same information for the new one.

Tip:

Use **`display toll`** ***xxx***, where ***xxx*** is the prefix you want to add, to see if the new area code or prefix number is set up as a toll call (y) or not. Some users may not be allowed to dial toll call numbers.

Instructions

We will add a new area code. When the California area code, 415, splits and portions change to 650, you will need to add this new area code to your system.

Tip:

If you do not need to use 1 for area code calls, omit the **1** in steps 1, 3, and 5 in our example. Also, enter **10** in the **Total Min** and **Total Max** fields (instead of 11) in step 6.

To add this non-local area code:

1. Type **`list ars route-chosen 14152223333`** and press **Enter**.

   You can use any 7-digit number after 1 and the old area code (415). We used 222-3333.

   The **ARS Route Chosen Report** screen appears.

Figure 112: ARS Route Chosen Report screen

```
                        ARS ROUTE CHOSEN REPORT

     Location: 1                        Partitioned Group Number:  1

      Dialed               Total      Route    Call      Node      Location
      String             Min  Max    Pattern  Type      Number
      141                11   11     30       fnpa                 all
```

2. Write down the **Total Min**, **Total Max**, **Route Pattern**, and **Call Type** values from this screen.

   In this example, the **Total Min** is **11**, **Total Max** is **11**, **Route Pattern** is **30**, and the **Call Type** is **fnpa**.

3. Type **`change ars analysis 1650`** and press **Enter**.

   The **AAR and ARS Digit Analysis Table** screen appears.

Figure 113: ARS Digit Analysis Table screen

```
                        ARS DIGIT ANALYSIS TABLE
                                Location:  all          Percent Full:    6
        Dialed          Total     Route   Call  Node  ANI
        String        Min Max     Pattern Type  Num   Req
1_____________________ 11  11      30      fnpa  ___   n
167___________________ 11  11      30      fnpa  ___   n
1800__________________ 11  11      30      fnpa  ___   n
2_____________________ 7   7       2       hnpa  ___   n
3_____________________ 7   7       2       hnpa  ___   n
4_____________________ 7   7       2       hnpa  ___   n
5_____________________ 7   7       2       hnpa  ___   n
7_____________________ 7   7       2       hnpa  ___   n
8_____________________ 7   7       2       hnpa  ___   n
911___________________ 3   3       1       emer  ___   n
976_________________  _11  11      deny    hnpa  ___   n
```

4. Move to a blank **Dialed String** field.

   If the dialed string is already defined in your system, the cursor appears in the appropriate **Dialed String** field, where you can make changes.

5. Enter **1650** in the **Dialed String** field.

6. Enter the minimum and maximum values from step 2 in the **Total Mn** and **Total Mx** fields.

   In our example, enter **11** in each field.

7. Enter the route pattern from step 2 in the **Route Pattern** field.

   In our example, enter **30**.

8. Enter **fnpa** in the **Call Type** field.
9. Enter the node number from step 2 in the **Node Num** field.

   For our example, leave the node number blank.

10. Press **ENTER** to save your changes.

To add a new prefix, follow the same directions, except use a shorter dial string (such as **`list ars route-chosen 2223333`**, where **`222`** is the old prefix) and a dial type of **hnpa**.

Tip:

If you change an existing area code for a network with multiple locations, be sure to change the **Number Plan Area Code** field on the **Locations** screen. See Setting up Multiple Locations on page 265.

Using ARS to restrict outgoing calls

ARS allows you to block outgoing calls to specific dialed strings. For example, you can restrict users from making international calls to countries where you do not do business, or in the U.S. you can restrict access to 900 and 976 pay-per-call numbers.

SECURITY ALERT:

To prevent toll fraud, deny calls to countries where you do not do business. The following countries are currently concerns for fraudulent calling.

country	code	country	code
Colombia	57	Pakistan	92
Ivory Coast	225	Peru	51
Mali	23	Senegal	221
Nigeria	234	Yemen	967

Instructions

To prevent callers from placing calls to Colombia (57):

1. Type **change ars analysis 01157** and press **Enter**.

   Enter 011 (international access) and the country code (57). The **ARS Digit Analysis Table** screen appears.

2. Move to a blank **Dialed String** field.

   If the dialed string is already defined in your system, the cursor appears in the appropriate **Dialed String** field. Skip to Step 5 to deny calls to this dialed string.

3. Enter **01157** in the **Dialed String** field.
4. Enter **10** in the **Total Min** and **23** in **Total Max** fields.
5. Enter **deny** (denied) in the **Route Pattern** field.
6. Enter **intl** in the **Call Type** field.
7. Press **Enter** to save your changes.

Overriding Call Restrictions

You can use authorization codes to enable callers to override a station's calling privileges. For example, you can give a supervisor an authorization code so they can make calls from a telephone that is usually restricted for these calls. Since each authorization code has its own COR, the system uses the COR assigned to the authorization code (and FRL assigned to the COR) to override the privileges associated with the employee's telephone.

Note that authorization codes do not override dialed strings that are denied. For example, if your ARS tables restrict users from placing calls to Colombia, a caller cannot override the restriction with an authorization code.

Before you start

Verify that the **Authorization Codes** field on the **System Parameters Customer Options** screen is set to **y**.

SECURITY ALERT:
You should make authorization codes as long as possible to increase the level of security. You can set the length of authorization codes on the **Feature-Related System Parameters** screen.

Instructions

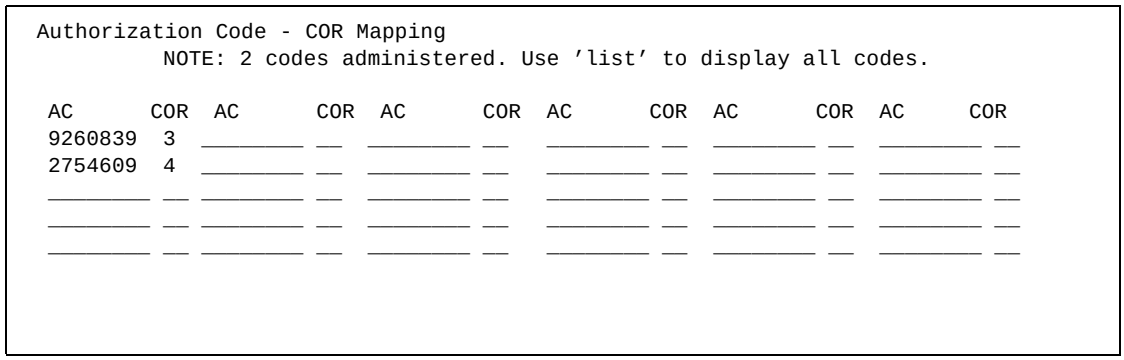
We will create an authorization code 4395721with a COR of 2.

1. Type **`change authorization-code 4395721`** and press **Enter**.

   The **Authorization Code — COR Mapping** screen appears.

2. In the **AC** field, type **4395721**.
3. In the **COR** field, enter **2**.
4. Press **Enter** to save your changes.

Figure 114: Authorization Code - COR Mapping



Defining ARS Partitions

Most companies want all their users to be able to make the same calls and follow the same route patterns. However, you may find it helpful to provide special calling permissions or restrictions to a group of users or to particular phones.

ARS partitioning allows you to provide different call routing for a group of users or for specific phones.

Note:

If you used partitioning on a prior release of Avaya Communication Manager and you want to continue to use partitioning, please read this section carefully. In this release of Avaya Communication Manager, partition groups are defined on the Partition Route Table. If you want to define routing based on partition groups, use the Partition Route Table. Partition groups are no longer defined on the Digit Analysis Table.

Before you start

- Ensure that the **Tenant Partitioning** field on the **System Parameters Customer Options** screen is **y**.
- Ensure that the **Time of Day Routing** field on the **System Parameters Customer Options** screen is **n**.

Setting up partition groups

Let us say you allow your employees to make local, long distance, and emergency calls. However, you have a lobby telephone for visitors and you want to allow users to make only local, toll-free, and emergency calls from this telephone.

To restrict the lobby telephone, you modify the routing for a partition group to enable only specific calls, such as U.S. based toll-free 1-800 calls, and then assign this partition group to the lobby telephone.

Instructions

To enable 1-800 calls for partition group 2:

1. Type **`list ars route-chosen 18002221000`** and press **Enter.**

   You can use any 7-digit number following the 1800 to create an example of the dialed string.

   The **ARS Route Chosen Report** screen for partition group 1 appears.

Figure 115: ARS Route Chosen Report screen

```
 ARS ROUTE CHOSEN REPORT

      Location :  1                    Partitioned Group Number: 1

          Dialed            Total     Route      Call      Node     Location
          String           Min  Max  Pattern     Type      Number

1800_____________________     11 11     p1___       fnpa     _____     all
```

2. Record the route pattern for the selected dialed string.

   In our example, the route pattern for 1800 is p1. This indicates that the system uses the Partition Routing Table to determine which route pattern to use for each partition.

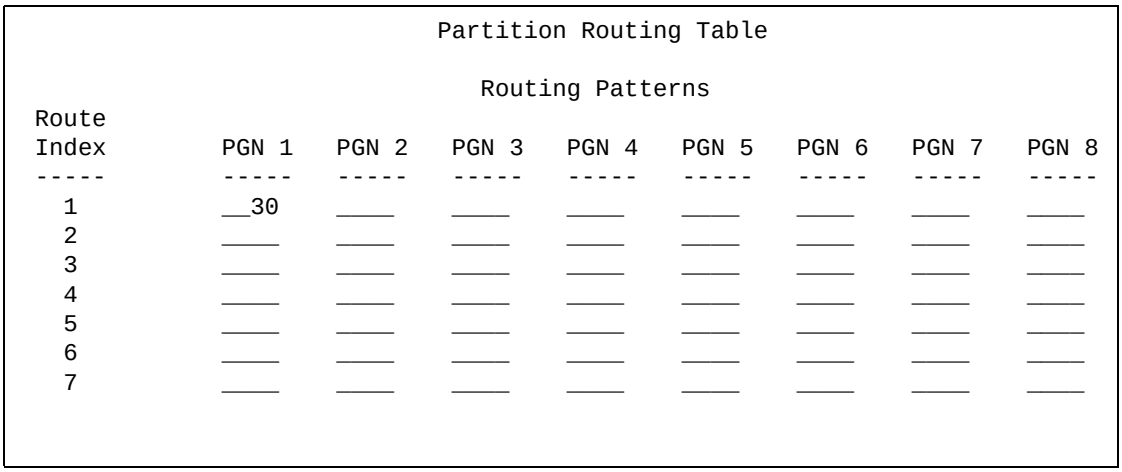
   Note:

   If there was a number (with no p) under **Route Pattern** on the **Route Chosen Report**, then all partitions use the same route pattern. You need to use the Partition Routing Table only if you want to use different route patterns for different partition groups.

3. Press **Cancel** to return to the command prompt.
4. Type **`change partition-route-table index 1`** and press **Enter.**

   The **Partition Routing Table** screen appears. In our example, partition group 1 can make 1800 calls and these calls use route pattern 30.

Figure 116: Partition Routing Table screen



```
                          Partition Routing Table

                             Routing Patterns
Route
Index        PGN 1   PGN 2   PGN 3   PGN 4   PGN 5   PGN 6   PGN 7   PGN 8
-----        -----   -----   -----   -----   -----   -----   -----   -----
  1          __30    _____   _____   _____   _____   _____   _____   _____
  2          _____   _____   _____   _____   _____   _____   _____   _____
  3          _____   _____   _____   _____   _____   _____   _____   _____
  4          _____   _____   _____   _____   _____   _____   _____   _____
  5          _____   _____   _____   _____   _____   _____   _____   _____
  6          _____   _____   _____   _____   _____   _____   _____   _____
  7          _____   _____   _____   _____   _____   _____   _____   _____
```

5. In the **PGN2** column that corresponds to Route Index 1, type **30** and press **Enter**.

   This tells the system to use route pattern 30 for partition group 2 and allow partition group 2 members to make calls to 1800 numbers.

Assigning a telephone to a partition group

To assign an extension to a partition group, first assign the partition group to a COR, and then assign that COR to the extension.

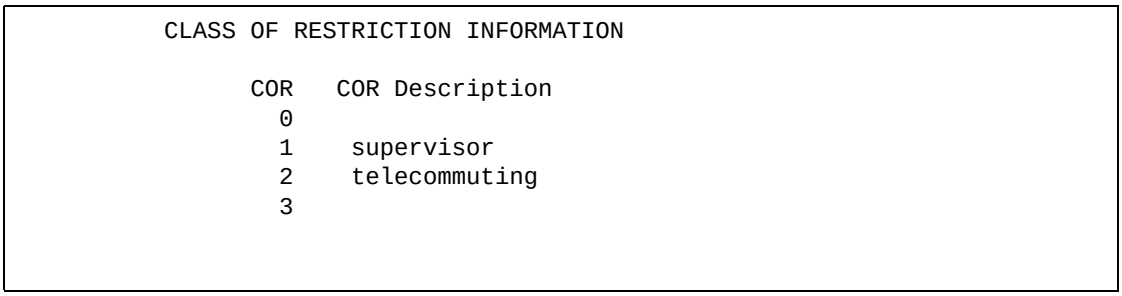
Instructions

To assign a Class of Restriction (COR) to partition group 2:

1. Type **`list cor`** and press **Enter**.

   The **Class of Restriction Information** screen appears.

Figure 117: Class of Restriction Information screen



```
          CLASS OF RESTRICTION INFORMATION

                COR   COR Description
                  0
                  1    supervisor
                  2    telecommuting
                  3
```

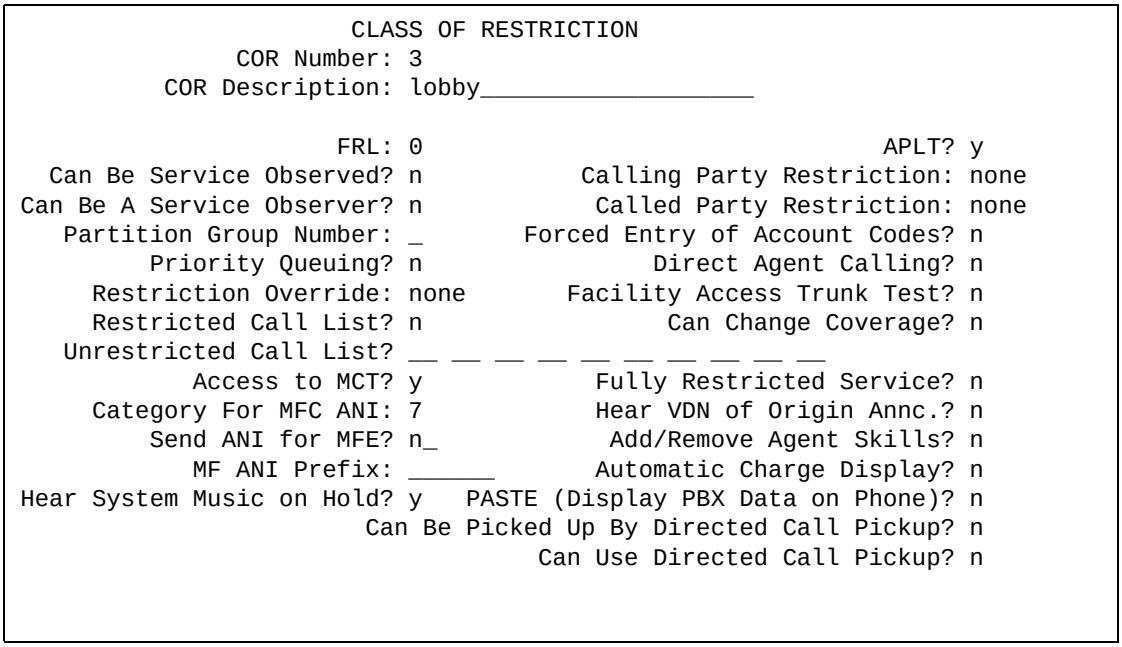
2. Choose a COR that has not been used and press **Cancel**.

   In our example, select **3**.

3. Type **change cor 3** and press **Enter**.

   The **Class of Restriction** screen appears.

Figure 118: Class of Restriction screen



```
                      CLASS OF RESTRICTION
               COR Number: 3
          COR Description: lobby____________________

                      FRL: 0                              APLT? y
  Can Be Service Observed? n          Calling Party Restriction: none
Can Be A Service Observer? n           Called Party Restriction: none
   Partition Group Number: _      Forced Entry of Account Codes? n
         Priority Queuing? n               Direct Agent Calling? n
     Restriction Override: none      Facility Access Trunk Test? n
     Restricted Call List? n                 Can Change Coverage? n
   Unrestricted Call List? __ __ __ __ __ __ __ __ __ __
            Access to MCT? y           Fully Restricted Service? n
     Category For MFC ANI: 7           Hear VDN of Origin Annc.? n
         Send ANI for MFE? n_           Add/Remove Agent Skills? n
            MF ANI Prefix: ______      Automatic Charge Display? n
Hear System Music on Hold? y   PASTE (Display PBX Data on Phone)? n
                        Can Be Picked Up By Directed Call Pickup? n
                                   Can Use Directed Call Pickup? n
```

4. Type a name for this COR in the **COR Description** field.

   In our example, type **lobby**.

5. Enter **2** in the **Partition Group Number** field.

6. Press **Enter** to save your changes.

Now assign COR 3 to the lobby telephone at extension 1234:

1. Type **change station 1234** and press **Enter**.

   The **Station** screen for 1234 appears.

2. In the **COR** field, enter **3**.
3. Press **Enter** to save your changes.

Setting up Time of Day Routing

Time of Day Routing lets you redirect calls to coverage paths according to the time of day and day of the week. You need to define the coverage paths you want to use before you define the time of day coverage plan.

You can route calls based on the least expensive route according to the time of day and day of the week the call is made. You can also deny outgoing long-distance calls after business hours to help prevent toll fraud. Time of Day Routing applies to all AAR or ARS outgoing calls and trunks used for call forwarding to external numbers.

Before you start

AAR or ARS must be administered on Communication Manager before you use Time of Day Routing.

- For AAR, verify that either the **Private Networking** field or the **Uniform Dialing Plan** field is **y** on the **System-Parameters Customer-Options** screen.
- For ARS, verify that the **ARS** field is **y** and the **Time of Day Routing** field is **y** on the **System-Parameters Customer-Options** screen.

Instructions

As an example, we will allow our executives to make long distance calls during business hours. Let us look at the Time of Day Routing Plan before we make any changes.

To display your Time of Day Routing Plan:

1. Type **display time-of-day 1** and press **Enter**.

   The **Time of Day Routing Plan** **1** screen appears.

Figure 119: Time of Day Routing Plan 1 screen

```
                                                          Page 1 of 1

                          TIME OF DAY ROUTING PLAN 1

     Act    PGN Act      PGN   Act      PGN    Act      PGN   Act      PGN Act
     Time    #  Time      #    Time      #     Time      #    Time      #  Time
Sun 00:01   1 __:__      _    __:__     _     __:__     _    __:__     _ __:__  _
Mon 00:01   1 08:__00    2    12:__00   1     13:__00   2    17:__00   1 __:__  _
Tue 00:01   1 08:__00    2    12:__00   1     13:__00   2    17:__00   1 __:__  _
Wed 00:01   1 08:__00    2    12:__00   1     13:__00   2    17:__00   1 __:__  _
Thu 00:01   1 08:__00    2    12:__00   1     13:__00   2    17:__00   1 __:__  _
Fri 00:01   1 08:__00    2    12:__00   1     13:__00   2    17:__00   1 __:__  _
Sat 00:01   1 __:__      _    __:__     _     __:__     _    __:__     _ __:__  _
```

Make a note of the routing plan that is currently in effect. In our example, this plan is for employees who can only make local calls.

You can see that in our example, two partition group numbers control time of day routing. PGN 1 begins one minute after midnight (00:01) every day of the week, and is used for after-business hours and all day Saturday and Sunday. PGN 2 is assigned to office hours Monday through Friday, not including noon (12:00) to 1:00 p.m. (13:00).

2. Press **Cancel** to clear the screen.

Now we will create a new time of day routing plan for long-distance calls for our executives.

1. Type **`change time-of-day 2`** and press **Enter**.

   The **Time of Day Routing Plan 2** screen appears.

2. Type **1** in each field as shown on **Time of Day Routing Plan 1**.

   In our example, this is the PGN used for after hours and the lunch hour.

3. Type **3** in all other fields.

   In our example, PGN 3 uses the route pattern for long-distance calls during business hours. We can save money by using the trunk lines provided by our new long-distance carrier.

4. Press **Enter** to save your changes.

Now assign your new Time of Day Routing Plan 2 to the COR assigned to your executives. See Class of Restriction on page 813 to see where to assign this field.

Example

For this example, assume the following:

- Jim is the user at extension 1234.
- Extension 1234 is assigned a COR of 2.
- COR 2 is assigned a Time of Day Plan Number of 1.
- The Time of Day Routing Plan 1 is administered as shown in the example above.

When Jim comes into work on Monday morning at 8:30 and makes an ARS call (dials the ARS access code followed by the number of the person he is calling), the system checks the Time of Day Plan Number assigned to Jim's COR.

Because Jim has a COR of 2 with Time of Day Plan Number 1, the system uses Time of Day Routing Plan 1 to route the call.

According to Time of Day Routing Plan 1, calls made between 8:00 a.m. and 11:59 a.m. route according to the route pattern set up on PGN 1.

If Jim makes a call between 12:00 p.m. and 1:00 p.m. on Monday, the Time of Day Routing Plan 1 is used again. However, this time the call is routed according to PGN 2.

Related topics

See Route Pattern on page 1367 screens for more information.

See Defining ARS Partitions on page 277 to see how to set up partition groups.

Location by Region

Location by Region provides a way to administer location by IP network region. This allows for the correct date and time information and trunk routing based on the IP network region.

Setting up a remote user by network region and time zone

Instructions

With your system located in New York and a remote user located in Germany, to create the correct time zone settings:

1. Type **change locations** and press **Enter**.

   The **Locations** screen displays.

Figure 120: Locations screen

```
change locations                                          Page   1 of   1
                                LOCATIONS

               ARS Prefix 1 Required For 10-Digit NANP Calls? y

 Loc. Name            Timezone  Daylight-  Number ARS Attd Loc.   Pre- Foreign
 No                    Offset   Savings    Plan   FAC FAC  Parms. fix  Domain
                                Rule       Area                      Route
                                           Code                      Pattern
   1  Main            + 00:00   0
 xxx ________________ _ __:__   __         ___    ____ ____ __    ____ ________
 xxx ________________ _ __:__   __         ___    ____ ____ __    ____ ________
 xxx ________________ _ __:__   __         ___    ____ ____ __    ____ ________
 xxx ________________ _ __:__   __         ___    ____ ____ __    ____ ________
```

2. On line 11, in the **Name** field, enter the Communication Manager server associated with the location (for instance, **Germany**).
3. In the first **Timezone Offset** field, enter **+** to indicate the time is ahead of the system time.
4. In the second **Timezone Offset** field, enter **08** for the number of hours difference between this location and system time.
5. In the **Daylight Savings** field, enter **1** if this country has daylight savings.
6. Press **Enter** to save your changes.
7. Type **change ip-network-map** and press **Enter**.

   The **IP Address Mapping** screen displays.

Figure 121: IP Address Mapping screen

```
change ip-network-map                                          Page 1 of X

                          IP ADDRESS MAPPING
                                                                  Emergency
                                        Subnet            802.1Q  Location
FROM IP Address   (TO IP Address or Mask)   Region     VLAN      Extension
  1.__2.__3.__0    1.__2.__3.255    24       __1        ___3      ________
  1.__2.__4.__4    1.__2.__4.__4    32       __2        ___0      ________
  1.__2.__4.__5    1.__2.__4.__5    __       __3        ___0      ________
  1.__2.__4.__6    1.__2.__4.__9    __       __4        ___4      ________
___.___.___.___  ___.___.___.___    __       ___        ____      ________
___.___.___.___  ___.___.___.___    __       ___        ____      ________
___.___.___.___  ___.___.___.___    __       ___        ____      ________
___.___.___.___  ___.___.___.___    __       ___        ____      ________
___.___.___.___  ___.___.___.___    __       ___        ____      ________
___.___.___.___  ___.___.___.___    __       ___        ____      ________
___.___.___.___  ___.___.___.___    __       ___        ____      ________
___.___.___.___  ___.___.___.___    __       ___        ____      ________
___.___.___.___  ___.___.___.___    __       ___        ____      ________
```

8. In the **From IP Address** field, enter the IP address for the remote station in Germany.
9. In the **To IP Address** field, enter the IP address of your system.
10. In the **Subnet or Mask** field, enter the subnet mask value of your network.
11. In the **Region** field, enter a number that is not being used. In this example, enter **3**.
12. Press **Enter** to save your changes.
13. Type **`change ip-network-region 3`** and press **Enter.**

    The **IP Network Region** screen displays.

Figure 122: IP Network Region screen

```
change ip-network-region 3                                    Page 1 of 19
                              IP NETWORK REGION
  Region: 3
Location: 11                     Home Domain:
    Name: North
                                Intra-region IP-IP Direct Audio: no
AUDIO PARAMETERS                Inter-region IP-IP Direct Audio: no
   Codec Set: 1                           IP Audio Hairpinning? y
UDP Port Min: 2048
UDP Port Max: 3049                       RTCP Reporting Enabled? y
                                  RTCP MONITOR SERVER PARAMETERS
DIFFSERV/TOS PARAMETERS           Use Default Server Parameters? y
 Call Control PHB Value: 34                Server IP Address:   .   .   .
        Audio PHB Value: 46                        Server Port: 5005
802.1P/Q PARAMETERS                     RTCP Report Period(secs): 5
 Call Control 802.1p Priority: 7
        Audio 802.1p Priority: 6      AUDIO RESOURCE RESERVATION PARAMETERS
H.323 IP ENDPOINTS                                       RSVP Enabled? y
  H.323 Link Bounce Recovery? y              RSVP Refresh Rate(secs): 15
 Idle Traffice Interval (sec): 20     Retry upon RSVP Failure Enabled? y
    Keep-Alive Interval (sec): 6               RSVP Profile:
guaranteed-service
             Keep-Alive Count: 5      RSVP unreserved (BBE) PHB Value: 40
```

14. In the **Name** field, enter the location name for familiarity.
15. In the **Location** field, enter the number from the **Locations** screen. In this example, it was **11**.
16. Press **Next Page** until you get to page 3, the **Inter Network Region Connection Management** screen.

Figure 123: Inter Network Region Connection Management screen

```
change ip-network-region 3                                    Page   3 of 19

                  Inter Network Region Connection Management

 src dst
 rgn rgn      codec-set  direct-WAN  WAN-BW limits  Intervening-regions
 3   1           1           y          256:Kbits
 3   2           1           n                      1    ___  ___  ___
 3   3           1
 3   4           1           n                      1    ___  ___  ___
 3   5           1           n                      6    ___  ___  ___
 3   6           1           y             :NoLimit
 3   7           1           y           10:Calls
 3   8
 3   9
 3   10
 3   11
 3   12
 3   13
 3   14
 3   15
```

17. Notice in the **src rgn** column that a "3" displays, and under **dst rgn** a "1," indicating that Network Region 3 (Germany) is connected to Network Region 1 (New York) using Codec Set 1.
18. Press **Enter** to save your changes.

Related Topics

See *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205*,* for more information on the Multiple Locations feature.

Chapter 9: Managing MultiMedia Calling

Multimedia Applications Server Interface

The Multimedia Applications Server Interface (MASI) defines a protocol and a set of operations that are used to extend Avaya Communication Manager feature functionality to a Multimedia Communications Exchange (MMCX) system. MASI architecture fits the client/server model, where Avaya Communication Manager functions as a server for MMCX clients. Examples of features supported by MASI include call detail recording (CDR), AUDIX/INTUITY voice mail integration, and Automatic Alternate Routing (AAR)/Automatic Route Selection(ARS).

MMCX can make use of both MASI features and MMCX autonomous features. Autonomous features are those that MMCX provides, even if MASI is not enabled. This document does not discuss them unless there is a consideration for MASI administration.

Some autonomous MMCX features:

- Basic Call (Place/Drop)
- Call Coverage
- Conference
- Transfer

Avaya Communication Manager/MASI features:

- Basic Call (Place/Drop) - Avaya Communication Manager tracks the status of all calls placed to or from a MASI terminal.
- Call Detail Recording - Avaya Communication Manager tracks calls to and from MASI terminals and can produce call records that indicate if a call uses MASI.
- Call Coverage - Avaya Communication Manager tracks MMCX calls that are sent to coverage. A Communication Manager coverage path can contain both MASI terminals and Communication Manager stations.
- Conference - Avaya Communication Manager tracks conference calls that involve MASI terminals, if a Communication Manager station originates the conference. Conferences that involve MASI terminals and Communication Manager stations are voice-only. If the Communication Manager station originates the call, the caller can use the consultative form of conference or transfer.
- World Class Routing (AAR or ARS) - Calls from MASI terminals can take advantage of Avaya Communication Manager World Class Routing capabilities.

- Voice messaging access to AUDIX/INTUITY - MMCX users can take advantage of AUDIX voice messaging, and receive message waiting indication.
- MMCX trunking - By assigning trunk access codes to interfaces from the MMCX to other MMCXs or the PSTN, Avaya Communication Manager can monitor traffic over those interfaces.

Before you start

CAUTION:

Avaya Communication Manager offers a wide range of features, and MMCX users may want to take advantage of this. In some cases, these features will operate as expected. However, some features are not supported for use over the MASI link, and their behavior is unpredictable. You may cause harm to your system by attempting to use these features. The Interactions section contains a list of features, and lists those features that are absolutely not supported for use with MASI. If you administer features on the DO NOT ADMINISTER list, Avaya cannot be responsible for the result.

For purposes of administration, there are feature buttons and groups of users that you must not administer with MASI terminal extensions. There are also features that you simply cannot administer for a MASI terminal, because the software does not allow it.

About this document

The following section describes the Multimedia Applications Server Interface (MASI), and provides instructions on how to set it up, including administration and monitoring. It also includes a section on troubleshooting.

You need to use both the DEFINITY Server system administration terminal (SAT) and the MMCX administration terminal to administer MASI. This document describes what you need to do at the DEFINITY Server SAT. It also occasionally mentions administration that you must do at the MMCX administration terminal. For more detailed MMCX information, see the *MMCX System Administrator's Guide & Technical Reference Manual*.

List of terms

This is a list of terms that are specific to MASI, or that have meanings in the context of MASI that are not standard.

- **chgmasi** - The command you use to administer MASI at the MMCX administration terminal.
- **Interserver** - Connections between MMCX terminals on different MMCX servers/nodes.
- **MASI domain** - A MASI domain consists of Avaya Communication Manager and one or more MASI nodes that share the same dial plan. That is, the extension numbers on the MMCX are known to Communication Manager, and fit in the Communication Manager dial plan.
- **MASI interworking** - MASI interworking refers to the completion of a voice connection within Avaya Communication Manager, involving at least one MASI terminal and a MASI path.
- **MASI link** - The connection between the MMCX and Avaya Communication Manager.
- **MASI node** - A single MMCX server. You may connect more than one MASI node to a Communication Manager. Each node has a separate number. This node number needs to be consistent whenever referring to a specific MMCX server.
- **MASI non-interworking** - MASI non-interworking refers to the completion of a call by MMCX, not involving a MASI path.
- **MASI path** - The Integrated Services Digital Network (ISDN) B-channels between MMCX and Avaya Communication Manager in a MASI environment. Paths are used for voice and data connections between Avaya Communication Manager and MMCX.
- **MASI signaling link** - ISDN D-channel used to transport a new ISO protocol called the MASI protocol between Avaya Communication Manager and the MMCX.
- **MASI terminal** - The representation in Avaya Communication Manager of MMCX terminals in a MASI environment.
- **MMCX interface** - PRI interface for connecting an MMCX server to other public, private or wide area network (WAN) switching systems or equipment that is part of the public network. Similar to a Communication Manager trunk group. These may include non-MASI trunks connecting Avaya Communication Manager and the MMCX.
- **MMCX trunk** - The representation in Avaya Communication Manager of trunk or network facilities terminating on MMCX. For purposes of MASI, they are called "interfaces."

Planning for MASI

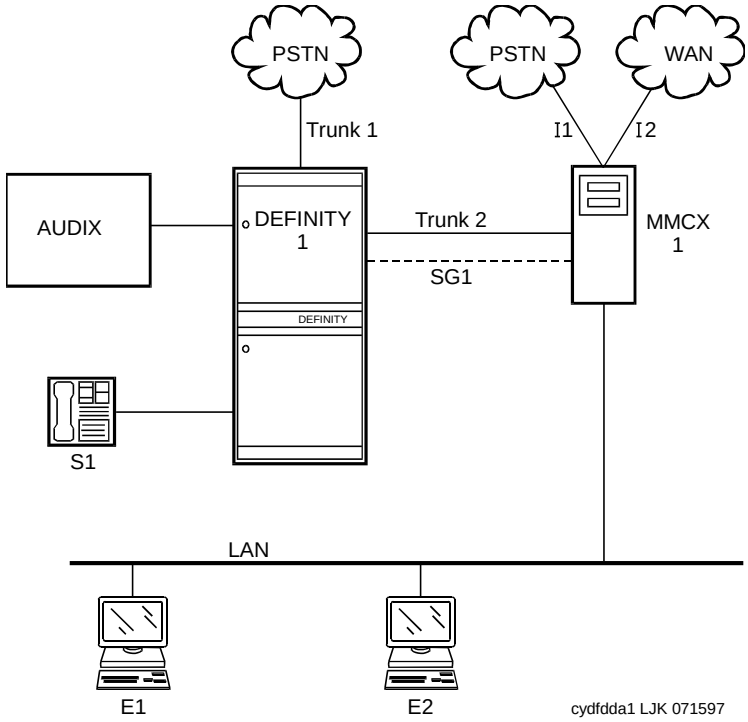
Before you start to administer MASI, you should make a plan for how to do it. Among the configurations on the following pages, there is probably one that matches the configuration of your system fairly closely. You might want to either write on these pages, or draw up your own configuration. It may help you if you have already determined trunk group and signaling group numbers, unused extensions, and so on. The following are things you need to consider:

- Establish the dial plan on the MMCX to agree with that of Avaya Communication Manager. If you use Universal Dial Plan and MMCX, you may need to make adjustments for the MMCX dial plan.
- Find unused extensions and trunk group numbers. You need:
  - one trunk group number for each ISDN-PRI connection to the MMCX.
  - one signaling group number for each MASI node and an unused Communication Manager extension for the signaling group.
  - one unused Communication Manager extension for the Near-End Path Termination number for all MASI Paths to this ECS. You can use the same number for all MASI nodes in the domain.
  - two unused MMCX extensions for the nearpath and tscnum arguments to the **chgmasi** command. This is the command you use to administer MASI on the MMCX.

MASI configurations

There are several ways to set up combinations of MASI nodes and DEFINITY servers.The following figures depict several possible configurations.

Figure 124: MASI domain of Avaya Communication Manager running on one DEFINITY Server and one MMCX



The parts of this drawing, for MASI, are as follows:

- Trunk 1 — This is any type of trunk connection to the public network.
- Trunk 2 — This is the link between the Avaya Communication Manager solution and the MMCX, and requires a TN464C or later DS1 circuit pack. You administer this link as an ISDN-PRI trunk group, a MASI path and an NCA-TSC.
- I1 and I2 — These are MMCX interfaces to destinations other than Avaya Communication Manager. Administer as MASI trunks.
- E1 and E2 — Endpoints (terminals) belonging to the MMCX. Administer as MASI terminals.
- MMCX — Determine a node number for each MMCX server. This can be any number from 1 to 15. Once the node number is established, Avaya Communication Manager informs the MMCX of its node number.
- S1 — Avaya Communication Manager station.

Figure 125: MASI domain of Avaya Communication Manager running on one DEFINITY Server and two (or more) MMCXs

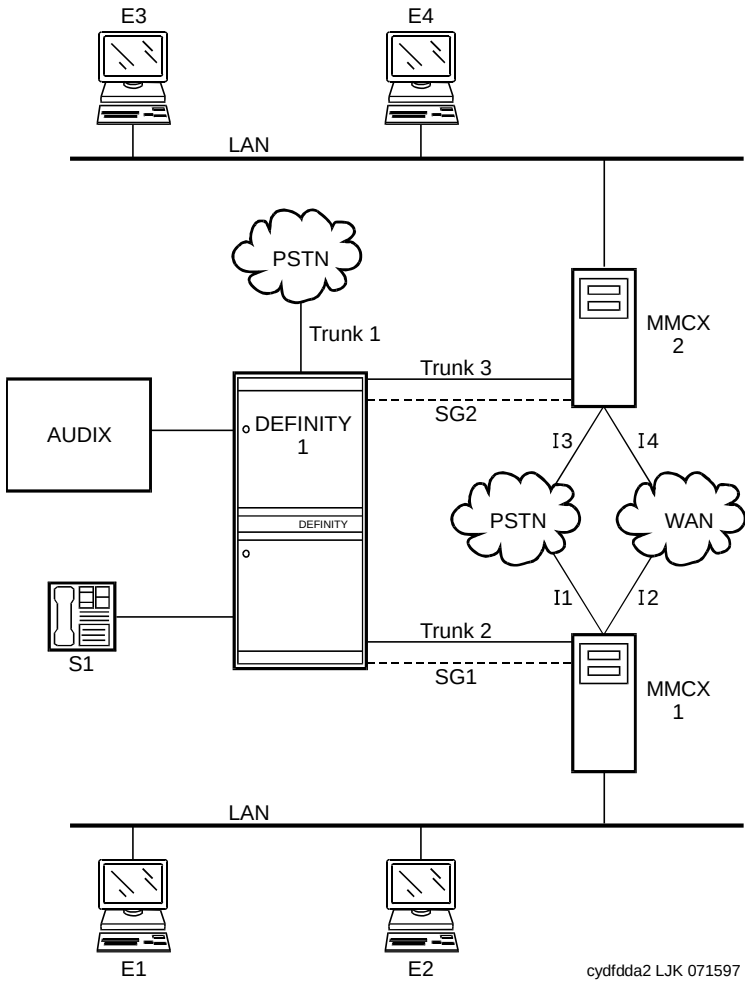


Figure 126: Two separate MASI domains

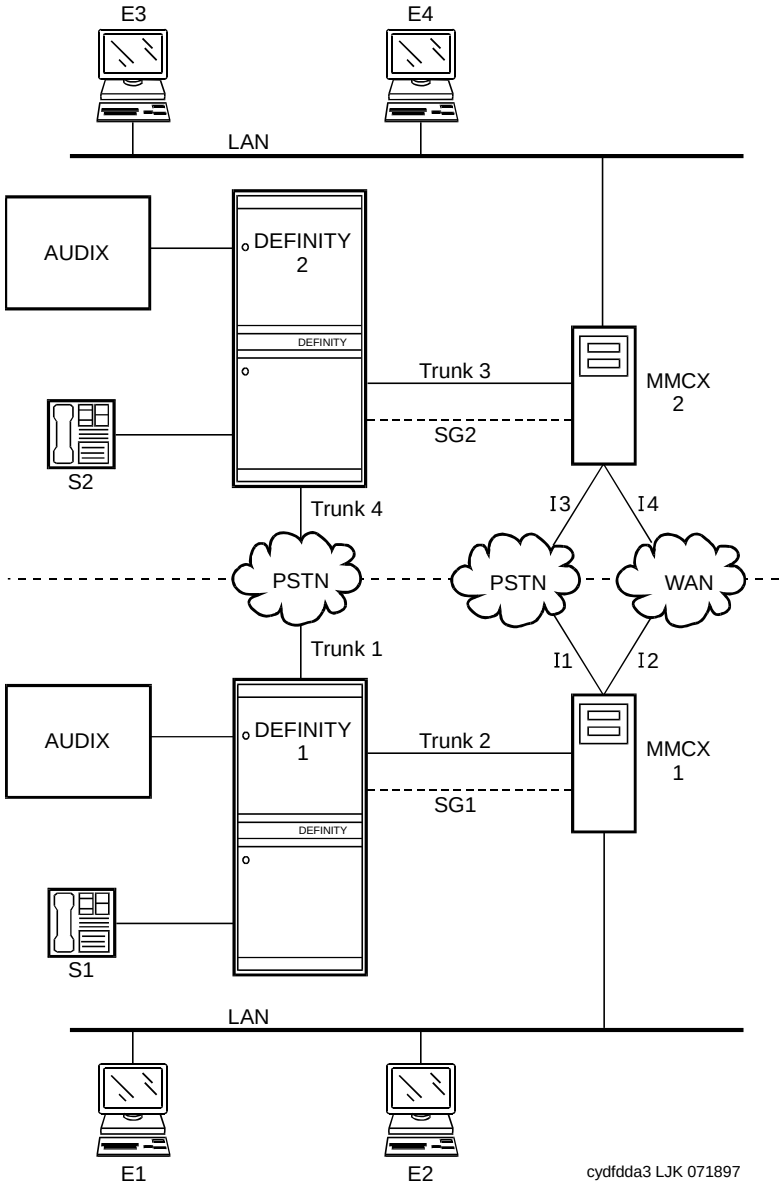
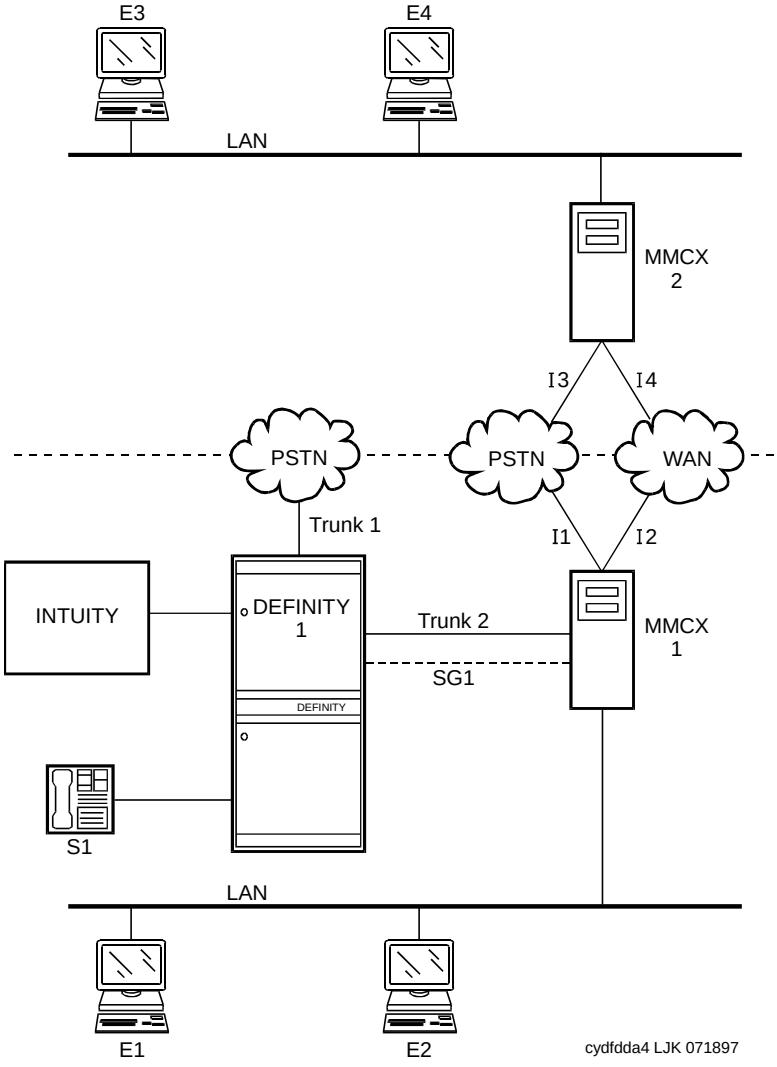


Figure 127: One MASI domain, and one non-MASI MMCX



The MASI node must be directly connected to the Avaya DEFINITY Server for MASI features to work. In this configuration, terminals that belong to MMCX 2 (E3 and E4) do not take advantage of MASI capabilities.

How to administer MASI

This section discusses the administration required to make MASI work. You perform most of this administration from the DEFINITY Server administration terminal. However, there are a few things you must do at the MMCX administration terminal. This section sometimes refers to the **chgmasi** command. This is the command you use to administer MASI parameters on the MMCX. For more information about using the chgmasi command, see the *MMCX System Administrator's Guide & Technical Reference Manual*.

Step 1 — Establishing customer options (Avaya)

An Avaya technical support representative must activate MASI using the **System-Parameters Customer-Options** screen. The technical support representative should also verify that ISDN-PRI (Avaya Communication Manager on a DEFINITY Server R configuration) or ISDN-PRI over PACCON (for DEFINITY Server SI/DEFINITY Server CSI configurations), and AAR/ARS are enabled.

On the MMCX, MASI must be enabled using the **chgmasi** command.

Step 2 — Establishing maintenance parameters and alarming options (Avaya)

Ensure that on the **Maintenance-Related System Parameters** screen, the **Packet Bus Activated** field is **y**.

Using the **set options** command (Avaya init or inads logins only), set **MASI alarming options**. For more information, see the *Maintenance for Avaya Communication Manager and DEFNITY®/S8000-series Media Server* document for your specific model and release.

Step 3 — Establishing the physical connection

Establish the physical connection between the Avaya DEFINITY Server and the MMCX. For more information about installing the cables, see *Installation for Adjuncts and Peripherals for Avaya Communication Manager*.

Step 4 — Administering circuit pack

Using the **DS1 Circuit Pack** screen, verify that the DS1 circuit pack you use to establish the MASI link is administered as follows:

- Bit Rate = 1.544
- Line Coding = b8zs
- Line Compensation = 1
- Signaling Mode = isdn-pri
- Interface = network
- Country Protocol = 1
- Protocol Version = a

Step 5 — Administering a signaling group

For each MASI node, you need to establish a unique signaling group. Use the command **`add signaling-group xxx`** to access the **Signaling Group** screen.

Figure 128: Signaling Group (Page 1 of 5)

```
 Page 1 of 5
                              SIGNALING GROUP
Group Number: _     Associated Signaling? y            Max number of NCA TSC: __
                       Primary D-Channel: ______       Max number of CA TSC: __
                                                     Trunk Group for NCA TSC: __
       Trunk Group for Channel Selection: ___
        Supplemental Service Protocol: ___     Network Call Transfer?
```

For each link, establish a Non-Call Associated Temporary Signaling Connection (NCA-TSC) with the following attributes:

- **Associated Signaling** - MASI requires Facility Associated Signaling, so this field must be set to **y**.
- **Primary D-channel** - Enter a 6- to 7-character port number associated with the DS1 Interface circuit pack port. The port address of the PRI that carries D-channel signaling.

  The port number is used to assign the primary D-channel in the Signaling Group. For 24-channel facilities, the 24th port is assigned as the D-channel. For 32-channel facilities, the 16th port is assigned as the D-channel.
- **Max Number of NCA TSC** - For MASI, this must be 1.
- **Max number of CA TSC** - Leave the default of 0.
- **Trunk Group For NCA TSC** - This can be left blank.

- **Trunk Group for Channel Selection** - This can be left blank.
- **Supplemental Service Protocol** - Values are **a** (AT&T) and **b** (Q-sig).
- **Network Call Transfer?** - Values are **y** (yes) and **n** (no).

Figure 129: Administered NCA TSC Assignment page of the Signaling Group screen

```
                                                                Page 2 of 5
                          ADMINISTERED NCA TSC ASSIGNMENT
Service/Feature: ___________              As-needed Inactivity Time-out
(min):_  TSC    Local                                                Adj.      Mac
h.
 Index    Ext.   Enabled  Established   Dest. Digits    Appl.    Name       ID
  1:     _____      __     _________    _____________   _______  _______    __
  2:     _____      __     _________    _____________   _______  _______    __
  3:     _____      __     _________    _____________   _______  _______    __
  4:     _____      __     _________    _____________   _______  _______    __
  5:     _____      __     _________    _____________   _______  _______    __
  6:     _____      __     _________    _____________   _______  _______    __
  7:     _____      __     _________    _____________   _______  _______    __
  8:     _____      __     _________    _____________   _______  _______    __
  9:     _____      __     _________    _____________   _______  _______    __
 10:     _____      __     _________    _____________   _______  _______    __
 11:     _____      __     _________    _____________   _______  _______    __
 12:     _____      __     _________    _____________   _______  _______    __
 13:     _____      __     _________    _____________   _______  _______    __
 14:     _____      __     _________    _____________   _______  _______    __
 15:     _____      __     _________    _____________   _______  _______    __
```

- **Service/Feature** - Leave blank.
- **As-needed Inactivity Time-out (min)** - This field only applies to as-needed NCA-TSCs. Since MASI requires a permanent connection, leave blank.
- **TSC Index** - This display-only field specifies the administered NCA-TSCs assigned.
- **Local Ext** - Enter a valid, unassigned Avaya Communication Manager extension. This extension does not need a port assignment and does not need to correspond to any other administration.
- **Enabled** - Enter **y** to enable the administered NCA-TSC. You may want to wait to enable this link until all other administration is in place. If this is **y**, Avaya Communication Manager attempts to establish the connection as soon as you submit the form. This may cause your system to alarm, if other administration is not finished.
- **Establish** - Used to indicate the strategy for establishing this administered NCA-TSC. Enter **permanent** for MASI.

- **Dest. Digits** - A valid MMCX extension. This must correspond to the value of the *tscnum* argument to the **chgmasi** command.

  Note:

  These digits are sent as entered to the destination MMCX; no routing or other digit manipulation is performed.

- **Appl.** - Specifies the application this administered NCA-TSC is going to be used for. Enter **masi**.
- **Machine ID** - Used to indicate the MASI node to which this administered NCA-TSC is connected. This number should be the same as the MASI node number found on other screens.

How to list TSCs or determine status of the TSC

To determine which TSCs are designated for MASI, use the **list masi tsc** command.

Figure 130: MASI Temporary Signaling Connections (TSC) Display

```
                    MASI TEMPORARY SIGNALING CONNECTIONS (TSC)

Sig.  Primary   TSC      Local                                          Mach.
Grp   D-Chan    Index    Ext.   Enabled  Established   Dest. Digits      ID

xxx   xxxxxxx   xxx      xxxxx     x     xxxxxxxx      xxxxxxxxxxxxxxx   xx
xxx   xxxxxxx   xxx      xxxxx     x     xxxxxxxx      xxxxxxxxxxxxxxx   xx
xxx   xxxxxxx   xxx      xxxxx     x     xxxxxxxx      xxxxxxxxxxxxxxx   xx
```

This command displays the following:

- **Sig Grp** — The number of the signaling group to which this TSC belongs
- **Primary D-Channel** — Port location of the Primary D-channel
- **TSC Index** — The number of the MASI TSC within the signaling group
- **Local Ext.** — Avaya Communication Manager extension associated with the TSC
- **Enabled** — Indicates the state of the connection - enabled (y/n)
- **Established** — Value of established flag (as-needed/permanent)
- **Dest. Digits** — The MMCX extension that indicates the TSC destination
- **Mach. ID** — MASI node number

Once you establish and enable the signaling group, you need to verify that it is active. Use the command **status signaling-group signaling-group#** or **status tsc-administered signaling-group# [/tsc-index] [print]** to determine if the link is active.

Step 6 — Administer ISDN-PRI trunk group

Use the command **add trunk-group *xxx*** to access the **Trunk Groups** screen. For a more detailed description of the ISDN-PRI trunk group, see Trunk Group on page 1534.

Establish an ISDN-PRI trunk group with the following attributes:

Page 1

- Group Type = **isdn-pri**
- TAC = valid TAC that conforms to your existing dial plan
- Direction = **two-way**
- Service Type = **tie**
- CDR Reports = **n**

You must also administer the PRI link from the MMCX to the ECS, using the MMCX administration terminal. See the *MMCX System Administrator's Guide & Technical Reference Manual* for information on the **addpri** command.

Step 7 — Administer MASI Path Parameters

Use the **change masi path-parameters** command to access the **MASI Path Parameters** screen.

Figure 131: MASI Path Parameters screen

```
                        MASI PATH PARAMETERS

Near-End Path Extension: _____

    MASI Node      Trunk Group        Far-End Path Termination Number
        1              __                    ____________________
        2              __                    ____________________
        3              __                    ____________________
        4              __                    ____________________
        5              __                    ____________________
        6              __                    ____________________
        7              __                    ____________________
        8              __                    ____________________
        9              __                    ____________________
       10              __                    ____________________
       11              __                    ____________________
       12              __                    ____________________
       13              __                    ____________________
       14              __                    ____________________
       15              __                    ____________________
```

Establish a MASI Path with the following attributes:

- **Near-End Path Extension** — An unassigned Communication Manager extension. When using the **chgmasi** command to administer the MMCX, this is the farpath extension. See the *MMCX System Administrator's Guide & Technical Reference Manual* for more information.
- **MASI Node** — The node number for the MMCX. For each MMCX/MASI node, this number must be the same everywhere it occurs (signaling group, masi trunk group, and masi terminal screens).
- **Trunk Group** — This is the trunk group number in Communication Manager for the ISDN-PRI trunk that will be used to establish call paths.
- **Far-End Path Termination Number** — This is an unassigned MMCX extension. When using the **chgmasi** command to administer the MMCX, this is the nearpath extension. See the *MMCX System Administrator's Guide & Technical Reference Manual* for more information.

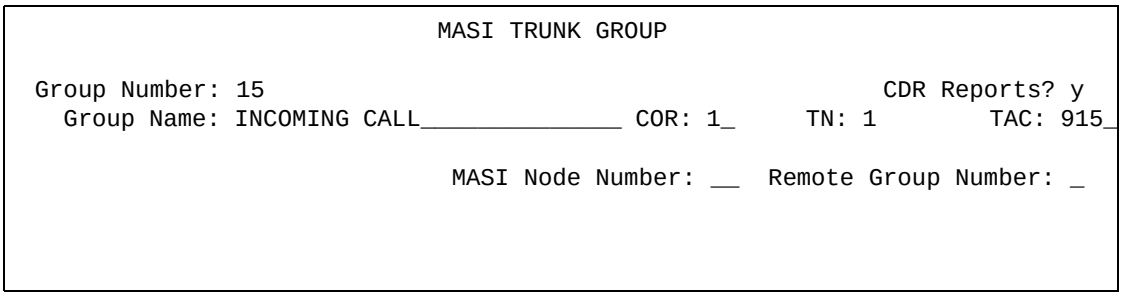
Step 8 — Administer MASI trunk groups

Use the **MASI Trunk Group** screen to define MMCX interfaces that interconnect MASI nodes, or that connect MMCX nodes to another private switch or central office. Examples of MMCX interfaces include:

- PRI trunks linking MMCX servers
- PRI trunks linking MMCX to the PSTN
- PRI trunks from MMCX to Avaya Communication Manager that are used for purposes other than MASI
- LAN interfaces linking MMCX servers

Use the command **add masi trunk-group *xxx* (or 'next')** to access the **MASI Trunk Group** screen. The trunk group number must not be assigned, and you cannot exceed the maximum total trunks for your system. Valid values for xxx are unused trunk group numbers in Avaya Communication Manager between 1 to 96 for DEFINITY Server SI/DEFINITY Server CSI configurations, and 1 to 120 for DEFINITY Server R configurations.

Figure 132: MASI Trunk Group screen



- **Group Number** - This field displays the MASI trunk group number. This is the number assigned when executing the **add masi trunk-group** command.
- **CDR Reports** - Valid entries are **y**, **n**, and **r**. Default is **y**.
  - If you enter **y**, Call Detail Recording (CDR) records will be generated by completed outgoing calls terminated on this trunk group. If incoming calls are being recorded (the **Record Outgoing Calls Only** field on the **CDR System Parameters** screen is set to **n**), then a single CDR record will be generated for answered calls with the call duration.
  - If you enter **n**, no CDR records will be generated by calls originated by or terminated on this trunk group.
- **Group Name** - Enter a unique name that identifies the trunk group. Up to 27 characters can be used; default is "**INCOMING CALL**."
- **COR** - Enter a Class of Restriction (COR) number (0 to 95) that reflects the desired restriction; default is "1."
- **TN** - This field displays the Tenant Partition number. All MASI trunks are associated with Tenant 1.
- **TAC** - Enter the trunk access code (TAC) that identifies the trunk group on CDR reports. You must assign a different TAC to each MMCX interface. Valid entries conform to the dial plan (1 to 4 digits, * and # are valid first digits).
- **MASI Node Number** — The node number assigned to this MMCX machine.
- **Remote Group Number** — This is the number of the remote trunk group. For ISDN-PRI interfaces, valid values are any number 1 to 8; for local area network (LAN) or WAN calling interfaces, the value must be 9. The combination of MASI Node Number and Remote Group Number must be unique. Remote group number corresponds to the group number on the MASI node.

How to view a list of all MASI trunk groups

To view a list of all the MASI trunks administered on the ECS, use the command **list masi trunk-group**.

Figure 133: List masi trunk-group output

```
                              MASI TRUNK GROUP

Group                                          Node    Remote
Number  TAC   Group Name                       Number  Grp No.  CDR  COR  TN

xxx     xxxx  xxxxxxxxxxxxxxxxxxxxxxxxxxx      xx      x        x    xx   xxx
```

How to determine the status of MASI trunk groups

To determine the status of a specific MASI trunk, use the command **status masi trunk-group *xxx***, where ***xxx*** is the trunk group number. This command provides descriptive information about the trunk, and the number of currently active trunk calls.

Figure 134: Status masi trunk-group output

```
                          MASI TRUNK GROUP STATUS

        Group Number: xxx       Number of Active MMCX Trunk Calls: xxx
   MASI Node Number: xx
Remote Group Number: xxx
```

Step 9 — Administer MASI terminals

Use the **add masi terminal *xxxxx*** or **next** command to administer each MASI terminal as a MASI terminal. You use available extensions on the ECS, so they need to conform to the Avaya Communication Manager dial plan. The extension must match the Communication Manager dial plan, and for the add command, the extension must not already be in use.The extension of the MASI terminal must match the number of the MASI terminal.

Avaya Communication Manager users dial the MASI Terminal Extension to reach MMCX users.

Note:

Anytime you add a terminal or other extension to the MMCX, you must administer a corresponding MASI terminal on Avaya Communication Manager. If you do not, you will not be able to dial this extension from Avaya Communication Manager.

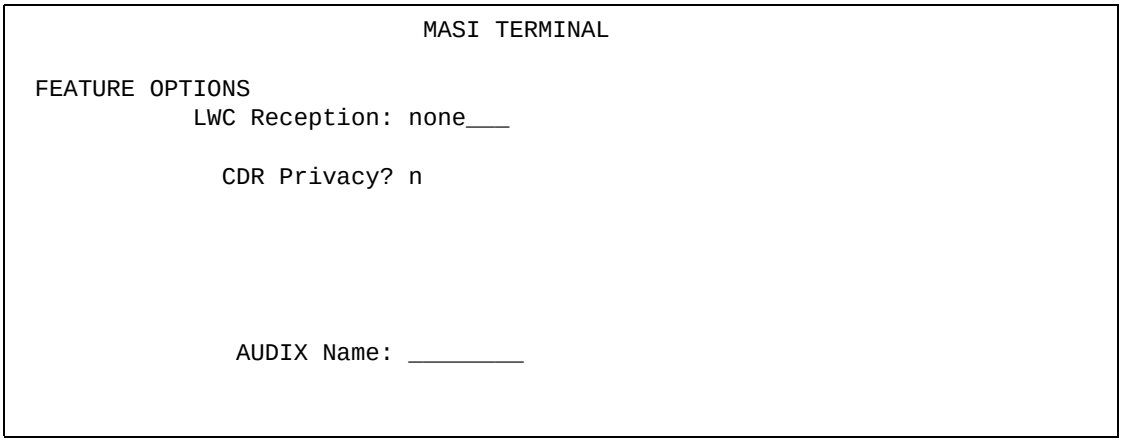
Figure 135: MASI Terminal screen — page 1

```
                            MASI TERMINAL

Extension: 1000                                          BCC: 0
                                      MASI Node Number: __ TN: 1__
                                                       COR: 1_
     Name: _____________________________


TERMINAL OPTIONS

       Send Display Info? y
```

- **Extension** — This field displays the extension that you entered on the command line.

- **BCC** — This field displays the bearer capability class of the terminal, and identifies the type of traffic the terminal supports. For MASI, this is always 0, for voice or voice-grade data.
- **MASI Node Number** — The number of the node on which this terminal resides.
- **TN** — The tenant partition in which this terminal resides. At present, all MASI terminals must reside within tenant 1. This field is display-only, and always 1.
- **COR** — The class of restriction associated with this terminal.
- **Name** — The name associated with the terminal. This can be any alphanumeric string up to 27 characters.
- **Send Display Info** — Indicates whether Avaya Communication Manager should forward display information associated with a call. Set to **y**.

Figure 136: MASI Terminal screen — page 2



- **LWC Reception** — This field indicates whether the terminal can receive Leave Word Calling (LWC) messages. Valid values are none, audix, and spe (for DEFINITY Server R configurations) or mas-spe (for DEFINITY Server SI/DEFINITY Server CSI configurations). SPE-based LWC is not supported for MASI terminals. However, if embedded AUDIX is used without a Data Control Link, you must administer MASI terminals to receive SPE-based LWC messages. For such cases, the LWC feature is used by AUDIX messaging systems to activate and deactivate message waiting lamps on MASI terminals.
- **CDR Privacy** — Indicates whether CDR Privacy is supported for this terminal. See "Call Detail Recording" in *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205 for more information.
- **AUDIX Name** — This field only appears on DEFINITY Server R configurations. This field contains the name of the AUDIX messaging adjunct for LWC messages. If the **LWC Reception** field is set to **audix**, this field must contain a name. The name must match a machine name on the **Adjunct Names** screen.

Figure 137: MASI Terminal screen — page 3

```
                              MASI TERMINAL
SITE DATA
       Room: ________
       Jack: _____
      Cable: _____
      Floor: ________
   Building: ________




BUTTON ASSIGNMENTS
 1: call-appr
```

- **Room** - Enter up to 10 characters to identify the MASI terminal location. This field may be blank.
- **Jack** - Enter up to 5 characters to identify the location where the MASI terminal is connected. This field may be left blank.
- **Cable** - Enter up to 5 characters to identify the cable that connects the MASI terminal to the system. This field may be left blank.
- **Floor** - Enter up to 7 characters to identify the floor where the MASI terminal is located.
- **Building** - Enter up to 7 characters to identify the building where the MASI terminal is located. Valid entries are listed in the site table.
- **Button Assignments** — This field contains a call appearance button and is display only.

Duplicate MASI terminal

Once you have one MASI terminal administered to your liking, you can use the **duplicate masi terminal** command to administer other stations with the same characteristics.

Figure 138: Duplicate MASI Terminal screen

```
                          MASI TERMINAL


Ext          Name                              Room       Jack  Cable
77777        _________________________________ _________  _____ _____
77778        _________________________________ _________  _____ _____
77779        _________________________________ _________  _____ _____
77781        _________________________________ _________  _____ _____
77782        _________________________________ _________  _____ _____
77783        _________________________________ _________  _____ _____
_____        _________________________________ _________  _____ _____
_____        _________________________________ _________  _____ _____
_____        _________________________________ _________  _____ _____
_____        _________________________________ _________  _____ _____
_____        _________________________________ _________  _____ _____
_____        _________________________________ _________  _____ _____
_____        _________________________________ _________  _____ _____
_____        _________________________________ _________  _____ _____
_____        _________________________________ _________  _____ _____
```

How to list and status MASI terminals

To view a list of all the MASI terminals administered on a server, use the command **`list masi terminals`**.This command only lists terminals within the domain of the Avaya DEFINITY Server from whose SAT you issue the command.

Figure 139: List MASI Terminal output

```
                          MASI TERMINALS


                                      Node
      Ext   Name                      Number CDR COR TN

      xxxxx xxxxxxxxxxxxxxxxxxxxxxxxx xx     x   xx  xxx
      xxxxx xxxxxxxxxxxxxxxxxxxxxxxxx xx     x   xx  xxx
      xxxxx xxxxxxxxxxxxxxxxxxxxxxxxx xx     x   xx  xxx
      xxxxx xxxxxxxxxxxxxxxxxxxxxxxxx xx     x   xx  xxx
```

To view the active or idle status of a specific MASI terminal, use the command **`status masi terminal (extension)`**.

Figure 140: Status MASI terminal command

```
                              GENERAL STATUS

                 TYPE: MASI        Service State: active
            Extension: 54001
      MASI Node Number: 14
```

To determine which extension you assigned as the MASI Near-End Path Termination extension, use the command **`list extension-type`**. This command displays the extension number and type (attendant, masi-terminal, etc.), as well as other information about the extension.

Figure 141: List extension type

```
                              EXTENSION TYPE

                                                                  Cv1/
Ext   Type                 Name                    COR  TN  COS  Cv2
---   ----                 ----                    ---  --  ---  ----
1234  masi-terminal                                1    1   1
4077  term-masi-path-call
```

Step 10 — Administer features

AAR/ARS

1. AAR/ARS is an optional feature on Avaya Communication Manager, and you need to purchase this option to use it with MMCX. To verify that this feature is enabled, use the command **`display system-parameters customer-options`**. If it is not enabled, contact your Avaya representative.
2. The MMCX dial plan must use the same feature access codes as Avaya Communication Manager. If this is not already the case, modify the MMCX dial plan using the **`chgdp`** command. See the *MMCX System Administrator's Guide & Technical Reference Manual* for more information.
3. Include this feature access code in the **`chgmasi`** command.

CDR

1. To get call detail records for calls over MMCX interfaces, set **CDR Reports = y** on the **MASI Trunk Group** screen.
2. To get call records for calls over the ISDN-PRI trunk group, set **CDR Reports = y** on the **ISDN-PRI Trunk Group** screen.
3. To track calls between a MASI terminal and other MASI terminals or Communication Manager stations, enter the MASI terminal extension on the **Intraswitch CDR** screen.
4. Enter **n** in the **Record Non-Call Assoc TSC** field on the **CDR System Parameters** screen.

Note:

If you use the same PRI trunks for MASI and non-MASI calls, Avaya strongly recommends that you do not enable CDR for these calls. Establish a separate trunk group for non-MASI calls and set CDR Reports = **n**.

Coverage

To establish coverage from a MASI terminal to AUDIX:

1. Use the MMCX user interface to enter the AUDIX hunt group extension as the coverage point. You cannot use Avaya Communication Manager coverage administration for MASI terminals.
2. If AUDIX ports are not administered in Avaya Communication Manager, you must administer them.
3. Set up the MASI terminal as an AUDIX subscriber. Enter the MASI terminal extension in the **Extension** field on the **Subscriber Administration** screen.

To establish coverage from a MASI terminal to another MMCX terminal or Avaya Communication Manager station:

1. Use the MMCX user interface to enter the desired extension as the coverage point for the MASI terminal. You cannot use Avaya Communication Manager coverage administration for MASI terminals.

Step 11 — Verify administration

You should make test calls from Avaya Communication Manager to MMCX, to ensure that you can indeed place and receive calls.

Call an unattended MASI terminal. Verify that the call goes to AUDIX. Retrieve the call from the MASI terminal. Verify that all works as expected.

MASI command permissions

If you are the super-user for your system, you can restrict other administrative logins from changing MASI administration. To do this, use the **change permissions (login-ID)** command. Enter **y** in the **Additional Restrictions** field, then move to the **Restricted Object List** page of the screen.

You may restrict the following MASI-related objects:

- masi-path-parameters
- masi-terminal
- masi-trunk-group
- masi-tsc

Detailed description of features

AAR/ARS

MMCX can take advantage of advanced routing features for voice-only calls to the public switched telephone network (PSTN) or an Avaya private network. Users must enter the AAR/ARS access code before the rest of the dialed digits. MASI will route the call over the Communication Manager private network (AAR) or the public network (ARS), based on digits supplied by the MMCX user.

Routing patterns must contain only trunk groups that actually terminate to Avaya Communication Manager. Calls from one MMCX to another MMCX do not use AAR/ARS. Authorization codes are not supported.

Call Detail Recording

Using the MASI link, Avaya Communication Manager is able to track call detail information for calls made using MMCX terminals and interfaces. CDR records all calls originating from or terminating at a MASI terminal. MASI CDR does not record ineffective call attempts when all MASI paths are busy.

The **Resource Flag** value of **8** indicates a MASI call. This field appears in unformatted, int-isdn, expanded and customized CDR formats. For formats other than these, you can determine that a call involves a MASI terminal or trunk by the trunk access code (TAC), dialed number or calling number fields.

The following are the CDR capabilities of MASI. Administration information is under the heading How to administer MASI on page 297.

- Incoming/Outgoing Trunk Call Splitting

  Call splitting does not produce separate records for MMCX calls that are transferred or conferenced.

- Intraswitch CDR

  You can administer intraswitch CDR to monitor MASI terminals. To do this, simply add the MASI terminal extension on the **Intraswitch CDR** screen. Avaya Communication Manager then monitors calls from MASI terminals to other MASI terminals, and calls between MASI terminals and Communication Manager stations.

- CDR Privacy

  You can administer a MASI terminal for CDR Privacy.

- Account Code Dialing and Forced Entry of Account Codes

  This is not supported for MASI terminals. Therefore, make sure the COR you assign does not force entry of account codes.

- Trunk CDR

  You can get call detail records for all incoming and outgoing calls made over MMCX interfaces.

Call redirection / Voice-messaging access

MMCX users can enter an Avaya Communication Manager extension, including an AUDIX hunt group, Callmaster agent, attendant console or telephone as their coverage point.

If AUDIX is established as the MASI terminal's coverage point, the MASI terminal receives message waiting indication, and dials the AUDIX hunt group extension to retrieve messages. Once connected to AUDIX, operation for the MMCX user is the same as for a Communication Manager station user, including use of # to identify the extension, if desired.

Note:

It is not possible to determine the call coverage status of a MASI terminal.

Avaya Communication Manager tracks calls to MASI terminals that follow the autonomous coverage path from the MASI terminal. MMCX calls redirected to Communication Manager stations contain display information.

MASI terminals that dial AUDIX directly, or that place calls to MASI terminals that cover to AUDIX, do not receive ringback if all AUDIX ports are busy. Instead, these callers see a message the called party is busy, and the call drops.

Transfer

MASI terminals cannot transfer calls to Communication Manager stations, and cannot transfer a call to another MASI terminal if the call involves a Communication Manager station.

Conferencing

Conferences can involve both MASI terminals and Avaya Communication Manager stations, and either one may initiate the conference. Communication Manager stations participate in such conferences in voice-only mode. If an MMCX user initiates a conference that involves Communication Manager stations, the conference will drop when the initiator drops from the call. If a Communication Manager station initiates the conference, that station may drop without affecting the other conferees.

Status tracking - terminals and trunks

Avaya Communication Manager tracks the active/idle status of all MASI terminals, and monitors traffic over MMCX interfaces.

Trunk groups

For MASI purposes, there are two kinds of trunk groups: the ISDN-PRI trunk groups that serve as paths for establishing calls between Avaya Communication Manager stations or trunks and MASI terminals or interfaces, and the remote trunks that are interfaces from the MMCX to other entities. Each MASI remote trunk group appears to Communication Manager as a single unit, with no concept of members within the group.

Note:

You cannot test, busy out, or release MASI remote trunk groups, since you cannot dial a MASI remote trunk TAC from the Avaya DEFINITY Server. The TAC merely identifies the trunk to Avaya Communication Manager for purposes of status and CDR records.

You cannot administer MASI trunks as part of Communication Manager route patterns.

Interactions & Unsupported Features

We can generalize feature interactions to some extent. For example, since there are no buttons available to a MASI terminal, any feature that requires a button is also not available. MASI cannot support features that require the user to dial a trunk access code for a MASI remote trunk, or a feature access code other than AAR/ARS. The MMCX dial plan may contain only those feature access codes that are supported. Note the following CAUTION.

CAUTION:

DO NOT ADMINISTER the following features! The following features are not supported for use over the MASI link, and Avaya cannot be responsible for the results if you attempt to administer them.

Unsupported Call Center features

- ASAI — You must not administer a MASI terminal in an ASAI domain. MASI terminals and MMCX trunks are not monitored by ASAI. It may be possible for a MASI terminal to place a call to a Communication Manager station that is part of an ASAI domain. ASAI will not be blocked from controlling this call, but there may be unpredictable results. The same is true for calls originating from an ASAI domain terminating at MASI terminals, and for ASAI-monitored hunt groups that contain MASI terminals.
- Automatic Call Distribution — You must not include a MASI terminal extension as part of an ACD hunt group. You must not mix MASI administration with anything related to ACD, including Outbound Call Management and PASTE.
- Call Vectoring — You must not include MASI terminal extensions in any step of a vector.

Unsupported Basic features

- Bridged Call Appearances — You must not administer a bridged appearance that involves a MASI terminal.
- Call Coverage — You must not administer a MASI terminal in the coverage path of an Avaya Communication Manager station.
- Call Forwarding — You must not forward a Communication Manager station to a MASI terminal.
- Call Pickup — You must not administer a MASI terminal as part of a pickup group.
- Intercom — You must not administer MASI terminals as members of any type of intercom group.
- Manual Message Waiting — You must not administer a manual message waiting button (man-msg-wt) with a MASI terminal as the referenced extension.
- Manual Signaling — You must not administer a manual signaling button (signal) with a MASI terminal as the referenced extension.
- Night Service — You must not administer a MASI terminal as a night service destination.

- Pull transfer — MASI terminals cannot perform a pull transfer operation. You must not administer this feature on an Avaya DEFINITY Server where MASI is active. This applies only in Italy.
- Station Hunting — You must not administer a MASI terminal as part of a station hunting path.
- Terminating Extension Groups — You must not administer a MASI terminal as part of a TEG.

Other interactions

The following section describes feature behaviors that may not be as expected, but that are not likely to be destructive.

Attendant features

- Dial Access to the Attendant — MASI terminals will be able to dial the attendant access code, if it is administered in the MMCX dial plan.
- Attendant Direct Extension Selection — Attendants are able to access MASI terminals via DXS buttons and busy lamp indicates status of the MASI terminal.
- Emergency Access to the Attendant — MASI terminals have emergency access using the attendant access code, if it is administered in the MMCX dial plan. However, off-hook alerting is not administrable.
- Attendant Intrusion — Attendants are able to activate intrusion towards MASI terminals.
- Attendant Override — Attendants are not able to activate override towards MASI terminals.
- Attendant Recall — MASI terminals cannot activate attendant recall.
- Attendant Remote Trunk Group Select — Attendants cannot use this feature to select MASI remote trunks.
- Attendant Return Call — Operates normally if a MASI terminal is the called party.
- Attendant Serial Call — Serial calls are denied if the calling party is an MMCX interface.
- Attendant Straightforward Outward Completion — The attendant is able to complete calls to Communication Manager trunks for MASI terminals.
- Attendant Through Dialing —The attendant can use Through Dialing to pass dial tone to MASI terminals.
- Attendant Timers — Attendant timers work the same no matter what kind of terminal is involved.

- Attendant Trunk Group Busy/Warning Indicators — You cannot administer Busy/Warning indicators for MASI trunks because they are not standard Avaya Communication Manager trunks. However, you can administer these indicators for the trunk group administered for MASI paths.
- Attendant Trunk Identification — The attendant is not able to identify the trunk name via button pushes.

Basic features

- Abbreviated Dialing — A Communication Manager station can enter an MMCX extension in an AD list. However, MASI terminals cannot use AD.
- Administered Connections — MASI terminals must not be the originator nor the destination of an administered connection.
- Automatic Callback — Automatic callback does not work towards a MASI terminal.
- Automatic Circuit Assurance — You must not administer a MASI terminal as an ACA referral destination. You cannot administer ACA for MASI remote trunks.
- Busy Verification of Terminals and Trunks — You cannot use Busy Verification for MASI terminals or remote trunks.
- Call Detail Recording — CDR Account Code Dialing and Forced Entry of Account Codes are not supported for MASI terminals.
- Call Park — The attendant can park calls at the extension of a MASI terminal, but users can only retrieve these calls from a Communication Manager station, since MASI terminals cannot dial the Answer Back FAC.
- Data Call Setup — Avaya Communication Manager users cannot place data calls to MASI terminals.
- Facility Busy Indication — You can use FBI to track the status of MASI terminals. The FBI button and indicator lamp must be on a Communication Manager station. You cannot use FBI to track MMCX interfaces.
- Facility Test Calls — Avaya Communication Manager users cannot make test calls to MMCX interfaces.
- Go to Cover — MASI terminals cannot activate this feature.
- Leave Word Calling — The only valid LWC destination for a MASI terminal is AUDIX. You cannot administer SPE-based LWC. MASI terminals cannot send LWC messages to Avaya Communication Manager stations or to MASI terminals.
- Loudspeaker paging — You can administer a MASI terminal as a code calling extension.
- Malicious Call Trace — MASI terminals cannot initiate malicious call trace.
- Message Retrieval — MMCX users can only retrieve messages through AUDIX messaging.
- Music on Hold — Music on hold will only be available if an Avaya Communication Manager station has placed the call on hold.

- Override — Executive override does not work towards MASI terminals.
- Priority Calling — Priority calling is not supported for calls to or from MASI terminals.
- Ringback Queueing — Ringback Queueing is not supported for MASI terminals.
- Send All Calls — MMCX has an autonomous SAC function.
- Tenant Partitioning — All MASI terminals exist in tenant 1, and you cannot change the tenant number.
- Time of Day coverage — As with all coverage, Avaya Communication Manager does not control coverage of the MASI terminal.
- Transfer out of AUDIX — A MASI terminal cannot use *T to transfer from AUDIX to another MASI terminal.

Hospitality features

- Do Not Disturb — MASI terminals cannot activate Do Not Disturb.

Multimedia features

- Multimedia Call Handling — Avaya MMCH users are not able to make H.320 calls to MASI terminals over the MASI link. Calls between MMCX terminals and MMCH terminals are voice only.

Troubleshooting

Verify proper operation using the following commands and follow normal escalation procedures to resolve any failures detected by the demand test.

1. Verify the DS1 trunk using the **`test board <board location> long`** command.
2. Verify the ISDN Signaling Group using the **`test signaling-group <group number>`** command. Also verify proper administration.
3. Verify the temporary signaling connection using the **`test tsc-administered <group number>`** command. Also verify proper administration.

Common error conditions

If the cable from an Avaya DEFINITY Server to the MMCX becomes disconnected, you should see alarms raised against ISDN-SGRP and UDS1-BD. In particular, you should observe ISDN-SGRP errors such as 769, 1793, and 257. To resolve, reconnect the cable and follow normal test procedures.

If the far-end path termination number is incorrect, you should observe MASI-PTH error 513. To resolve, correct administration using the **MASI Path Parameters** screen.

If the Layer 3 TSC is not administered properly or is out of service, you should observe errors (but no alarms) raised against TSC-ADM. Verify the signaling group administration and follow normal escalation procedures for TSC-ADM. See the appropriate Avaya Communication Manager Maintenance manual for your server for more information.

If the TSC fails to come up even through Layer 2 Signaling Group and below pass tests, you may run **`test tsc-administered <group number>`** to force a server heartbeat test, or simply wait 5 to 10 minutes for the link to recover. This situation may happen if the server running Communication Manager is rebooted or if the MASI interface is administered before the MMCX is properly administered.

You may want to use busy port and release port commands to unlock things if features are not working.

Multimedia Call Handling

Multimedia Call Handling (MMCH) enables users to control voice, video, and data transmissions using a telephone and PC. Users can conduct video conferences and route calls like a standard voice call. They can also share PC applications to collaborate with others working from remote sites.

Operations in Basic or Enhanced modes

There are two distinct levels of functionality: Basic and Enhanced. The Basic mode of operation treats a standard-protocol H.320 multimedia call as a data call. If the call is redirected, it is converted to a voice call. As a voice call, certain features are enabled, such as coverage, voice mail, and multiparty video conferencing.

The Enhanced mode of operation allows a multifunction telephone to control a multimedia call as if it were a standard voice call. Spontaneous video conferencing, call forwarding, coverage, hold, transfer and park, along with many routing features, are available to multimedia calls. Both modes of operation allow data collaboration between multiple parties using the T.120 standard protocol.

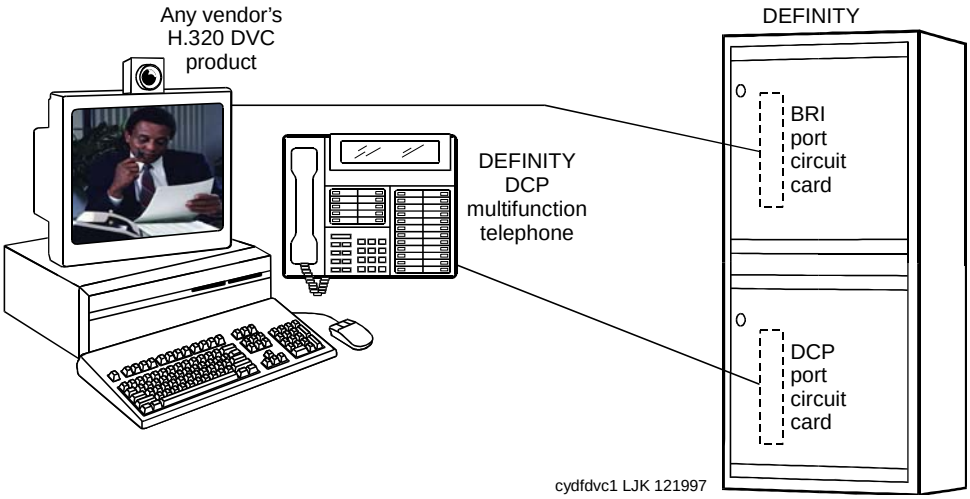
Definitions: MMCH features and components

Multimedia call

A multimedia call, for MMCH, is one that conforms to the H.320 and T.120 suite of protocol standards. These standards allow video-conferencing packages from different vendors to communicate with one another. The capabilities of the individual multimedia-endpoint package can vary, however.

- An H.320 call can contain voice, video and data.
- The bandwidth for MMCH calls is limited to 2 B-channels.

Figure 142: MMCH multimedia complex



Basic multimedia complex

A Basic multimedia complex consists of a BRI-connected multimedia-equipped PC and a non-BRI-connected multifunction telephone administered in Basic mode. With a Basic multimedia complex, users place voice calls at the multifunction telephone and multimedia calls from the multimedia equipped PC. Voice calls will be answered at the multifunction telephone and multimedia calls will alert first at the PC and, if unanswered, will next alert at the voice station. A Basic multimedia complex provides a loose integration of the voice station and H.320 DVC system.

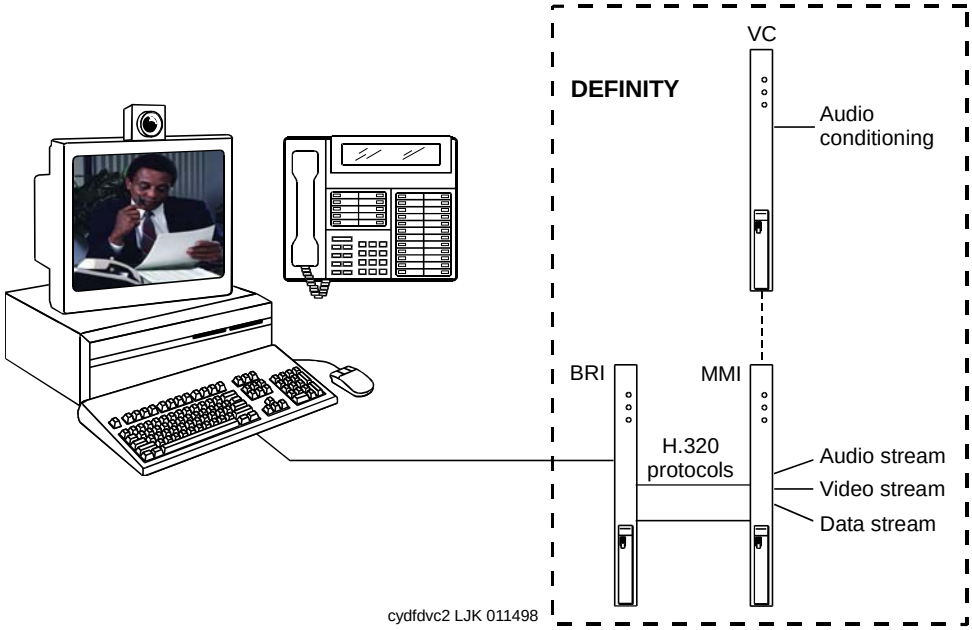
Enhanced multimedia complex

An Enhanced multimedia complex consists of a BRI-connected multimedia-equipped PC and a non-BRI-connected multifunction telephone administered in Enhanced mode. The Enhanced multimedia complex acts as though the PC were directly connected to the multifunction telephone. Thus, voice call control, multimedia call control and call status are enabled at the telephone. An Enhanced multimedia complex provides a tight integration of the voice station and H.320 DVC system.

Multimedia endpoint

The multimedia endpoint is a user's PC that has been equipped with an H.320 multimedia package. The PC is physically connected to Avaya Communication Manager with a BRI line.

Figure 143: Enhanced MMCH service link



Enhanced mode service link

The service link is the combined hardware and software multimedia connection between the user's multimedia endpoint and the Avaya DEFINITY Server which terminates the H.320 protocol. The service link provides video, data, and, optionally, voice streams to augment the capabilities of the telephone and PC. A service link only applies to an Enhanced multimedia complex, never to a Basic multimedia complex. The service link is administered on the **Station** screen and can be either "permanent" or "as-needed."

Feature Description

MMCH's two levels of functionality for a multimedia complex, Basic and Enhanced mode, are enabled either by administration on Communication Manager or by an mm-basic feature button or FAC.

Basic Mode Operation

In Basic Mode:

- All voice-only calls originate at the voice station.
- All multimedia calls originate with the H.320 DVC system.
- All incoming voice calls attempt to alert at the voice station and receive all standard voice call treatment.
- All incoming H.320 multimedia calls attempt to alert on the H.320 DVC system initially. If answered, a 2-way video call will result. The Basic multimedia complex voice station will not be involved in the call in any way.

  If the H.320 multimedia call is not answered at the H.320 DVC system and the Basic multimedia complex voice station has the **H.320** field administered to **y**, the call will:

  - Time out at the DVC system.
  - Alert at the associated voice station set as a voice-only call.
  - Receive all standard voice call treatment.
- Call control depends on what type of call is being originated.
  - Video is received and controlled at the PC.
  - Voice is received and controlled at the telephone set.
- The voice station of a Basic multimedia complex must manually add their multimedia endpoint to a multimedia conference. There is limited support for multimedia feature interactions. A specific set of voice features work for multimedia calls.
- Service Links are not used by Basic mode complexes.
- A single number may be used to reach the Basic multimedia complex for voice or H.320 multimedia calls.

Enhanced Mode Operation

The Enhanced multimedia complex provides a much more tightly coupled integration of the complex voice station and H.320 DVC system. In Enhanced Mode:

- Both multimedia and voice calls must originate at the telephone set.
- Voice and multimedia calls can be controlled at the telephone set.
- Conferencing is spontaneous and established just like a voice-only conference call.
- There is extensive support for multimedia feature interaction. Most voice features work the same for multimedia calls.
- Service Links can be either "permanent" or "as-needed."

Physical Installation

The physical components necessary to utilize MMCH capabilities include:

- H.320 DVC systems that are BRI connected to the Avaya DEFINITY Server.
- Non-BRI multifunction telephones.
- Avaya TN787 MultiMedia Interface (MMI) and TN788 Voice Conditioner (VC) boards.
- A T.120 Extended Services Module (ESM) server (necessary only if you plan to do T.120 data collaboration). Connectivity of the ESM requires an additional TN787 along with a TN2207 DS1 circuit pack.

Dual Port Desktop

Both Basic and Enhanced multimedia complexes are dual-port desktops that consist of:

- A BRI-connected multimedia-equipped PC that supports the H.320 protocol.
- A non-BRI-connected multifunction telephone set.

The PC and the multifunction telephone are individually wired to the Avaya DEFINITY Server. These two pieces of equipment can be administratively associated to form a Basic or ENHANCED multimedia complex

MMCH works with any H.320 system that is fully H.320 compliant and operates at the 2B or 128K rate.

Note:

If you intend to share applications among users or whiteboard capabilities, the endpoint software you choose must also support the T.120 protocol.

The following endpoint-software packages have been tested:

- PictureTel PCS 50 & PCS 100, Release 1.6T.
- Proshare 2.0a, 2.1.
- Zydacron Z250 Ver. 2.02, Z350 Ver. 1.2 (With Netmeeting 2.0).

MMI & VC hardware

The MMCH feature requires the use of two additional circuit packs:

- Multi Media Interface (MMI) TN787J.
- Voice Conditioner (VC) TN788B.

The TN787 and TN788 are service circuit packs. The TN787 supports simultaneous operation of 16 2B H.320 calls. The TN788 supports the voice processing for 4 H.320 endpoints.

- These service circuit packs may be located in any Port Network.
- These packs do not require any translations as part of their implementation.
- The MMI and VC circuit packs are resource circuit packs akin to the Tone Detector circuit packs.
- These circuit packs require no administration on Communication Manager and may be located in multiple port networks.

T.120 Data Collaboration Server

The Extended Services Module (ESM) provides T.120 data collaboration capability on a MMCH multipoint H.320 video conference.

- Each person in the conference who wants to participate in the data collaboration session, must have a personal computer with an H.320 video application that supports the T.120 protocol.
- The Avaya DEFINITY Server must have an ESM installed.

ESM Installation

Figure 144: Typical Multimedia Call handling ESM Connections

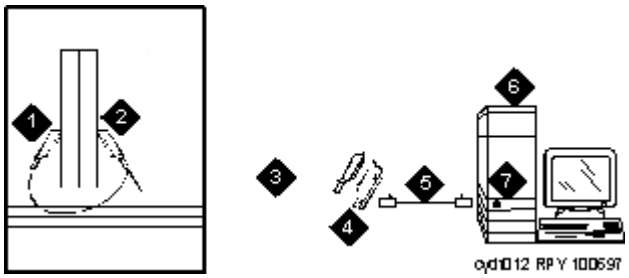


Figure notes:

1. **Port B Y-cable connector to a TN787 multimedia interface (MMI) circuit pack**
2. **Port A Y-cable connector to a TN2207 PRI circuit pack**
3. **25-pair Y-cable**
4. **356A adapter**
5. **D8W cord connected to 356A adapter S/B port 8**
6. **Extended services module (ESM)**
7. **Port B on compatible primary rate interface (PRI) card**

Use the following procedure and Typical Multimedia Call handling ESM Connections on page 323 to connect to the ESM equipment:

1. Install the TN2207 primary rate interface (PRI) circuit pack and the TN787 multimedia interface (MMI) circuit pack in the port carrier of the server for Avaya Communication Manager.

   Note:

   These two circuit packs should be co-located in the cabinet since they must be connected by a Y-cable on the back plane of the Avaya DEFINITY Server.

2. Record the circuit pack locations.
3. Connect the ESM Y-cable as shown.
4. Administer the **DS1 Circuit Pack** screen and the **Signaling Group** screen for the ESM (see ESM T.120 Server Administration on page 332).
5. Configure the ESM adjunct.

Planning for MMCH

The following are some of the tasks you perform in planning and administering MMCH.

Planning the system

Questions to help you use Avaya Communication Manager for multimedia are:

- How many MMCH users are you going to have?
- How many multimedia calls do you expect to have at any given time?

With the information above you can determine how many Voice Conditioner (VC) and Multimedia Interface (MMI) circuit packs you need.

- Will users need data collaboration capabilities? If so, you need to install the Extended Services Module (ESM).
- Which stations, hunt groups or vectors need early answer?
- Do you have ISDN-PRI trunks? It is possible to use separate DS1 trunks for data, but ISDN-PRI trunks are recommended.

Installation checklist

1. Purchase MMCH right-to-use.
2. Avaya — enable MMCH on **System Parameters Customer-Options** screen.
3. Administer default multimedia outgoing trunk parameter selection on the **Feature-Related System-Parameters Features** screen.
4. Administer MMCH related feature access codes on the **Feature Access Code** screen.
5. Install and administer hardware:
   - Install MMIs, VCs and the ESM.
   - Administer the ESM to ECS connection — **DS1 Circuit Pack** and **Signaling Group** screens.
   - Establish maintenance parameters — **Maintenance-Related System Parameters** screen.
6. Administer multimedia complexes:
   - Administer data modules — **Data Module** screen, or **Data Module** page of the **Station** screen.
   - Administer stations as part of a multimedia complex, assign associated data module extension, multimedia mode, service link mode and appropriate multimedia buttons — **Station** screen.

7. Administer early answer and H.320 flag for stations, the early answer flag for hunt groups, and the multimedia flag for vectors as appropriate.
8. Train end users.
9. Monitor traffic and performance.

Related screens

- **System-Parameters Customer-Options**
  - Multimedia Call Handling (Basic)
  - Multimedia Call Handling (Enhanced)
- **Feature Related System-Parameters**
  - Default Multimedia Outgoing Trunk Parameter Selection (p.2)
- **Maintenance-Related System Parameters**
  - Packet Bus Activated = **y**
  - Minimum Maintenance Thresholds - MMIs, VCs
- **Data Module** (type = 7500 or WCBRI)
  - Multimedia (p. 1) = **y**
  - XID (p. 2) = **n**
  - MIM Support (p. 2) = **n**
- **Station**
  - MM Complex Data Ext (p. 1)
  - H.320 Conversion (p. 2)
  - Multimedia Early Answer (p. 2)
  - Multimedia Mode (p.2)
  - Service Link Mode (p.2)
  - Feature Buttons (p.3) (optional)
- **Hunt Group**
  - MM Early Answer (optional)
- **Call Vector**
  - Multimedia (optional)

- **Feature Access Codes**
  - Basic Mode Activation (p.5)
  - Enhanced Mode Activation (p.5)
  - Multimedia Call Access Code (p.5)
  - Multimedia Data Conference Activation & Deactivation (p.5)

  The Multimedia Data Conference Deactivation FAC must be entered after you are active on a multimedia call. To enter the FAC:

  1. Select **Transfer**.
  2. Receive a dialtone.
  3. Dial the FAC.
  4. Receive a confirmation tone.
  5. Re-select the call appearance for the held multimedia call.

  - Multimedia Multi-Address Access Code (p.5).
  - Multimedia Parameter Access Code (p.5).
- **DS1 Circuit Pack** (ESM Only)
  - Bit Rate=**2.048**.
  - Line Coding=**hdb3**.
  - Signaling Mode=**isdn-pri**.
  - Connect=**pbx**.
  - Interface=**network**.
  - Country Protocol=**1**.
  - CRC=**y**.
  - MMI Cabling Board.
- **Signaling group** (ESM Only)
  - Primary D-Channel.

Administration commands

System-Parameters Customer-Options

To enable the **MMCH** feature, you need to purchase a new license file.

System-Parameters Features screen

The default bandwidth for MMCH calls is defined on the **System-Parameters Features** screen.

Note:

Originating a multimedia call with the mm-call button will originate a call according to the Default Multimedia Parameters selected on the **System Parameters -Features** screen.

- This default parameter will be either 2x56 or 2x64.
- The bearer capability of the multimedia calls will either be 56K or 64K and the bandwidth will be 2B channels.

Data module screen

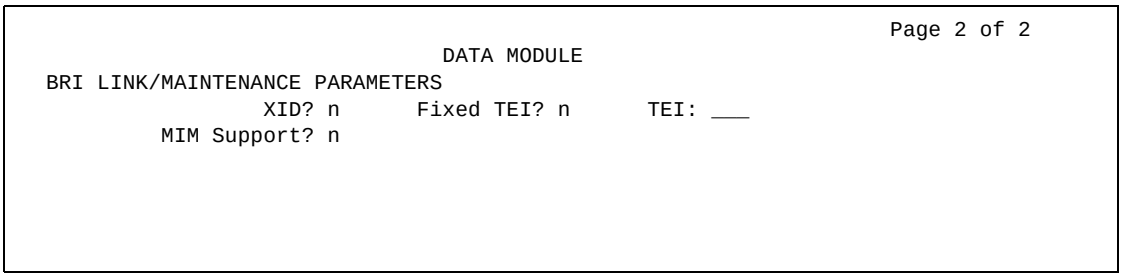
The H.320 DVC system should contain a BRI interface. You must connect this BRI interface to a port on a TN556 BRI circuit pack and administer it as a BRI data module.

- You may administer the data endpoint type as 7500 (recommended) or WCBRI.
- The fields for multimedia are the same on either screen.
- The administration for a Basic mode and an Enhanced mode data module are exactly the same.

Figure 145: Data Module screen (Page 1 of 2)

```
                                                          Page 1 of 2
                            DATA MODULE
   Data Extension: 65001     Name: _____________________________
             Type: 7500_______ COS: 1_                      Multimedia? y
             Port: _____       COR: 1_            MM Complex Voice Ext: 67434
              ITC: _____

ABBREVIATED DIALING
  List1: ____________
SPECIAL DIALING OPTION: hot-line
   HOT LINE DESTINATION
        Abbreviated Dialing Dial Code (From above list): _
CIRCUIT SWITCHED DATA ATTRIBUTES (used for modem pooling)
   Default Duplex: full     Default Mode: async     Default Speed: 1200_
DATA MODULE CAPABILITIES
  Default ITC: restricted                  Default Data Application? M2_A
```

Figure 146: Data Module screen (Page 2 of 2)



- **Type** — Set the data module type to 7500 or WCBRI.
- **Multimedia** — This field appears on the **Data Module** screen only if **MM** is set to **y** on the **System-Parameters Customer Options** screen. Enter **y** to enable this data module to be multimedia compliant.
- **MM Complex Voice Ext:** (display only) — This field contains the number of the associated telephone in the complex. This is a display-only field, and is blank until you enter the data module extension in the **Station** screen **MM Complex Data Ext** field. Once you have done that, these two extensions are associated as two parts of a multimedia complex.
- **XID and MIM Support** — Valid entries are **y** (default) and **n**. These fields must be set to **n**.

Station screen

After you have administered the BRI data module, use the **Station** screen to associate it with a voice station to screen a multimedia complex. This is a one-to-one relationship: you can administer only one station and one data endpoint per multimedia complex. Neither the voice station, nor the data endpoint can be a member of another multimedia complex.

Note:

A BRI station cannot be part of a multimedia complex.

- **H.320 Conversion** — Valid entries are **y** and **n** (default). This field is optional for non-multimedia complex voice stations and for Basic multimedia complex voice stations. It is mandatory for Enhanced multimedia complex voice stations. Because the system can only handle a limited number of conversion calls, you may need to limit the number of telephones with H.320 conversion. Enhanced multimedia complexes must have this flag set to **y**.

  For non-multimedia complex voice stations, setting this field to **y** allows H.320 calls to convert to voice and alert at the stand-alone voice station. If the call is unanswered at the voice station, the call will follow standard voice treatment. Any subsequent station that is reached in the routing of this call, that is, coverage points, forwarded destinations, call pickup members, and so forth, do not need to have the **H.320** field enabled. The **H.320** field is only needed at the *first* station that may receive the H.320 call.

For Basic multimedia complex voice stations, setting this field to **y** allows H.320 calls to convert to voice and alert at the Basic multimedia complex voice station *after* an attempt has been made to offer the call to the H.320 DVC system. If the call is unanswered at the H.320 DVC system, the call will alert at the voice station after 5 seconds or after the administered number of rings as specified in the voice station's coverage path. If the call is unanswered at the voice station, the call will follow standard voice treatment. Any subsequent station that is reached in the routing of this call, that is, coverage points, forwarded destinations, call pickup members, and so forth, do not need to have the **H.320** field enabled. The H.320 field is only needed at the *first* station that may receive the H.320 call.

- **Service Link Mode** - The service link is the combined hardware and software multimedia connection between an Enhanced mode complex's H.320 DVC system and the Avaya DEFINITY Server which terminates the H.320 protocol. A service link is never used by a Basic mode complex H.320 DVC system. Connecting a service link will take several seconds. When the service link is connected, it uses MMI, VC and system timeslot resources. When the service link is disconnected it does not tie up any resources. The Service Link Mode can be administered as either 'as-needed' or 'permanent' as described below:
  - As-Needed - Most non-call center multimedia users will be administered with this service link mode. The as-needed mode provides the Enhanced multimedia complex with a connected service link whenever a multimedia call is answered by the station and for a period of 10 seconds after the last multimedia call on the station has been disconnected. Having the service link stay connected for 10 seconds allows a user to disconnect a multimedia call and then make another multimedia call without having to wait for the service link to disconnect and re-establish.
  - Permanent - Multimedia call center agents and other users who are constantly making or receiving multimedia calls may want to be administered with this service link mode. The permanent mode service link will be connected during the station's first multimedia call and will remain in a connected state until the user disconnects from their PC's multimedia application or the Avaya DEFINITY Server restarts. This provides a multimedia user with a much quicker video cut-through when answering a multimedia call from another permanent mode station or a multimedia call that has been early answered.
- **Multimedia Mode** - There are two multimedia modes, Basic and Enhanced, as described below:
  - Basic - A Basic multimedia complex consists of a BRI-connected multimedia-equipped PC and a non-BRI-connected multifunction telephone set. When in Basic mode, users place voice calls at the multifunction telephone and multimedia calls from the multimedia equipped PC. Voice calls will be answered at the multifunction telephone and multimedia calls will alert first at the PC and if unanswered will next alert at the voice station if it is administered with H.320 = **y**. A Basic mode complex has limited multimedia feature capability as described in Basic Mode Operation on page 320).
  - Enhanced - An Enhanced multimedia complex consists of a BRI-connected multimedia-equipped PC and a non-BRI-connected multifunction telephone. The Enhanced mode station acts as though the PC were directly connected to the

multifunction telephone; the service link provides the actual connection between the Avaya DEFINITY Server and the PC. Thus, voice and multimedia calls are originated and received at the telephone set. Voice and multimedia call status are also displayed at the telephone set. An Enhanced mode station allows multimedia calls to take full advantage of most call control features as described in Enhanced Mode Operation on page 321.

- **Multimedia Early Answer** — Valid entries are **y** and **n** (default). This field lets you set this telephone for early answer of multimedia calls. The system will answer the incoming multimedia call on behalf of the station and proceed to establish the H.320 protocol. After audio path has been established to the caller, the call will then alert at the voice station. The station may then answer by going off-hook and will have immediate audio path. No hourglass tone will be heard by the answering party (see Hourglass Tone on page 339).

  Example: An administrative assistant who does not have a multimedia PC, but may get multimedia mode calls from forwarding or coverage, may want to set the H.320 flag to **y** and the early answer flag to **y** on their voice station. This allows any multimedia call to be presented to the station with immediate voice path rather than hourglass tone. The answered call could then be transferred as voice to voice mail or transferred as multimedia to a user equipped with a multimedia endpoint.

Multimedia Buttons

There are six new multimedia specific buttons that may be added to a voice station. Most of them may be placed on any voice station, whether it is part of a Basic multimedia complex, an Enhanced multimedia complex or not part of any multimedia complex. Two feature buttons, mm-basic and mm-pcaudio, may only be placed on stations which are part of an Enhanced multimedia complex.

All of the multimedia specific feature buttons have a corresponding feature access code except mm-pcaudio and mm-cfwd.

- The mm-pcaudio feature can only be used via button.
- The mm-cfwd button may be replaced by the standard "*call forward*" FAC followed by the "*multimedia call*" FAC.
- **mm-call** -This button may exist on any voice station. Most multimedia enabled users will want an mm-call button. This button (or its corresponding FAC) must be used to indicate that the user is placing a multimedia mode call. To place a multimedia mode call the user would go off-hook, select an idle call appearance, and press the mm-call button followed by the destination extension digits. If the user has a speakerphone they can simply press the mm-call button, which preselects an idle call appearance, followed by the destination extension digits.

  The mm-call button lamp lights when you press this button during call origination. The lamp also lights to indicate that the selected call appearance is a multimedia mode call.

- **mm-basic** - This button is only allowed on the voice station of a multimedia complex. The mm-basic button toggles a station between Basic and Enhanced modes. This button can NOT be used to change the station's multimedia mode when the station has an active multimedia call appearance.

  Toggling between Basic and Enhanced mode changes the station's administered Multimedia mode. When in Basic mode this field on the **Station** screen will show **basic**. When in Enhanced mode this field on the **Station** screen will show **enhanced**. The current station Multimedia mode will be saved to translation when a **`save translation`** command is executed.

- **mm-pcaudio** - This button only works for an Enhanced multimedia complex voice station. When originating or receiving a multimedia call, the audio path is connected to the voice station's handset or speakerphone device. The mm-pcaudio button allows a user to switch the audio portion of any call to their PC's audio input/output device (if available). To switch the audio path to the PC while active on a call, the user presses the mm-pcaudio button (if off-hook you may now hang up the handset). The mm-pcaudio button's status lamp will light up when the button is pushed to move the audio path to the PC and remain lit while the audio path is at the PC device.

  Note:

  If you are on a voice only call, the voice path will switch to the PC device but you will get muted or loopback video depending on the multimedia endpoint software.

  A user may simply go off-hook on their voice station or press the speakerphone button to move the audio path of a multimedia call from the PC back to the voice station. Pressing the mm-pcaudio button while the status lamp is lit and the voice station's handset is on-hook will disconnect the user from the active call.

- **mm-datacnf** - Pressing the mm-datacnf button from any voice station that is participating in a multimedia call will light the status lamp and alert the Avaya DEFINITY Server that you want to enable T.120 data collaboration with the other parties on the call. The button status lamp will also light for other participants in the multimedia call who have mm-datacnf buttons.

  Pressing this button from the voice station that enabled data collaboration on a multimedia mode call will deactivate the data session and revert to a voice and video call. If you are participating on a multimedia call with data collaboration, but did not initiate the data collaboration, and you press this button, the status lamp led will flash momentarily and the T.120 data services will not be terminated, (only the station that activated the collaboration session can deactivate it). This button only works for stations connected to an Avaya DEFINITY Server equipped with an ESM adjunct.

- **mm-cfwd** - The mm-cfwd button allows a user to indicate that multimedia mode calls will be forwarded as multimedia mode calls to a specific forwarded-to destination. If voice call forwarding is active and multimedia call forwarding is not active then multimedia calls going off of the Avaya DEFINITY Server will be forwarded as voice only calls.

  The mm-cfwd button status lamp will be lit to indicate that multimedia call forwarding is activated. Pressing the mm-cfwd button when the lamp is lit will deactivate multimedia call forwarding.

  Note - pressing the mm-cfwd button is the same as dialing the regular call-fwd FAC followed by the mm-call button or FAC followed by the desired forwarded-to extension digits.

- **mm-multinbr** - The mm-multinbr call button is similar to the mm-call button. It allows origination of a multimedia call from any voice station. It is used when the destination being dialed requires a different address for each of the 2 B-channels. An example of this is Central Office provided ISDN-BRI. This type of BRI line is provisioned with separate listed directory numbers for each B-channel. In order to make a 2B multimedia call to such a device, two sets of address must be entered.

  Originating a multimedia call with the mm-multinbr button will originate a call according to the Default Multimedia Parameters selected on the **System Parameters Features** screen. This default parameter will be either 2x56 or 2x64. The bearer capability of the multimedia calls will either be 56K or 64K and the bandwidth will be 2B channels.

ESM T.120 Server Administration

Figure 147: DS1 Circuit Pack screen

```
change ds1 1c09                                              Page   1 of   1
                              DS1 CIRCUIT PACK

            Location: 01C09                          Name:
            Bit Rate: 2.048                  Line Coding: hdb3

      Signaling Mode: isdn-pri
             Connect: pbx                      Interface: network
                                      Country Protocol: 1

                                                    CRC? y
           Idle Code: 11111111
                                DCP/Analog Bearer Capability: 3.1kHz
   MMI Cabling Board: 01C10                 MMI Interface: ESM



      Slip Detection? n                Near-end CSU Type: other
```

From the system administration terminal:

1. Type **`list configuration all`**, and a list of the installed carriers, circuit packs, and ports appears.
2. Record the location (board number) of the MMI board cabled to the TN2207 slot and verify that all other required circuit packs are present.
3. Enter **`add DS1 xxxxx`**, (where ***xxxxx*** is the location of the TN2207 PRI circuit pack recorded in step 2), and the **DS1 Circuit Pack** administration screen appears.
4. Set the **Name** field to **ESM DS1**.
5. Set the **Bit Rate** field to **2.048**.

   The TN2207 DS1 must have a bit rate of 2.048, even if all other DS1 boards in the system are operating at 1.544. Verify the 24/32 channel switch on the circuit pack is in the 32 channel position.
6. Set the **Line Coding** field to **hdb3**.
7. Set the **Signaling Mode** field to **isdn-pri**.
8. Set the **Connect** field to **pbx**.
9. Set the **Interface** field to **network**.
10. Set the **Country Protocol** `field` to **1**.
11. Set the **CRC** field to **y**.
12. The **Idle Code** default is **11111111**.
13. The **DCP/Analog Bearer Capability** default is **3.1 kHz**.
14. Set the **MMI Cabling Board** field to ***xxxxx*** (where ***xxxxx*** is the location of the TN787 MMI circuit pack recorded in step 2). This must be the slot for port B of the Y-cable.
15. The **MMI Interface** field **ESM** appears.
16. Enter **`add signaling-group next`**. The **Signaling Group** screen appears.

Figure 148: Signaling Group screen

```
change signaling-group 6                                    Page   1 of   5
                                SIGNALING GROUP

 Group Number: 6
                    Associated Signaling? y          Max number of NCA TSC: 0
                       Primary D-Channel: 01C0916     Max number of CA TSC: 0
                                                    Trunk Group for NCA TSC:
       Trunk Group for Channel Selection:
          Supplementary Service Protocol: a
```

17. Set the **Associated Signaling** field to **y**.

18. Set the **Primary D-Channel Port** field to ***xxxx*16** (where ***xxxx*** is the address of the TN2207 PRI circuit pack, for example: 1B0516).
19. The **Max Number of NCA TSC** default is **0**.
20. The **Max Number of GA TSC** default is **0**.
21. **Trunk Group for NCA TSC** ____ (leave blank).
22. **Trunk Group for Channel Selection**____ (leave blank).
23. Logoff the terminal and then log back on the terminal to view your changes.

Troubleshooting

To determine ESM link status, enter the following commands from the system administration terminal:

1. **`Status esm`**
2. **`Status signaling-group`**
3. **`List MMI`**

Note:

When you move ESM circuit packs, you MUST remove the DS1 and signaling group translations. You cannot use the **`change circuit pack`** command.

When a vector is used to route video (56K/64K) calls to a hunt group comprised of data extensions, the vector *must* have the **Multimedia** field set to **n**. This field causes multimedia calls routed through the vector to receive early answer treatment prior to processing the vector steps. This provides a talk path to the caller for announcements or immediate conversation with an agent and starts network billing for the incoming call when vector processing begins.

Understanding the Multimedia Complex

1-number access

1-number access permits originating users to make voice or multimedia calls to a Basic multimedia complex by dialing the same number for either type of call. The number may be the voice station extension or the data module extension. If the incoming call is a voice call, Avaya Communication Manager directs it to the telephone. If the incoming call is 56K or 64K data call, Avaya Communication Manager recognizes it as such and sends it to the multimedia endpoint. Likewise, if a voice call is addressed to the data extension, the system recognizes this and directs the call to the voice station.

Calls originating on the same server as the Basic mode complex destination may always use 1-number access for voice or video. In order to take advantage of 1-number access for calls originating from a remote location, the incoming calls must arrive over ISDN-PRI trunks. If the system is setup with separate data non-PRI digital facilities multimedia calls must be made to the data extension.

AVD (alternate voice/data) trunk groups cannot be used to provide 1-number access with MMCH. If the AVD trunk group has a BCC of 0, all calls arriving over the AVD trunk to the Basic mode complex will be assumed to be voice calls. If the AVD trunk group has a BCC of 1 or 4, all calls arriving over the AVD trunk to the Basic mode complex will be assumed to be multimedia calls.

Originating voice calls

All voice calls are originated at the voice station.

Originating multimedia calls

For a Basic mode complex, multimedia calls are normally originated at the user's multimedia equipped PC. These multimedia calls use the associated station's COR/COS.

The voice station of a Basic multimedia complex may also use the **mm-call** button or FAC, and the **mm-multinbr** button or FAC to originate multimedia calls. When these methods are used, a multimedia call is originated from the voice station. In order for the Basic multimedia complex to receive video, the user must make a call from the H.320 DVC system to the voice station of the complex or must make a multimedia call from the voice station to the H.320 DVC. This allows the station to spontaneously add themselves or other parties to a multimedia conference.

1. **H.320 DVC system GUI**. The normal way for a Basic multimedia complex endpoint to originate a multimedia call is through the vendor provided user interface. Generally, digits to dial are entered, speed is selected and the call originates from the DVC system. The voice station is not involved in such as origination.

   Any voice station may use the following mechanisms to originate a multimedia call from the voice station. For stations that are not part of a multimedia complex, video cannot be provided. For voice stations that are part of a Basic multimedia complex, video is not provided until a multimedia call is made from the complex's H.320 DVC system to the voice station or a multimedia call is made from the voice station to the H.320 DVC system. Video is automatically included for Enhanced multimedia complexes.

2. **mm-call (Multimedia Call) button**. If the station has an mm-call button administered, the user goes off-hook and selects the mm-call button. The user may select the mm-call button and then go off-hook. If the user has a speakerphone on the station, the user may originate the call simply by selecting the mm-call button. The speakerphone will automatically be placed off-hook and dialtone will be heard. Upon selection of the mm-call button, the mm-call status lamp (green LED) should become solid.

   The user now dials the destination address digits. The destination address may be provided by dialing digits, using abbreviated dial entries, last number dialed, station busy indicators, etc. Originating a multimedia call with the mm-call button will originate a call according to the Default Multimedia Parameters selected on the **Feature-Related System Parameters**screen. This default parameter will be either 2x56 or 2x64. The bearer capability of the multimedia calls will either be 56K or 64K and the bandwidth will be 2B channels.

For calls with a bandwidth of 2B, use of the mm-call button to originate will cause the same destination address to be used for both channels of the 2B call. The section below on the mm-multinbr button/FAC provides information on originating a 2B call where the destination has a different address for each B-channel.

Note:

The mm-call feature button is generally used by stations that are part of an Enhanced multimedia complex, but may be used by any station to originate a multimedia call.

3. **Multimedia Call feature access code**. For stations that do not have an administered mm-call button, the Multimedia call feature access code may be used instead. The user goes off-hook on the station, waits for dialtone, then dials the MM-call FAC, receives dialtone again and then dials the call normally. The destination address may be provided by dialing digits, using abbreviated dial entries, last number dialed, station busy indicators, etc.

   Originating a multimedia call with the mm-call button will originate a call according to the Default Multimedia Parameters selected on the **Feature-Related System Parameters** screen. This default parameter will be either 2x56 or 2x64. The bearer capability of the multimedia calls will either be 56K or 64K and the bandwidth will be 2B channels.

   For calls with a bandwidth of 2B, use of the mm-call button to originate will cause the same destination address to be used for both channels of the 2B call. The section below on the mm-multinbr button/FAC provides information on originating a 2B call where the destination has a different address for each B-channel.

   Note:

   The mm-call feature access code is generally used by stations that are part of an Enhanced multimedia complex, but may be used by any station to originate a multimedia call.

4. **mm-multinbr (Multimedia Multi-number) button**. The mm-multinbr button is similar to the mm-call button. It allows origination of a multimedia call from a voice station. It is used when the destination being dialed requires a different address for each of the 2 B-channels. An example of this is Central Office provided ISDN-BRI. This type of BRI line is provisioned with separate listed directory numbers for each B-channel. In order to make a 2B multimedia call to such a device, two sets of addresses must be entered.

   The user goes off-hook and selects the mm-multinbr button. The user may select the mm-multinbr button and then go off-hook. If the user has a speakerphone on the station, the user may originate the call simply by selecting the mm-multinbr button. The speakerphone will automatically be placed off-hook and dialtone will be heard. Upon selection of the mm-multinbr button, the mm-multinbr and mm-call (if present) status lamp (green led) should light steadily. The user now dials the first destination address digits. The destination address may be provided by dialing digits, using abbreviated dial entries, last number dialed, etc. The system will provide dialtone after the first address has been completed. The user now dials the second destination address digits. The destination address may be provided by dialing digits, using abbreviated dial entries, last number dialed, etc. After the 2nd address has been collected the mm-multinbr status lamp will go off.

Originating a multimedia call with the mm-multinbr button will originate a call according to the Default Multimedia Parameters selected on the **Feature-Related System Parameters** screen. This default parameter will be either 2x56 or 2x64. The bearer capability of the multimedia calls will either be 56K or 64K and the bandwidth will be 2B channels.

Note:

The mm-multinbr feature button is generally used by stations that are part of an Enhanced multimedia complex, but may be used by any station to originate a dual address multimedia call.

5. **Multimedia Multi-number Call feature access code**. For stations that do not have an administered mm-multinbr button, the Multimedia Multi-number call feature access code may be used instead. It allows origination of a multimedia call from a voice station. It is used when the destination being dialed requires a different address for each of the 2 B-channels. An example of this is Central Office provided ISDN-BRI. This type of BRI line is provisioned with separate listed directory numbers for each B-channel. In order to make a 2B multimedia call to such a device, two sets of addresses must be entered.

   The user goes off-hook and dials the MM-multinbr feature access code. Upon dialing of the MM-multinbr FAC, the mm-call (if present) status lamp (green led) should become solid. The user now dials the first destination address digits. The destination address may be provided by dialing digits, using abbreviated dial entries, last number dialed, etc. The system will provide dialtone after the first address has been completed. The user now dials the second destination address digits. The destination address may be provided by dialing digits, using abbreviated dial entries, last number dialed, etc.

   Originating a multimedia call with the MM-multinbr FAC will originate a call according to the Default Multimedia Parameters selected on the **Feature-Related System Parameters** screen. This default parameter will be either 2x56 or 2x64. The bearer capability of the multimedia calls will either be 56K or 64K and the bandwidth will be 2B channels.

   Note:

   The mm-multinbr FAC is generally used by stations that are part of an Enhanced multimedia complex, but may be used by any station to originate a dual address multimedia call.

6. **Multimedia parameter selection feature access code**. This FAC is used to originate a multimedia call that wishes to use a different bearer and bandwidth than the system default. For example, if the system has a default multimedia parameter of 2x64 and the user wishes to make a call to a destination that is known to only have 56K digital facilities, the MM parameter selection FAC can be used to select a bearer and bandwidth of 2x56 for this specific call.

   The MM parameter selection FAC may be used in conjunction with the mm-multinbr button or FAC to make a single or dual address multimedia call at the desired bearer and bandwidth. The user goes off-hook and dials the MM-parameter selection feature access code. Dialtone is returned. The user enters a single digit, 1 or 2, where 1 = 2x64, 2 = 2x56. All other digits will produce reorder. Dialtone is returned. Upon dialing of the MM-parameter selection FAC, the mm-call (if present) status lamp (green led) should become solid. The user may indicate a dual-address call at this point with the mm-multinbr button or FAC. The

user now dials one or two sets of destination address digits. The destination address may be provided by dialing digits, using abbreviated dial entries, last number dialed, etc.

Note:

The mm-parameter selection FAC is generally used by stations that are part of an Enhanced multimedia complex, but may be used by any station to originate a dual address multimedia call.

7. Dialing sequences that include TACs, AAR, ARS, Authorization codes, CDR account codes, FRLs
   1. Single address with TAC
      - **Dial** mm-call button or FAC, **Hear** dialtone
      - **Dial** TAC, **Dial** destination digits
   2. Dual address with TAC
      - **Dial** mm-multinbr button or FAC, **Hear** dialtone
      - **Dial** TAC, **Dial** 1st dest. digits, **Hear** dialtone
      - **Dial** TAC, **Dial** 2nd dest. digits
   3. Single address with AAR/ARS
      - **Dial** mm-call button or FAC, **Hear** dialtone
      - **Dial** AAR/ARS, **Dial** destination digits
   4. Dual address with AAR/ARS
      - **Dial** mm-multinbr button or FAC, **Hear** dialtone
      - **Dial** AAR/ARS, **Dial** 1st dest. digits, **Hear** dialtone
      - **Dial** AAR/ARS, **Dial** 2nd dest. digits
   5. Single address with AAR/ARS and authorization code
      - **Dial** mm-call button or FAC, **Hear** dialtone
      - **Dial** AAR/ARS FAC, **Dial** destination digits, **Hear** stutter dialtone
      - **Dial** authorization code
   6. Dual address with AAR/ARS and authorization code
      - **Dial** mm-multinbr button or FAC, **Hear** dialtone
      - **Dial** AAR/ARS FAC, **Dial** 1st dest. digits, **Hear** dialtone
      - **Dial** AAR/ARS FAC, **Dial** 2nd dest. digits, **Hear** stutter dialtone
      - **Dial** authorization code

7. Single address with TAC or AAR/ARS and CDR account code
   - **Dial** mm-call button or FAC, **Hear** dialtone
   - **Dial** CDR FAC, **Hear** dialtone
   - **Dial** CDR account code, **Hear** dialtone
   - **Dial** TAC or AAR/ARS, **Dial** destination digits
8. Dual address with TAC or AAR/ARS and CDR account code
   - **Dial** mm-multinbr button or FAC, **Hear** dialtone
   - **Dial** CDR FAC, **Hear** dialtone
   - **Dial** CDR account code, **Hear** dialtone
   - **Dial** TAC or AAR/ARS, **Dial** 1st dest. digits
   - **Dial** TAC or AAR/ARS, **Dial** 2nd dest. digits

Receiving voice calls

Any voice calls directed to the voice or data extension of a Basic multimedia complex will ring at the voice station.

Receiving multimedia calls

Any data calls directed to the voice or data extension of a Basic multimedia complex will ring at the multimedia equipped PC if it is available. You may answer the multimedia call at the PC and voice and video will connect to the PC. If the data endpoint is unavailable, the system verifies that the telephone of the complex is administered with the **H.320** field set to **y**. If so, the system converts the call to voice and sends it to the telephone of the multimedia complex, where the call then alerts.

Hourglass Tone

When a voice station answers a converted multimedia call, the answering party may hear different things depending on the nature of the originator. If the origination is directly from an H.320 DVC system or if the originator is an Enhanced mode complex on a remote server, an immediate audio path will not exist between the two parties. This is because the H.320 protocol must be established after the call is answered. It takes several seconds for the H.320 protocol to establish an audio path. During this interval the answering party will hear special ringback. When the audio path exists the special ringback will be removed and replaced with a short incoming call tone indicating that audio now exists. The combination of special ringback followed by incoming call tone is referred to as "hourglass tone." Hourglass tone is an indication to the answering party that they should wait for the H.320 call to establish audio.

Early Answer

The answering party may administer their station to avoid hearing hourglass tone. With the **Station** screen **Early Answer** field set to **y**, the system answers the incoming multimedia call on behalf of the station and establishes the H.320 protocol. After audio path has been established, the call will then alert at the voice station of the Basic complex destination. The station may then answer by going off-hook and will have immediate audio path. No hourglass tone will be heard by the answering party.

If the **H.320** field is not set to **y** for the telephone of a Basic multimedia complex, H.320 calls alert at the multimedia endpoint until the caller drops. If an H.320 call is directed to a telephone with H.320 set to **n**, the system denies the call.

You can assign H.320 conversion to any voice station.

Authorization

Multimedia complexes require the same types of authorization (COR/COS) as standard telephones. If a call is addressed to the voice extension, the system checks the COR/COS of the telephone, whether the call is voice-only or multimedia. If a call is addressed to the data extension, the system checks the COR/COS of the data endpoint. If the call is subsequently redirected to the voice station, the system does a second COR/COS check for the authorization of the voice station. Calls originated from the PC use the COR/COS of the voice station.

Adjunct Switch Applications Interface

ASAI is not expected to support call-association for data calls. Therefore Avaya does not recommend that you use ASAI for multimedia.

Administered Connections

Basic Multimedia endpoints may serve as the origination point or destination of an administered connection.

Authorization and Barrier Codes

Basic Mode multimedia users or off-premises PC users may not be able to respond to prompts for authorization or barrier codes. Multimedia endpoints do not recognize the prompts.

An on-premises user might be able to use Remote Access and enter the entire digit string at once before launching the call, but it would be better to eliminate the need for such codes for multimedia users who need to call off premises.

Bridged Appearances

Voice users can bridge onto a call if the user has a bridged appearance of a voice member of the call.

Call redirection

Calls directed to either member of the Basic multimedia complex are subject to redirection (coverage, forwarding). Avaya Communication Manager converts calls to voice before sending them to coverage. Calls redirected through call forwarding maintain multimedia status if forwarded from the data endpoint.

Conferencing

A multimedia conference can consist of multimedia and voice-only conferees. All multimedia conferees are added to a multimedia conference by a voice-terminal user on Communication Manager , who acts as the controller of the multimedia conference. When the controller is a Basic complex voice station, the controller must remain on the conference until all parties have joined. Once all endpoints are on the conference, the voice-terminal user may put the call on hold or drop, if the user wishes.

Video conferees can see only their local video and one other party. If more than two people are involved in a video conference, the person who is speaking is the one whose video appears to other conferees. The speaker's video shows the previous speaker. This changes dynamically as the speaker changes.

Creating a multi-party video conference

All multimedia conferences must be controlled by a voice telephone. Multimedia conferees may be added by calling the voice telephone or by having the voice telephone make a multimedia call to other DVC endpoints. The controller can then conference together individual parties to create a multimedia conference.

To set up a multimedia conference:

1. Determine who is going to be the conference controller.
2. At the appointed time, the conference controller calls his or her telephone from the multimedia endpoint by dialing the 1-number extension. Once this call is established, the controller conferences in other calls as if this were a voice conference. The controller continues to add conferees in this manner until all conferees have joined, or until the number of conferees reaches the administered limit.
3. The conference controller may also add voice or multimedia parties to the conference spontaneously. The controller presses **CONFERENCE**, makes a voice or multimedia call to a new party. To make a multimedia call, the controller must originate a call using the mm-call button or FAC or the mm-multinbr button or FAC. After the new party begins alerting, the controller may press **CONFERENCE** to add the party to the existing conference call on hold.

Coverage

Multimedia calls to a Basic mode complex are subject to the same coverage criteria as voice calls and follow the coverage path administered for the voice station of the Basic multimedia mode complex.

If a plain voice station or a Basic mode complex is the covering party, the answering voice station will receive audio only. If all voice stations in the coverage path have the station screen **Early Answer** field set to **n** and the originator of the multimedia call was not a local Enhanced mode complex, the answering station will hear hourglass tone.

If an Enhanced mode complex is the covering party, the answering voice station will receive voice and video. If all voice stations in the coverage path have the station screen **Early Answer** field set to **n** and the originator of the multimedia call was not a local Enhanced mode complex, the answering station will hear hourglass tone.

Coverage: Multimedia calls and off-net call coverage

If the principal station's coverage path include a remote coverage point, the multimedia call will cover off-switch as voice only. If the call is unanswered off-switch and proceeds to the next coverage point on-switch, the multimedia nature of the call is preserved.

Coverage: Multimedia calls and coverage to voice mail

Voice mail systems such as AUDIX are typically the last point in a coverage path and are usually implemented as a hunt group. In order to guarantee that the originator of an H.320 multimedia call hears the voice mail greeting, the hunt group that defines the list of voice mail ports should have the **Early Answer** field on the hunt group set to **y**. This field will have no effect on voice calls to the voice mail system.

Call Detail Recording

Each channel of a 2-channel call generates a separate CDR record.

Data Collaboration

Once you have established a multi-point video conference, multi-point T.120 data collaboration may be enabled for that call. This will allow all video parties on the current conference to collaborate.

T.120 Data conferencing is made possible through the Extended Services Module (ESM) server, which is an adjunct to Avaya Communication Manager. Up to six parties may participate in a single data conference, and up to 24 parties may use the ESM facilities for data collaboration at any given time.

Adding data sharing to a video conference

1. Set up a multimedia conference.
2. Once a multimedia call is active, any voice station in the conference, can initiate data collaboration by pressing the mm-datacnf button. Or, to use the feature access code to initiate a data conference, press the Transfer button. A second line-appearance becomes active and you hear dial tone. Dial the multimedia data conference feature access code. Confirmation tone is heard and the system automatically reselects the held call appearance of the multimedia conference. Avaya Communication Manager will select a data rate which is acceptable to all H.320 DVC systems in the current call.

   If the system does not have sufficient ESM server resources available for all parties currently in the call, the activation of T.120 data sharing will be denied. The mm-datacnf status lamp will flash denial or the mm-datacnf FAC will produce reorder.
3. Each H.320 DVC system in the conference call is joined to the data conference. On many DVC systems, the provided GUI may prompt the user with a dialog box, requesting the user to select a specific conference to join. With MMCH, there should only be one conference available to select.
4. The user must now use the PC's GUI to begin application sharing. The method for beginning application sharing or file transfer is different for each H.320 multimedia application. One of the H.320 DVC systems activates data sharing from the H.320 DVC vendor provided GUI. See your H.320 DVC system documentation for details.
5. The same H.320 DVC system as in step 4, opens an application, whiteboard, etc. to share and the image of the application is displayed on all H.320 DVC systems in the conference.

   For details on how multiple users may control the shared application, see the vendor provided documentation for your specific H.320 DVC system.
6. To end the data collaboration session and retain the voice/video conference, the station that selected the mm-datacnf button or FAC may press the mm-datacnf button or hit transfer and dial the mm-datacnf deactivation FAC.

   Note:

   As of this writing, many endpoints do not respond correctly to ending the data collaboration session and retaining voice/video. Some H.320 DVC systems drop the entire call. Avaya recommends that once T.120 data sharing has been enabled for a conference, that it remain active for the duration of the conference call. When all endpoints have dropped from the call, the T.120 resources will be released.

Joining a multimedia conference after T.120 data sharing has been enabled.

If a multimedia conference with T.120 data sharing is already active and it is desired to conference in a new video endpoint, the new video endpoint can be conferenced into the existing call. The new endpoint will be allowed into the data conference if there exists sufficient ESM server resources for the new endpoint. The new endpoint will get voice/video and data sharing if the new endpoint supports the multi-layer protocol (MLP) data rate chosen by the system when T.120 data collaboration was activated. If the endpoint does not support the pre-existing MLP data rate, the new endpoint will only receive voice and video.

Single server or switch data collaboration.

When all parties involved in data collaboration conference are located on the same physical Avaya DEFINITY Server or media server, there is no restriction on the type of user. The parties may be any combination of Enhanced multimedia complexes, Basic multimedia complexes or stand-alone H.320 DVC systems.

Multi-switch data collaboration.

When all parties involved in data collaboration conference are *not* located on the same physical Avaya media server, the parties located on the Avaya server hosting the data conference (i.e. the server which activated mm-datacnf) may be any combination of Enhanced multimedia complexes, Basic multimedia complexes or stand-alone H.320 DVC systems. *All parties on remote servers must not be Enhanced multimedia complexes:* they must be Basic multimedia complexes or stand-alone H.320 DVC systems. Prior to originating or receiving a multimedia mode call, the mm-basic feature button or feature access code can be used to dynamically change an Enhanced mode complex into a Basic mode complex and back again.

Forwarding of voice/multimedia calls

In Basic mode you can forward calls from either the telephone or the multimedia endpoint. To forward a call from the multimedia endpoint:

1. At the PC’s multimedia application, enter the call-forwarding feature access code (FAC)
2. Enter the forward-to number in the **Dialed Number** field on the endpoint software
3. Click the **Dial** button (or equivalent)

Note:

The PC multimedia software will probably respond with a message that the call failed, since it does not recognize the FAC. In fact, Avaya Communication Manager *does receive* the message, and forwards all multimedia calls addressed to the 1-number.

If a call is forwarded from the telephone, the call converts to voice first. If using the multimedia endpoint to forward, the calls arrive at the forwarded-to extension as a data call. Such calls continue to ring until answered or abandoned, rather than follow a coverage path.

Users can forward calls from the multimedia endpoint using the call forward FAC. You can also assign a call-forward button at the voice station to forward calls for the data endpoint. If a Basic multimedia complex has console permissions, that user can forward calls for others by dialing the FAC, the data extension, and then the forwarded-to number.

Call Park

A voice-terminal user can park any active call, voice or multimedia, and unpark the call from another telephone. Users cannot park or unpark calls using multimedia endpoints.

Call Pickup

Members of a pickup group can answer an H.320 call using a telephone after the call has been converted to voice. This is true for standard or directed call pickup.

Consult

After a call is converted to voice, consult may be used when transferring or conferencing the call.

COR/COS

The Class of Restriction and Class of Service for H.320 calls originated from a 1-number complex are the same as those of the telephone in the complex.

Data Call Setup

Basic complex multimedia endpoints are BRI data endpoints, and may use data call-setup procedures as provided by the software vendor.

Data Hotline

If endpoint software allows users to select the dial function without entering a number, the endpoint can be used for hotline dialing.

Dial Access to Attendant

Access to Attendant is blocked for a data call from a Basic mode multimedia endpoint.

Data Trunk Groups

Data trunk groups may be used to carry H.320 calls of a fixed (administered) bearer capability.

Hold

The voice station and multimedia endpoint of a Basic complex are each independent devices with respect to call control. When a Basic multimedia complex voice station executes hold only the voice station is held. If the user has conferenced their multimedia endpoint into a multimedia conference, activating hold will *not* disconnect the multimedia endpoint from the conference, it will only disconnect the Basic multimedia complex voice station. Executing hold with an Enhanced mode complex will fully disconnect voice and video from the current active call.

Hunt Groups using Basic Mode complexes

Since Basic mode complexes may receive point to point multimedia calls at the DVC system and voice calls to the station simultaneously, the voice station extension may be placed in any normal voice hunt group or ACD skill and the data extension may be placed in a simple hunt group made up of only data extensions.

Basic mode complex data extensions or stand-alone data extensions may be used to create simple data hunt groups. Data extensions are not allowed in ACD hunt groups. Avaya recommends that you do not mix voice and data stations in a hunt group.

If you want multimedia calls to hunt to multimedia endpoints (i.e. 2B point to point data hunting), put the data extension in the hunt group. If you place the voice extension in a hunt group, only voice calls hunt to that extension. Multimedia calls to a hunt group with a Basic mode voice station as the hunt group member will *not* be offered to the DVC system of the Basic mode complex. If either the voice or data extension of a Basic mode complex is busy, the entire complex is considered busy for hunting purposes.

In order to guarantee that all members of a voice hunt group or skill can receive voice or multimedia calls, all members should have the **H.320** field on the station screen set to **y**. Simple voice stations and Basic complex mode voice stations will receive voice only. Enhanced mode stations will receive voice and video.

The **MM Early Answer** field (on the **Hunt Group** screen) tells the system to answer the incoming multimedia call and establish audio before it reaches the first member of the hunt group. Thus, when the talk path is established, the caller is able to speak with an agent immediately. This is not necessary for hunt groups comprised of data extensions.

Hunting, Other considerations

Agents that are part of a Basic mode complex may dial a feature access code to remove themselves from availability (and to indicate that they are available again) from both the multimedia endpoint and the telephone independently. This allows the voice member or the data member to be individually made unavailable. To make the data extension unavailable, the agent must dial the FAC from the DVC system.

CMS measurements may indicate unusually slow ASA, because of the time required for the system to establish early-answer before offering the call to an agent.

Hunting Call association (routing)

Typically incoming voice calls consist of 2 B-channel calls to the same address, to provide greater bandwidth and better video resolution. Avaya Communication Manager attempts to correctly pair up incoming calls and offer them as a unit to a single agent. MMCH uses call association to route both calls to the extension that answered the first call, regardless of how the call was routed internally.

Two 56K/64K data calls with the same calling party number to the same destination number are considered to be associated. The system makes every attempt to route both calls of a 2-channel call to the same answering party. If the first call terminates at a member of a hunt group, the second call does not have to hunt, but goes directly to the same member. In order for 2B multimedia calls to be correctly given to a single agent, incoming calls to the hunt group

must have ANI information. The ANI information may be in the form of ISDN calling party number or DCS calling party number. Multimedia calls made on the same media server as the hunt group are easily associated. If multimedia calls into a hunt group have incorrect ANI information (i.e. all calls from server X to server Y include the LDN for server X), then as the volume of calls increases, the number of mis-associated calls will increase. If multimedia calls into a hunt group have no ANI information, Communication Manager will never associate pairs of calls and all calls will be treated independently and routed to separate agents. This is not a recommended configuration.

Hunting with Multimedia vectors

Calls are often routed to hunt groups or skills via a vector. The existing VDNs and vectors which exist for routing voice calls can be used to route multimedia calls.

In order to use a vector for multimedia calls that will terminate to voice stations, you must set the **Multimedia** field on the **Vector** screen to **y**. This field has no effect on voice calls routing through the vector. This field will cause multimedia calls routed through the vector to receive early answer treatment prior to processing the vector steps. This provides a talk path to the caller for announcements or immediate conversation with an agent.

Note:

Vectors which have the **Multimedia** field set to **y** must eventually route to hunt groups, skills or numbers which are voice extensions. A vector with the **Multimedia** field set to **y** should never be set up to route to a hunt group or number which is a data extension.

When a vector is used to route video (56K/64K) calls to a hunt group comprised of data extensions, the vector *must* have the **Multimedia** field set to **n**.

Intercept Treatment

H.320 calls that receive intercept treatment are treated like other data calls. H.320 calls cannot be directed to an attendant for service because the attendant cannot have H.320 conversion service.

ISDN Trunk Groups

Avaya highly recommends that you use ISDN trunks for multimedia calls. ISDN PRI trunks allow complete 1-number access for an Enhanced multimedia complex. ANI provided over PRI trunks allows correct routing of multiple bearer channels to the correct destination device. ISDN also provides the bearer capability on a call by call basis which can be used to distinguish voice calls from multimedia calls.

Malicious Call Trace

If a malicious call terminates at a Basic multimedia complex endpoint, the user can dial the feature access code from the telephone to activate malicious call trace, followed by the extension of the multimedia endpoint. If the user does not dial the multimedia extension, MCT traces any call held on the telephone.

Message Waiting

Message Waiting indication is handled at the telephone. Because H.320 calls are converted to voice before going to coverage, all messages are voice only.

Night Service

Incoming Basic mode data calls follow established night-service processing for data calls.

Remote Access

Communication Manager does not prevent Basic multimedia complexes from attempting to use remote access. However, these Basic mode endpoints will most likely not be able to dial the necessary codes.

Station Hunting

Basic mode data calls to endpoints that have an extension administered in the **Hunt-to-station** field hunt based on established hunting criteria. The call is converted to voice before station hunting.

Tenant Partitioning

Permission to make multimedia calls or add parties of any type to a conference is subject to standard tenant-partitioning restrictions.

Terminating Extension Groups

Basic mode data calls to a TEG are converted to voice and can terminate only at a voice endpoint. Effectively, Communication Manager treats the multimedia-complex extension as a voice-only endpoint.

Telephone Display

Display information for calls to or from a Basic multimedia complex contains the 1-number.

Enhanced Mode MM Complex

The Enhanced multimedia complex provides a much greater unified and integrated interface for control of voice and multimedia calls. The multifunction voice station is used to control all calls, whether voice or multimedia. The H.320 desktop video system is used to present the video stream, data stream and (optionally) audio stream to the user. The H.320 desktop video system is not used for call control. The Enhanced multimedia complex allows the multifunction voice station to handle voice or multimedia calls in an almost identical manner. Each call appearance on the voice station may represent a voice or multimedia call, allowing multiple voice or multimedia calls to be present simultaneously on the station. The user may manage the separate call appearances without regard to the voice or multimedia nature of the specific call. The standard HOLD/TRANSFER/CONFERENCE/DROP actions may be applied to any call, without regard to the voice or multimedia nature of the call.

1-number access

1-number access permits originating users to make voice or multimedia calls to an Enhanced multimedia complex by dialing the same number for either type of call. The number may be the voice station extension or the data module extension. If the incoming call is a voice call, Avaya Communication Manager alerts the station of an incoming voice call. If the incoming call is 56K or 64K data call, Communication Manager recognizes it as a multimedia call, inserts resources to terminate the H.320 protocol, and alerts the voice station with a multimedia call.

Calls originating on the same media server as the Enhanced mode complex destination may always use 1-number access for voice or video. In order to take advantage of 1-number access for calls originating from a remote location, the incoming calls must arrive over ISDN-PRI trunks. If the system is setup with separate non-PRI digital facilities for data, multimedia calls must be made to the data extension of the Enhanced mode complex.

AVD (alternate voice/data) trunk groups cannot be used to provide 1-number access with MMCH. If the AVD trunk group has a BCC of 0, all calls arriving over the AVD trunk to the Basic mode complex will be assumed to be voice calls. If the AVD trunk group has a BCC of 1 or 4, all calls arriving over the AVD trunk to the Basic mode complex will be assumed to be multimedia calls.

ORIGINATION

The basic call sequence from an Enhanced mode complex is to originate a multimedia call and alert the destination. When the destination answers the call, the originating station’s H.320 desktop video system will be alerted (that is, called by Communication Manager to establish the service link). If the H.320 DVC is not configured for auto-answer, the user must answer the H.320 calls via the DVC GUI. If the H.320 DVC is configured for auto-answer, no action is needed via the DVC GUI.

Note:

Avaya recommends, but does not require, that Enhanced mode complexes place their desktop video system into an auto-answer mode of operation.

If the far-end is providing a video signal, the 2-way video will be observed. If the destination is not providing a video signal (call was answered by a simple voice telephone), then loopback video will be provided at the Enhanced mode complex originator. The audio signal will exist at the handset of the voice telephone. The audio signal may be moved to the H.320 DVC system via activation of a mm-pcaudio button on the voice telephone.

Hourglass tone

The originating party may hear different things when the incoming multimedia call is answered depending on the nature of the answering party. If the call is being answered directly by an H.320 DVC system or if the answering party is an Enhanced mode complex on a remote server, an immediate audio path will not exist between the two parties. This is because the H.320 protocol must be established after the call is answered. It takes several seconds for the H.320 protocol to establish an audio path. During this interval the originating party will hear special ringback. When the audio path exists the special ringback will be removed and replaced with a short incoming call tone indicating that audio path now exists. The combination of special ringback followed by incoming call tone is referred to as "hourglass tone." Hourglass tone is an indication to the originating party that they should wait for the H.320 call to establish audio.

- originating voice calls

Voice calls are originated from the voice station of an Enhanced mode complex in the normal manner as for any voice station.

- originating multimedia calls

Multimedia calls from an Enhanced multimedia complex are originated from the VOICE STATION, NOT the H.320 desktop video system. All multimedia originations require the user to indicate the multimedia nature of the call prior to providing any address digits. There are several different ways to originate a multimedia call from the voice station.

1. **mm-call (Multimedia Call) button**. If the station has an mm-call button administered, the user goes off-hook and selects the mm-call button. The user may select the mm-call button and then go off-hook. If the user has a speakerphone on the station, the user may originate the call simply by selecting the mm-call button. The speakerphone will automatically be placed off-hook and dialtone will be heard. Upon selection of the mm-call button, the mm-call status lamp (green LED) will light steadily, indicating a multimedia call.

   The user now dials the destination address digits. The destination address may be provided by dialing digits, using abbreviated dial entries, last number dialed, station busy indicators, etc. Originating a multimedia call with the mm-call button will originate a call according to the Default Multimedia Parameters selected on the **System Parameters Features** screen. This default parameter will be either 2x56 or 2x64. The bearer capability of the multimedia calls will either be 56K or 64K and the bandwidth will be 2B channels.

   For calls with a bandwidth of 2B, use of the mm-call button to originate will cause the same destination address to be used for both channels of the 2B call. The section below on the mm-multinbr button/FAC provides information on originating a 2B call where the destination has a different address for each B-channel.

Note:

The mm-call feature button is generally used by stations that are part of an Enhanced multimedia complex, but may be used by any station to originate a multimedia call.

2. **Multimedia Call feature access code**. For stations that do not have an administered mm-call button, the Multimedia call feature access code may be used instead. The user goes off-hook on the station, waits for dialtone, then dials the MM-call FAC, receives dialtone again and then dials the call normally. The destination address may be provided by dialing digits, using abbreviated dial entries, last number dialed, station busy indicators, etc.

   Originating a multimedia call with the mm-call button will originate a call according to the Default Multimedia Parameters selected on the **System Parameters Features** screen. This default parameter will be either 2x56 or 2x64. The bearer capability of the multimedia calls will either be 56K or 64K and the bandwidth will be 2B channels.

   For calls with a bandwidth of 2B, use of the mm-call button to originate will cause the same destination address to be used for both channels of the 2B call. The section below on the mm-multinbr button/FAC provides information on originating a 2B call where the destination has a different address for each B-channel.

   Note:

   The mm-call feature access code is generally used by stations that are part of an Enhanced multimedia complex, but may be used by any station to originate a multimedia call.

3. **mm-multinbr (Multimedia Multi-number) button**. The mm-multinbr button is similar to the mm-call button. It allows origination of a multimedia call from a voice station. It is used when the destination being dialed requires a different address for each of the 2 B-channels. An example of this is Central Office provided ISDN-BRI. This type of BRI line is provisioned with separate listed directory numbers for each B-channel. In order to make a 2B multimedia call to such a device, two sets of addresses must be entered.

   The user goes off-hook and selects the mm-multinbr button. The user may select the mm-multinbr button and then go off-hook. If the user has a speakerphone on the station, the user may originate the call simply by selecting the mm-multinbr button. The speakerphone will automatically be placed off-hook and dialtone will be heard. Upon selection of the mm-multinbr button, the mm-multinbr and mm-call (if present) status lamp (green led) should become solid. The user now dials the first destination address digits. The destination address may be provided by dialing digits, using abbreviated dial entries, last number dialed, etc. The system will provide dialtone after the first address has been completed. The user now dials the second destination address digits. The destination address may be provided by dialing digits, using abbreviated dial entries, last number dialed, etc. After the second address has been collected, the mm-multinbr status lamp will go off.

   Originating a multimedia call with the mm-multinbr button will originate a call according to the Default Multimedia Parameters selected on the System Parameters Features screen. This default parameter will be either 2x56 or 2x64. The bearer capability of the multimedia calls will either be 56K or 64K and the bandwidth will be 2B channels.

Note:

The mm-multinbr feature button is generally used by stations that are part of an Enhanced multimedia complex, but may be used by any station to originate a dual address multimedia call.

4. **Multimedia Multi-number Call feature access code**. For stations that do not have an administered mm-multinbr button, the Multimedia Multi-number call feature access code may be used instead. It allows origination of a multimedia call from a voice station. It is used when the destination being dialed requires a different address for each of the 2 B-channels. An example of this is Central Office provided ISDN-BRI. This type of BRI line is provisioned with separate listed directory numbers for each B-channel. In order to make a 2B multimedia call to such a device, two sets of addresses must be entered.

   The user goes off-hook and dials the MM-multinbr feature access code. Upon dialing of the MM-multinbr FAC, the mm-call (if present) status lamp (green led) should become solid. The user now dials the first destination address digits. The destination address may be provided by dialing digits, using abbreviated dial entries, last number dialed, etc. The system will provide dialtone after the first address has been completed. The user now dials the second destination address digits. The destination address may be provided by dialing digits, using abbreviated dial entries, last number dialed, etc.

   Originating a multimedia call with the MM-multinbr FAC will originate a call according to the Default Multimedia Parameters selected on the System Parameters Features screen. This default parameter will be either 2x56 or 2x64. The bearer capability of the multimedia calls will either be 56K or 64K and the bandwidth will be 2B channels.

   Note:

   The mm-multinbr FAC is generally used by stations that are part of an Enhanced multimedia complex, but may be used by any station to originate a dual address multimedia call.

5. **Multimedia parameter selection feature access code**. This FAC is used to originate a multimedia call that wishes to use a different bearer and bandwidth than the system default. For example, if the system has a default multimedia parameter of 2x64 and the user wishes to make a call to a destination that is known to only have 56K digital facilities, the MM parameter selection FAC can be used to select a bearer and bandwidth of 2x56 for this specific call.

   The MM parameter selection FAC may be used in conjunction with the mm-multinbr button or FAC to make a single or dual address multimedia call at the desired bearer and bandwidth. The user goes off-hook and dials the MM-parameter selection feature access code. Dialtone is returned. The user enters a single digit, 1 or 2, where 1 = 2x64, 2 = 2x56. All other digits will produce reorder. Dialtone is returned. Upon dialing of the MM-parameter selection FAC, the mm-call (if present) status lamp (green led) should become solid. The user may indicate a dual-address call at this point with the mm-multinbr button or FAC. The user now dials one or two sets of destination address digits. The destination address may be provided by dialing digits, using abbreviated dial entries, last number dialed, etc.

Note:

The mm-parameter selection FAC is generally used by stations that are part of an Enhanced multimedia complex, but may be used by any station to originate a dual address multimedia call.

6. Dialing sequences that include TACs, AAR, ARS, Authorization codes, CDR account codes, FRLs
    1. Single address with TAC
        - **Dial** mm-call button or FAC, **Hear** dialtone
        - **Dial** TAC, **Dial** destination digits
    2. Dual address with TAC
        - **Dial** mm-multinbr button or FAC, **Hear** dialtone
        - **Dial** TAC, **Dial** 1st dest. digits, **Hear** dialtone
        - **Dial** TAC, **Dial** 2nd dest. digits
    3. Single address with AAR/ARS
        - **Dial** mm-call button or FAC, **Hear** dialtone
        - **Dial** AAR/ARS, **Dial** destination digits
    4. Dual address with AAR/ARS
        - **Dial** mm-multinbr button or FAC, **Hear** dialtone
        - **Dial** AAR/ARS, **Dial** 1st dest. digits, **Hear** dialtone
        - **Dial** AAR/ARS, **Dial** 2nd dest. digits
    5. Single address with AAR/ARS and authorization code
        - **Dial** mm-call button or FAC, **Hear** dialtone
        - **Dial** AAR/ARS FAC, **Dial** destination digits, **Hear** stutter dialtone
        - **Dial** authorization code
    6. Dual address with AAR/ARS and authorization code
        - **Dial** mm-multinbr button or FAC, **Hear** dialtone
        - **Dial** AAR/ARS, **Dial** 1st dest. digits, **Hear** dialtone
        - **Dial** AAR/ARS, **Dial** 2nd dest. digits, **Hear** stutter dialtone
        - **Dial** authorization code
    7. Single address with TAC or AAR/ARS and CDR account code
        - **Dial** mm-call button or FAC, **Hear** dialtone
        - **Dial** CDR FAC, **Hear** dialtone
        - **Dial** CDR account code, **Hear** dialtone
        - **Dial** TAC or AAR/ARS, **Dial** destination digits

8. Dual address with TAC or AAR/ARS and CDR account code
   - **Dial** mm-multinbr button or FAC, **Hear** dialtone
   - **Dial** CDR FAC, **Hear** dialtone
   - **Dial** CDR account code, **Hear** dialtone
   - **Dial** TAC or AAR/ARS, **Dial** 1st dest. digits
   - **Dial** TAC or AAR/ARS, **Dial** 2nd dest. digits

Answering

The user actions required to answer voice or multimedia calls at an Enhanced multimedia complex are identical if the H.320 DVC system is configured for auto-answer. If the H.320 DVC system is not configured for auto-answer an additional step is required. See Answering multimedia calls below.

Note:

Avaya recommends, but does not require, that Enhanced mode complexes place their desktop video system into an auto-answer mode of operation.

Answering voice calls

Incoming voice calls will alert at the voice station of the Enhanced multimedia complex in the normal manner. Standard alerting and call appearance flashing will occur. They are answered in the normal manner by selecting the alerting call appearance and going off-hook on the voice station.

Answering multimedia calls

Incoming multimedia calls will alert at the voice station of the Enhanced multimedia complex in the same manner as voice calls with one addition. If the alerting station has an administered mm-call button and the alerting call appearance is the selected call appearance (for instance, the red LED is lit, on the alerting call appearance), then the mm-call button status lamp will go on indicating that the call on the selected call appearance is a multimedia call.

The incoming multimedia call is answered in the normal manner by selecting the alerting call appearance and going off-hook on the voice station. If the H.320 DVC system for the answering party is configured for auto-answer, no other action is needed to complete the multimedia call. If the H.320 DVC system for the answering party is not configured for auto-answer, the H.320 DVC system will alert and must also be answered by the user.

Note:

Avaya recommends, but does not require, that Enhanced mode complexes place their desktop video system into an auto-answer mode of operation.

If the originating party is providing a video signal, then a complete 2-way multimedia call will exist. If the originating party is not providing a video signal, the answering party will receive loopback video. The audio signal will exist at the handset of the voice station. The audio signal may be moved to the H.320 DVC system via activation of a **mm-pcaudio** button on the voice station.

Hourglass Tone

The answering party may hear different things when the incoming multimedia call is answered depending on the nature of the originator. If the origination is directly from an H.320 DVC system or if the originator is an Enhanced mode complex on a remote server, an immediate audio path will not exist between the two parties. This is because the H.320 protocol must be established after the call is answered. It takes several seconds for the H.320 protocol to establish an audio path. During this interval the answering party will hear special ringback. When the audio path exists the special ringback will be removed and replaced with a short "incoming call tone" indicating that audio now exists. The combination of special ringback followed by incoming call tone is referred to as "hourglass tone." Hourglass tone is an indication to the answering party that they should wait for the H.320 call to establish audio.

Early Answer

The answering party may administer their station in such a way as to avoid hearing hourglass tone. If the station screen has set the **Early Answer** field to **y**, then the system will answer the incoming multimedia call on behalf of the station and proceed to establish the H.320 protocol. After audio path has been established, the call will then alert at the voice station of the Enhanced mode complex destination. The station may then answer by going off-hook and will have immediate audio path. No hourglass tone will be heard by the answering party.

- multiple call appearance operation

With an Enhanced mode complex all calls to or from the complex are controlled via the voice station. Each voice or multimedia call has its own call appearance which may be selected without regard for the nature of the call using the specific call appearance. This allows a multifunction station to control multiple voice or multimedia calls in exactly the same way they would control multiple voice calls.

As an example, a user may originate a simple voice call on the first call appearance. A multimedia call may then arrive on the second call appearance. The user activates **HOLD** on the first call appearance and selects the second call appearance to answer the multimedia call. The user may then activate **HOLD** on the second call appearance and reselect the first call appearance or select a third call appearance and originate another call.

- creating a multi-party video conference

An Enhanced multimedia complex can create a spontaneous video conference in the same way that a spontaneous voice conference is created. Given an active call, the user activates the **CONFERENCE** button. This puts the current call on **HOLD** and activates a new call appearance. The user makes a multimedia call according to the instructions for originating a multimedia call and then selects **CONFERENCE** to combine or merge the two call appearances. This results in a 3-way conference.

If all three parties are video equipped, then a 3-way video conference results. Conference members see the current speaker on video. The current speaker sees the last speaker on video. If one of the parties is not video equipped, then a 3-way audio conference exists and the two video equipped parties have 2-way video. The **CONFERENCE** action may be repeated until 6 parties have been conferenced together. The 6 parties may be any mix of voice or video, local or remote parties.

The following steps create a multi-party voice/video conference:

1. Enhanced mode complex station A originates a multimedia call to, or receives a multimedia call from, party B. Station A and party B have 2-way voice and video.
2. Station A, activates **CONFERENCE**.
3. Station A originates a multimedia call (i.e. uses the mm-call button/FAC/etc.) and dials the party to be added, Enhanced multimedia complex C.
4. Party C, answers the call from station A.
5. Station A selects **CONFERENCE** to complete the 3-way conference. Parties A,B and C will be in a 3-way voice/video conference.

   Note:

   If party C is another Enhanced mode complex on the same Communication Manager server as station A, station A does not need to indicate a multimedia call prior to dialing the new party in step 3. While A consults with C, the call will be audio only. When A completes the conference in step 5, party C's video will be activated.

A multi-party video conference uses voice-activated switching to determine which parties are seen. The current speaker is seen by all other parties. The current speaker sees the previous speaker.

Additional voice or video parties may be added by repeating these steps.

Data Collaboration

Once you have established a multi-point video conference, multi-point T.120 data collaboration may be enabled for that call. This will allow all video parties on the current conference to collaborate.

T.120 Data conferencing is made possible through the Extended Services Module (ESM) server, which is an adjunct to the Avaya DEFINITY Server. Up to six parties may participate in a single data conference, and up to 24 parties may use ESM facilities for data collaboration at any given time.

The following steps add data sharing to a video conference:

1. Set up a multimedia conference.
2. Once a multimedia call is active, any member can initiate data collaboration by pressing the mm-datacnf button. Or, to use the feature access code to initiate a data conference, press the Transfer button. A second line-appearance becomes active and you hear dial tone. Dial the multimedia data conference feature access code. Confirmation tone is heard and the system automatically reselects the held call appearance of the multimedia conference. Avaya Communication Manager will select an MLP data rate acceptable to all H.320 DVC systems in the current call.

   If the system does not have sufficient ESM server resources available for all parties currently in the call, activation of T.120 data sharing will be denied. The mm-datacnf status lamp will flash denial or the mm-datacnf FAC will produce reorder.

3. Each H.320 DVC system in the conference call is joined to the data conference. On many DVC systems, the provided GUI may prompt the user with a dialog box, requesting the user to select a specific conference to join. With MMCH, there should only be one conference available to select.
4. The user must now use the PC's GUI to begin application sharing. The method for beginning application sharing or file transfer is different for each H.320 multimedia application. One of the H.320 DVC systems activates data sharing from the H.320 DVC vendor provided GUI. See your H.320 DVC system documentation for details.
5. The same H.320 DVC system as in step 4, opens an application, whiteboard, etc. to share and the image of the application is displayed on all H.320 DVC systems in the conference.

   For details on how multiple users may control the shared application, see the vendor provided documentation for your specific H.320 DVC system.
6. To end the data collaboration session and retain the voice/video conference, the station that selected the mm-datacnf button or FAC may press the mm-datacnf button or hit transfer and dial the mm-datacnf deactivation FAC.

   Note:

   Currently, many endpoints do not respond correctly to ending the data collaboration session and retaining voice/video. Some H.320 DVC systems drop the entire call. Avaya recommends that once T.120 data sharing has been enabled for a conference, that it remain active for the duration of the conference call. When all endpoints have dropped from the call, the T.120 resources will be released.

Joining a multimedia conference after T.120 data sharing has been enabled.

If a multimedia conference with T.120 data sharing is already active and it is desired to conference in a new video endpoint, the new video endpoint can be conferenced into the existing call. The new endpoint will be allowed into the data conference if there exists sufficient ESM server resources for the new endpoint. The new endpoint will get voice/video and data sharing if the new endpoint supports the data rate chosen by the system when T.120 data collaboration was activated. If the endpoint does not support the pre-existing data rate, the new endpoint will only receive voice and video.

Activating HOLD while on a T.120 data collaboration conference.

If an Enhanced multimedia complex is active on a multimedia call and the call has activated T.120 data collaboration, the user should be receiving voice/video and data. If the station places this existing call on hold, audio and video will be disconnected for the current call. The data collaboration portion of the call will remain intact and unaffected. While this T.120 data conference is on hold, the user will only be allowed to receive audio on all other call appearances. Thus a user is limited to one call appearance that has T.120 data collaboration active.

Single server or switch data collaboration.

When all parties involved in data collaboration conference are located on the same physical Avaya DEFINITY Server or media server, there is no restriction on the type of user. The parties may be any combination of Enhanced multimedia complexes, Basic multimedia complexes or stand-alone H.320 DVC systems.

Multi-switch data collaboration.

When all parties involved in data collaboration conference are *not* located on the same physical Avaya media server, the parties located on the Avaya server hosting the data conference (i.e. the server that activated mm-datacnf) may be any combination of Enhanced multimedia complexes, Basic multimedia complexes or stand-alone H.320 DVC systems.

Note:

All parties on remote servers must not be Enhanced multimedia complexes. They must be Basic multimedia complexes or stand-alone H.320 DVC systems.

Prior to originating or receiving a multimedia mode call, the mm-basic feature button or feature access code can be used to dynamically change an Enhanced mode complex into a Basic mode complex and back again.

Voice station audio vs. H.320 DVC system audio

When an Enhanced mode complex originates or receives a voice or multimedia call, the call is originated with the station handset or answered with the station handset. The audio path will be through the handset. If the user's H.320 DVC system has speakers and a microphone, the user may wish to use the H.320 DVC system for audio in much the same manner as a built-in or separate telephone speakerphone. The user can move the station's audio to the H.320 DVC system by selecting an **mm-pcaudio** feature button on the voice station. There is no feature access code for this function.

The mm-pcaudio feature button works very much like a speakerphone on/off button. If the station is off-hook and selects mm-pcaudio, audio is directed to the PC DVC system. The switch-hook may be placed on-hook. If the handset is taken off-hook, the audio moves back to the handset. If the mm-pcaudio button is selected while audio is already on the DVC system and the handset is on-hook, this acts as a speakerphone off action and disconnects the current call.

The mm-pcaudio feature button may be used for voice as well as multimedia calls. If the mm-pcaudio feature button is selected while on a voice only call, the DVC system is alerted and brought into the call. No video will be transmitted or displayed. Audio will be directed through the PC DVC system.

Switching between Basic and Enhanced modes

There may exist occasions when an Enhanced mode complex needs to switch to Basic mode operation temporarily. One example is when a user wishes to make a direct point to point multimedia call originated directly from the H.320 DVC. Basic mode operation allows this functionality at the expense of losing multimedia call handling capabilities (i.e. hold/xfer/conf). To switch from Enhanced mode to Basic mode, the station may either select a mm-basic feature button or dial the mm-basic feature access code. Both of these actions are valid only if the Enhanced mode station has no multimedia calls active.

When in Basic mode, the status lamp for the mm-basic button, if present, will be on solid. The mm-basic feature button acts as a toggle. If the status lamp is on, when the button is selected, the lamp will go off and the station will return to Enhanced mode. The mm-enhanced feature access code will set the state of the station back to Enhanced. Switching to Enhanced mode is only valid if the associated H.320 DVC system is idle.

Note:

Toggling between Basic and Enhanced mode changes the station's administered Multimedia mode. When in Basic mode this field on the station screen will show **basic**. When in Enhanced mode this field on the station screen will show **enhanced**. The current station Multimedia mode will be saved to translation when a **`save translation`** command is executed.

Forwarding of voice and multimedia calls

The Enhanced multimedia mode complex voice station may use the existing standard call forwarding mechanisms to activate forwarding for voice calls. If the forwarding destination is on the same server, then this will also forward multimedia calls as multimedia calls to the destination. If the forwarding destination is off-switch, multimedia calls will forward off-switch as voice-only calls. This is appropriate when the user will be at a location that is not able to receive multimedia calls.

To forward multimedia calls off-switch as multimedia calls, the user must activate multimedia call forwarding. This may be done with an mm-cfwd button or feature access code. The user may also activate standard voice call forwarding and select the mm-call button prior to entering the forwarding address.

Coverage

Multimedia calls to an Enhanced mode complex are subject to the same coverage criteria as voice calls and follow the coverage path administered for the voice Station of the Enhanced multimedia mode complex.

If a plain voice station or a Basic mode complex is the covering party, the answering voice station will receive audio only. If all voice stations in the coverage path have the Station screen Early Answer field set to **n** and the originator of the multimedia call was not a local Enhanced mode complex, the answering station will hear hourglass tone.

If an Enhanced mode complex is the covering party, the answering voice station will receive voice and video. If all voice stations in the coverage path have the **Station** screen **Early Answer** field set to **n** and the originator of the multimedia call was not a local Enhanced mode complex, the answering station will hear hourglass tone.

Multimedia calls and off-net call coverage

If the principal station's coverage path include a remote coverage point, the multimedia call will cover off-switch as voice only. If the call is unanswered off-switch and proceeds to the next coverage point on-switch, the multimedia nature of the call is preserved.

Multimedia calls and coverage to voice mail.

Voice mail systems such as AUDIX are typically the last point in a coverage path and are usually implemented as a hunt group. In order to guarantee that the originator of an H.320 multimedia call hears the voice mail greeting, the hunt group that defines the list of voice mail ports should have the **Early Answer** field on the **Hunt Group** screen set to **y**. This field will have no effect on voice calls to the voice mail system.

Hunt Groups using Enhanced Mode Complexes

When creating hunt groups with Enhanced multimedia mode complexes, only the station extension should ever be entered as a hunt group member. Any hunt group or ACD skill can include the voice station of an Enhanced multimedia complex as a member. The data extension of an Enhanced mode complex should never be entered as any hunt group member. A hunt group or skill may have a mix of members that are stand-alone stations and Enhanced mode complex stations. In order to guarantee that all members of the hunt group or skill can receive voice or multimedia calls, all members should have the **H.320** field on the **Station** screen set to **y**. Simple voice stations will receive voice only. Enhanced mode stations will receive voice and video.

The **MM Early Answer** field on the **Hunt Group** screen tells the system to answer an incoming multimedia call and establish audio before it reaches the first member of the hunt group. Thus, when the talk path is established, the caller is able to speak with an agent immediately.

Other considerations

CMS measurements may indicate unusually slow ASA, because of the time required for the system to establish early-answer before offering the call to an agent.

Call association (routing)

Typically incoming voice calls consist of 2 B-channel calls to the same address, to provide greater bandwidth and better video resolution. Avaya Communication Manager attempts to correctly pair up incoming calls and offer them as a unit to a single agent. MMCH uses call association to route both calls to the extension that answered the first call, regardless of how the call was routed internally.

Two 56K/64K data calls with the same calling party number to the same destination number are considered to be associated. The system makes every attempt to route both calls of a 2-channel call to the same answering party. If the first call terminates at a member of a hunt group, the second call does not have to hunt, but goes directly to the same member.

In order for 2B multimedia calls to be correctly given to a single agent, incoming calls to the hunt group must have ANI information. The ANI information may be in the form of ISDN calling party number or DCS calling party number. Multimedia calls made on the same server as the hunt

group are easily associated. If multimedia calls into a hunt group have insufficient ANI information (i.e. all calls from server X to sever Y include the LDN for server X), then as the volume of calls increases the number of mis-associated calls will increase. If multimedia calls into a hunt group have no ANI information, Communication Manager will never associate pairs of calls and all calls will be treated independently and routed to separate agents. This is not a recommended configuration.

Multimedia vectors

Very often, calls are routed to hunt groups or skills via a vector. The existing VDNs and vectors which exist for routing voice calls can be used to route multimedia calls.

In order to use a vector for multimedia calls, you must set the **Multimedia** field on the **Vector** screen to **y**. This field has no effect on voice calls routing through the vector. This field will cause multimedia calls routed through the vector to receive early answer treatment prior to processing the vector steps. This provides a talk path to the caller for announcements or immediate conversation with an agent.

Note:

Vectors which have the **Multimedia** field set must eventually route to hunt groups, skills or numbers which are voice extensions. A vector with the **Multimedia** field set to **y** should never be set up to route to a hunt group or number which is a data extension.

Interactions

Interactions are listed here only if the operation is different from standard.

- Administered Connections

  An Enhanced multimedia complex voice station may serve as the origination point or destination of an administered connection. If the Multimedia call feature access code is included in the administration of the administered connection, this will result in a video AC.

  An Enhanced multimedia complex H.320 DVC system may not serve as the origination point of an administered connection.

- X-porting

  You cannot use X in the **Port** field when administering a data module or the data endpoint in a multimedia complex. However, you can use this to administer the telephone.

- Bridged Appearances

  Enhanced multimedia complex voice station users can bridge onto a call if the user has a bridged appearance. If the bridged appearance is for a multimedia call, selecting the bridged appearance will result in a multimedia call.

- Call Detail Recording

  Each channel of a 2-channel multimedia call generates a separate CDR record that is tagged as data.

- Call forwarding

  Users cannot forward calls from a multimedia complex using multi-number dialing, either by mm-multnmbr button or feature access code.

- Call Park

  Any station can park a multimedia call, and unpark the call from another telephone. If a multimedia call is unparked by an Enhanced mode complex station, a multimedia call will result. Users cannot park or unpark calls using multimedia endpoints.

- Call Pickup

  Any member of a pickup group can answer a multimedia call after the call has begun alerting at a station call appearance. If the station picking up the call is an Enhanced mode complex station and the call is multimedia, a multimedia call will result. This is true for standard or directed call pickup.

- Consult

  After a multimedia call has been answered, consult may be used when transferring or conferencing the call.

- COR/COS

  The Class of Restriction and Class of Service for a multimedia call originated from an Enhanced multimedia complex are those of the voice station in the complex.

- Data Call Setup

  An Enhanced mode multimedia H.320 DVC system may not originate calls from the DVC system. All calls, both voice or video are originated from the voice station.

- Data Hotline

  An Enhanced multimedia complex H.320 DVC endpoint may not be used to originate a call for hotline dialing. In order to setup a video hotline function with an Enhanced mode complex, the hotline number administered for the voice station should include the Multimedia call feature access code.

- Data Trunk Groups

  Data trunk groups may be used to carry H.320 calls of a fixed (administered) bearer capability.

- ISDN Trunk Groups

  Avaya highly recommends that you use ISDN trunks for multimedia calls. ISDN PRI trunks allow complete 1-number access for an Enhanced multimedia complex. ANI provided over PRI trunks allows correct routing of multiple bearer channels to the correct destination device. ISDN also provides the bearer capability on a call by call basis that can be used to distinguish voice calls from multimedia calls.

- Night Service

  Incoming H.320 calls follow established night-service processing for data calls.

- Remote Access

  Communication Manager does not prevent Enhanced multimedia complexes from attempting to use remote access. However, these endpoints will most likely not be able to dial the necessary codes.

- Station Hunting

  Multimedia calls to Enhanced mode complex voice stations that have an extension administered in the hunt-to-station field hunt based on established hunting criteria. If the hunt-to-station is also an Enhanced mode complex station, a multimedia call will result when the call is answered.

- Terminating Extension Groups

  A multimedia call to a TEG may be answered by any member of the TEG. If the member answering the call is an Enhanced mode complex station, a multimedia call will result.

- Telephone Display

  Display information for calls to or from an Enhanced multimedia complex contains the display information associated with the voice station.

Troubleshooting

If one channel of a 2 B-channel call goes down, your choices are to continue with reduced transmission quality, or to hang up the call and start over. It is not possible to re-establish the second channel while the call is still active.

If you cannot share data with others, it may be that both parties do not have the same endpoint software. This is true for some data collaboration, but most whiteboard and file transfer software implementations are compatible.

Monitoring MMCH

This section briefly discusses some of the commands you can use to monitor multimedia complexes and conferences. The Maintenance manual for your Avaya server may discuss some of these commands and their output in more detail.

Action	Objects	Qualifier
display	station data module	xxxxx (extension) xxxxx (extension)
list	mmi measurements multimedia	 multimedia-interface voice-conditioner esm endpoints [‘print’ or ‘schedule’] h.320-stations [‘print’ or ‘schedule’]
status	attendant conference conference conference data module station trunk esm	xxxx (console number) all xxx (conference ID) xxx (conference ID) endpoint (endpoint ID) xxxxx (extension) xxxxx (extension) (group number or group number/ member number)

Status commands

The **status** commands for data module, station, trunk, and attendant provide the conference ID and endpoint ID for any of these involved in an active multimedia conference.

Figure 149: Status Station 1002 — General Status screen

```
status station 1002

                          GENERAL STATUS

             Type: 7405D              Service State: in-service/on-hook
        Extension: 1002             Maintenance Busy? no
             Port: 01C0702             SAC Activated? no
      Call Parked? no               User Cntrl Restr: none
  Ring Cut Off Act? no             Group Cntrl Restr: none
Active Coverage Option: 1         CF Destination Ext:
                                     MM Conference ID:
                                       MM Endpoint ID:
   Message Waiting:
   Connected Ports:

           ACD STATUS                        HOSPITALITY STATUS
    Agent Logged In   Work Mode           AWU Call At:
                                             User DND: not activated
                                           Group DND: not activated
                                          Room Status: non-guest room
       On ACD Call? no
```

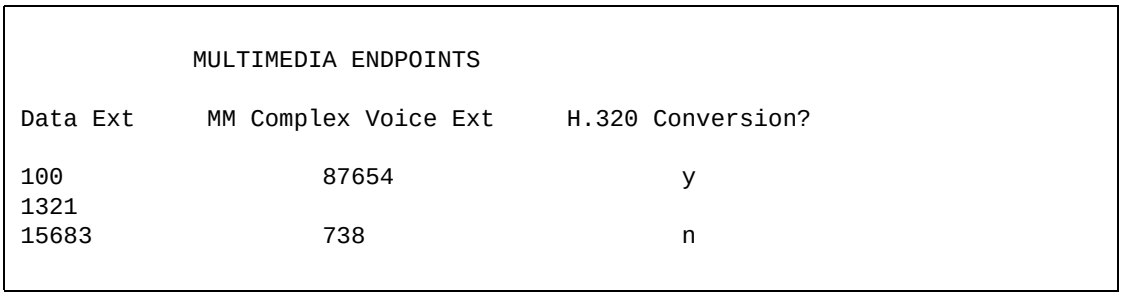
The following fields specific to multimedia appear on the status station, attendant, data module and trunk screens.

- **MM Conference ID** — This field appears only if the station is active on a multimedia conference. It displays the ID for the conference. Enter this number with the status conference command to get more information about this conference.
- **MM Endpoint ID** — This field appears only if the station is active on a multimedia conference. It displays the endpoint ID for the station. Enter this number with the status conference endpoint command to learn more about this endpoint's involvement in the conference.

List commands

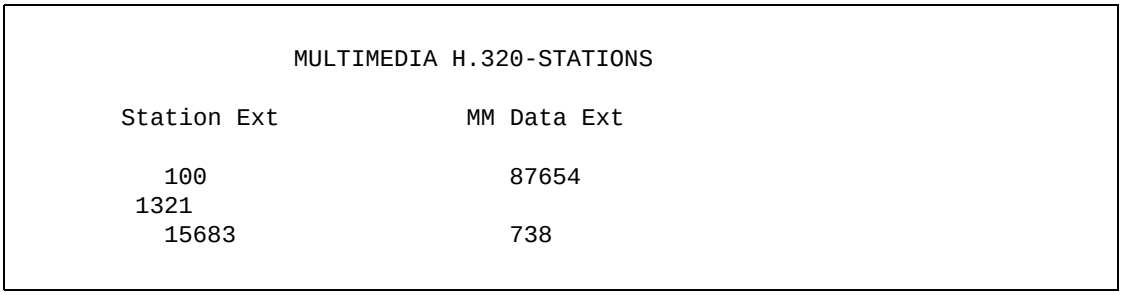
The **`list multimedia endpoints`** command shows you all the multimedia data modules that exist in your system, and their associated telephones, if any. The **`list multimedia H.320-stations`** command shows you all the stations that are administered for H.320 conversion. The **`list multimedia ip-stations`** command shows you the administered IP stations/modules and whether they are registered.

Figure 150: List Multimedia Endpoints screen



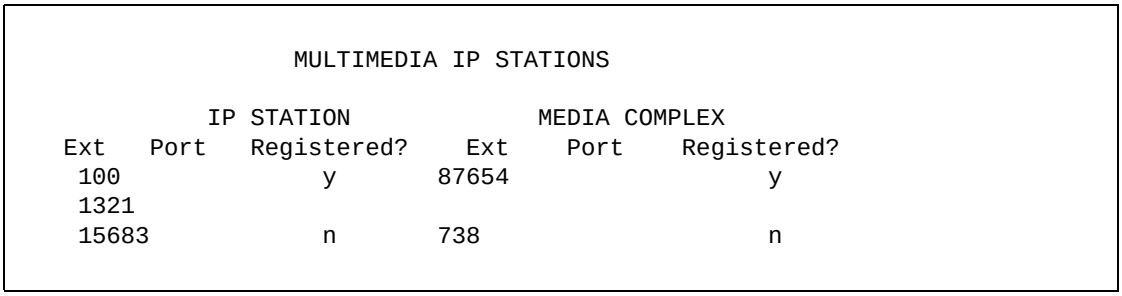
```
            MULTIMEDIA ENDPOINTS

Data Ext     MM Complex Voice Ext     H.320 Conversion?

100                  87654                    y
1321
15683                738                      n
```

Figure 151: List Multimedia H.320-Stations screen



```
                MULTIMEDIA H.320-STATIONS

      Station Ext             MM Data Ext

         100                     87654
       1321
         15683                   738
```

Figure 152: List Multimedia IP-Stations screen



```
                MULTIMEDIA IP STATIONS

           IP STATION              MEDIA COMPLEX
  Ext   Port   Registered?    Ext    Port    Registered?
   100              y       87654                  y
   1321
   15683            n       738                    n
```

Considerations

Each channel of a 2-channel BRI call takes one port on an MMI circuit pack. This alone limits the number of multimedia calls your system can handle. In addition, each conference takes one port on a voice-conditioner circuit pack. Also note that there is a limit to the total number of conversion calls the system can handle simultaneously. If you experience traffic problems after installing multimedia, you may want to reduce the number of stations that use H.320 conversion.

Chapter 10: Setting Up Telecommuting

Configuring Avaya Communication Manager for Telecommuting

Telecommuting emphasizes the ability to perform telephony activities while remote from Avaya Communication Manager. It is a combination of four features which permit you to remotely perform changes to your station's Coverage and Call Forwarding.

Note:

If you are operating in a Distributed Communications System (DCS) environment, you need to assign a different telecommuting-access extension to each media server and tell your users which extension they should use. A user can set up call coverage from any of the DCS nodes, but needs to dial the telecommuting-access extension of the node on which their station is defined before using the feature access code.

You can also set up telecommuting with an IP (internet protocol) telephone. See Adding an IP Softphone on page 98 for more information.

- Coverage of Calls Redirected Off Net (CCRON) allows you to redirect calls off your network onto the public network and bring back unanswered calls for further coverage.

  Note:

  If a call covers or forwards off-net and an answering machine answers the call, or it is directed to a cellular telephone and a cellular announcement is heard, the server views this call as an answered call. Communication Manager does not bring the call back to the server for further routing.

- The Extended User Administration of Redirected Calls feature allows you to change the direction of calls to your station. This activates the capability to have two coverage-path options. These two path options can be specified on the station screen; however, unless the **Can Change Coverage** field is set to **y** on the **Class of Restriction** screen, the second path option cannot be populated.
- The Personal Station Access feature gives you an extension number, a Merge feature access code, and a personalized security code, and tells you which office telephone you can use. This allows you to take your telephone, as long as the phones are the same type, anywhere on the same server running Communication Manager.
- The Answer Supervision feature provides supervision of a call directed out of the server either by coverage or forwarding and determines whether Communication Manager should bring the call control back to its server.

Before you start

You can also set up telecommuting with an IP (internet protocol) telephone or IP Softphone. For example, see Adding an IP Softphone on page 98 for more information.

For DCP/ISDN telecommuting, ensure you have the following equipment:

- Call Classifier — Detector
- 1264-TMx software
- Avaya Communication Manager extender — switching module or standalone rack mount (Digital Communications Protocol (DCP) or Integrated Services Digital Network (ISDN))

  For more information about this equipment, see the *Hardware Guide for Avaya Communication Manager,* 555-245-207.

Verify the following fields on the **System-Parameters Customer-Options** screen are set to **y**.

- **Cvg Of Calls Redirected Off-Net**
- **Extended Cvg/Fwd Admin**
- **Personal Station Access**
- **Terminal Translation Initialization (TTI)**

If neither **Avaya Communication Manager extender** nor the **System Parameters Customer Options** fields are configured, contact your Avaya technical support representative.

Verify the telecommuting access extension is a direct inward dialing (DID) or a central office (CO) trunk destination for off-premises features to work.

Configure **TTI** for personal station access (PSA). For information about configuring TTI, see Setting up Personal Station Access on page 369.

Configure Security Violation Notification for Station Security Codes. For information about Security Violation Notification, see Setting up Security Violations Notification on page 415.

Instructions

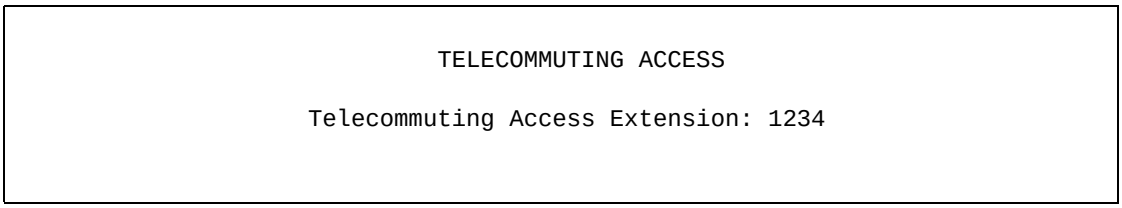
In our example, we set up the telecommuting extension and enable coverage of calls redirected off-net.

To configure Avaya Communication Manager for telecommuting:

1. Type **`change telecommuting-access`** and press **Enter**.

   The **Telecommuting Access** screen appears.

Figure 153: Telecommuting Access screen



2. In the **Telecommuting Access Extension** field, type **1234** and press **Enter**.

   This is the extension you are configuring for telecommuting.

3. Type **change system-parameters coverage** and press **Enter**.

   The **System Parameters Call Coverage/Call Forwarding** screen appears.

4. In the **Coverage Of Calls Redirected Off-Net Enabled** field, type **y** and press **Enter**.

Related topics

See Telecommuting Access on page 1503 for information about and field descriptions on the Telecommuting Access screen.

Setting up Personal Station Access

Personal Station Access (PSA) allows you to associate the preferences and permissions assigned to your own extension with any other compatible telephone. When you request a PSA associate, the system automatically dissociates another extension from the telephone.

Preferences and permissions include the definition of terminal buttons, abbreviated dial lists, and class of service (COS) and class of restriction (COR) permissions assigned to your station. Extensions without a COS, such as Expert Agent Selection (EAS) agents or hunt groups, cannot use PSA.

PSA requires you to enter a security code and can be used on-site or off-site. Invalid attempts to associate a telephone generate referral calls and are recorded by Security Violation Notification, if that feature is enabled. If you interrupt the PSA dialing sequence by pressing the release button or by hanging up, the system does not log the action as an invalid attempt.

The disassociate function within PSA allows you to restrict the features available to a telephone. When a telephone has been dissociated using PSA, it can be used only to call an attendant, or to accept a TTI or PSA request. You can enable a dissociated set to make other calls by assigning a special class of restriction.

When a call that goes to coverage from a PSA-disassociated extension, Avaya Communication Manager sends a message to the coverage point indicating that the call was not answered. If the coverage point is a display telephone, the display shows "da" for "don't answer." If the coverage point is a voice-messaging system, the messaging system receives an indication from Communication Manager that this call was not answered, and treats the call accordingly.

Note:

Once a telephone has been associated with an extension, anyone using the terminal has the capabilities of the associated station. Be sure to execute a dissociate request if the terminal can be accessed by unauthorized users. This is particularly important if you use PSA and DCP extenders to permit remote DCP access.

Before you start

Verify that the **Personal Station Access** field is set to **y** on the **Class of Service** screen.

Verify that the extension has a COS that allows PSA.

Instructions

In our example, we specify the TTI State, the Record PSA/TTI Transactions, the class of service, and the feature access codes set up for PSA.

To set up Personal Station Access:

1. Type **change system-parameters features** and press **Enter**.

   The **Feature-Related System Parameters** screen appears.

2. Complete the following fields and press **Enter**.

   a. Type **voice** in the **TTI State** field.

   b. (Optional) Type **y** in the **Record CTA/PSA/TTI Transactions in History Log** field.

   These fields display only when the **Terminal Translation Initialization (TTI) Enabled** field on this screen is set to **y**.

3. Type **change cos** and press **Enter**.

   The **Class of Service** screen appears.

4. Type **y** in the **Personal Station Access (PSA) 1** field and press **Enter**.

5. Type **change feature-access-codes** and press **Enter**.

   The **Feature Access Code** screen appears.

6. Complete the following fields and press **Enter**.
   a. Type **#4** in the **Personal Station Access (PSA) Associate Code** field.

      This is the feature access code you will use to activate Personal Station Access at a telephone.

   b. Type **#3** in the **Dissociate Code** field.

      This is the feature access code you will use to deactivate Personal Station Access at a telephone.

More information

You can allow users to place emergency and other calls from phones that have been dissociated. To enable this, you must first assign a class of restriction (COR) for PSA-dissociated phones. You do this on the **Feature-Related System Parameters** screen. In addition, you must set the restrictions for this COR on the **Class of Restriction** screen.

If you want users to be able to place emergency calls from dissociated phones, it is also a good idea to have the system send calling party number (CPN) or automatic number identification (ANI) information for these calls. To do this, you must set the CPN, ANI for PSA Dissociated Sets field to y on the **Feature-Related System Parameters** screen.

Related topics

See Training Users on page 385 for information on how to associate or disassociate PSA.

Creating a Station Security Code

A Station Security Code (SSC) provides security to a station user by preventing other users from accessing functions associated with the user's station. Each station user can change their own SSC if they know the station's current settings.

You must create a system-wide SSC change feature access code (FAC) before users can change their SSC. You must also provide users with their individual SSC. A user cannot change a blank SSC.

Instructions

In our example, we set the station security code for a user.

To create a station security code:

1. Type **change feature-access-codes** and press **Enter**.

   The **Feature Access Code** screen appears.

2. Type **#5** in the **Station Security Code Change Access Code** field and press **Enter**.

   This sets the access codes for this features. The Command prompt appears.

3. Type **change system-parameters security** and press **Enter**.

   The **Security-Related System Parameters** screen appears.

4. Type **4** in the **Minimum Station Security Code Length** field and press **Enter**.

   This determines the minimum required length of the Station Security Codes you enter on the **Station** screen. Longer codes are more secure. If station security codes are used for external access to telecommuting features, the minimum length should be 7 or 8.

5. Type **change station 1234** and press **Enter**.

   This is the station extension you configured for telecommuting. The **Station** screen appears.

Figure 154: Station screen

```
change station 1014                                          Page 1 of X
                              STATION

Extension: 1014               Lock Messages? n                    BCC: 0
     Type:                    Security Code:4321                  TN: 1
     Port:                  Coverage Path 1:                      COR: 1
     Name:                  Coverage Path 2:                      COS: 1
                            Hunt-to Station:

STATION OPTIONS
              Loss Group: 2               Personalized Ringing Pattern: 3
            Data Module? n                          Message Lamp Ext: 1014
           Speakerphone: 2-way                   Mute button enabled? y
        Display Language? English                   Expansion Module?
                  Model:
                                                    Media Complex Ext:
                                                         IP Softphone? y
                                                  Remote Office Phone? y
```

6. Type **4321** in the **Security Code** field and press **Enter**.

Related topics

See Station on page 1409 for information about and field descriptions on the **Station** screen.

See "Station Security Codes" in *Feature Description and Implementation for Avaya Communication Manager (*555-245-205*)* for a description of the Station Security Codes feature.

Assigning an Extender Password

Avaya Communication Manager allows you assign an extender password to a user. You can assign one password for each Avaya Communication Manager port.

Before you start

Use the Remote Extender PC in the server room to perform this procedure.

Instructions

In this example, we will set a system-generated random password for a user named John Doe.

To assign an extender password:

1. Double-click the **Security** icon.

   The **Password Manager** screen appears.

2. Double-click **User Password for User 01**.

3. Select **Enable Password** to enable the password.

4. Click **random**.

   This means that the password is a system generated random number. The system displays a 10-digit number in the **Password** field. Take note of this number, your user will need it at home to access the server running Communication Manager.

5. Type **Doe, John** and click **OK**.

   This is the last name and first name of the user. The system returns you to the **Password Manager** screen.

6. Select **CommLink:Select Cards**.

   A screen containing a list of cards (for example, Card A, Card B, and so on) appears. Each card corresponds to a port on your media server.

7. Select **Card A** and click **OK**.

   The system returns you to the **Password Manager** screen.

8. Select **CommLink:Upload Password**.

   The error message screen appears with the message **Administrator password not loaded"**.

9. Click **OK**.

   The **Administrator** screen appears.

10. Type **123456** and click **OK**.

    This is the administrator's password.

11. Select **CommLink:Upload Password**.

    The password is uploaded.

12. When upload is complete, click **OK**.

    The system returns you to the **Password Manager** screen.

13. Select **File:Save As**.

    The **Save As** screen appears.

14. Type **doe.fil** in the **File** field and click **OK**.

The system saves the User01 information.

Setting up Call Forwarding

Avaya Communication Manager allows you to change your call forwarding from any on-site or off-site location.

Instructions

In our example, we assign the feature access codes and class of service to set up call forwarding. This allows your users to forward their calls to another extension.

To set up call forwarding:

1. Type **change feature-access-codes** and press **Enter**.

   The **Feature Access Code** screen appears.

2. Set a 2-digit access code for the following fields and press **Enter**.

   a. Type ***8** in the **Extended Call Fwd Activate Busy D/A** field.

   b. Type ***7** in the **Extended Call Fwd Activate All** field.

   c. Type ***6** in the **Extended Call Fwd Activate Deactivation** field.

      This sets the access codes for these features. The Command prompt appears.

3. Type **change cos** and press **Enter**.

   The **Class of Service** screen appears.

4. Set the following fields to **y**.

- **Extended Forwarding All**
  - **Extended Forwarding B/DA**

  This allows you to change the forwarding of all your calls from an off-site location.

5. Set the **Restrict Call Fwd-Off Net** field to **n** and press **Enter**.

   This allows your users to forward calls off-site.

Interactions

- Bridged Appearance

  When the pound key (#) is pressed from a bridged appearance immediately following any of this feature's four FACs, the system assumes that the currently active bridged extension will be administered. The station security code of the currently active bridged extension must be entered after the initial # to successfully complete the command sequence.

  If the station has only bridged appearances, the station's extension must be dialed after the FAC to successfully complete the command sequence, since the station's extension is not associated with any appearances.

- Distributed Communications System

  Assign a different telecommuting access extension for each server running Communication Manager. You can use Extended User Administration of Redirected Calls from any of the DCS nodes, but you must dial the extension of the node on which your station is defined before dialing the FAC.

- Tenant Partitioning

  The telecommuting access extension is always automatically assigned to Tenant Partition 1, so it can be accessed by all tenants.

  The tenant number of the extension being administered must be accessible by the tenant number from which the Extended User Administration of Redirected Calls FAC is dialed or the request is denied. If the FAC is dialed on site, the tenant number of the station or attendant must have access to the tenant number of the extension administered. If the FAC is dialed off site, the tenant number of the incoming trunk must have access to the tenant number of the extension administered.

Related topics

See Training Users on page 385 for information on how to change call forwarding.

See *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205*,* for a description of the Call Forwarding feature.

See *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205, for a description of the Tenant Partitioning feature.

Assigning Coverage Options

Avaya Communication Manager allows you to assign two previously administered coverage paths and/or time of day coverage tables on the **Station** screen. This allow telecommuters to alternate between the two coverage paths and/or time of day coverage tables administered to better control how their telephone calls are handled.

For information about creating a coverage path, see Creating coverage paths on page 192. For information about creating a time of day coverage table, see Assigning a coverage path to users on page 194.

Instructions

In our example, we assign two coverage options so a user can choose from either option to control how their calls are handled.

To assign 2 coverage options:

1. Type **change feature-access-codes** and press **Enter**.

   The **Feature Access Code** screen appears.

2. Type **#9** in the **Change Coverage Access Code** field and press **Enter**.

3. Type **change cor 1** and press **Enter**.

   The **Class of Restriction** screen appears.

4. In the **Can Change Coverage** field, type **y** and press **Enter** to save your work.

   The Command prompt appears.

5. Type **change station 1234** and press **Enter**.

   This is the station extension you configured for telecommuting. The **Station** screen appears.

6. Complete the following fields:

   a. Type **2** in the **Coverage Path 1** field.

   b. Type **8** in the **Coverage Path 2** field.

Related topics

See Coverage Path on page 859 for information about and field descriptions on the **Coverage Path** screen.

See *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205*,* for a description of the Call Coverage feature.

See Training Users on page 385 for information on how to alternate your coverage path option.

See *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205, for information about the Extended User Administration of Redirected Calls feature.

Installing Home Equipment

Avaya Communication Manager allows you to install equipment in your home so that you can utilize system facilities from off-site.

Before you start

You can also set up telecommuting with an IP (internet protocol) telephone or IP Softphone. For example, see Adding an IP Softphone on page 98 for more information.

For DCP telecommuting, you need the following equipment:

- Avaya Communication Manager extender remote module
- DCP sets (office and home must match)

Configure a feature access code for associating your home number to your office number. For information about configuring an associate feature access code, see Setting up Personal Station Access on page 369.

Instructions

Installing home equipment

To install your home equipment:

1. Plug the telephone cord into the slot labeled line on the back of the module and into the wall jack.

2. Plug the telephone cord into the slot labeled port on the back of the module and into the slot labeled line on the telephone.
3. Plug the power cord into slot labeled power on the back of the module and the wall socket.

   The telephone display **Go Online** appears.
4. Press **3 (Nxt)**.

   The telephone display **Set Phone Number** appears.
5. Press **2 (OK)** to set the telephone number.
6. Type **5551234** and press **Drop.**

   This is the assigned analog telephone number. In some areas, you may need to include your area code (for example, 3035551234). The telephone display **Set Phone Number** appears.
7. Press **1(Prv)**.

   This returns you to the **Go Online** telephone display.
8. Press **2 (OK)**.

   The module dials the number. When the modules connect, the telephone display **Enter Password** appears.
9. Type **0123456789** and press **Drop**.

Associating your office telephone number to the home station

To associate your telephone number:

1. On your home station, type **#4**.

   This is the associate feature access code.
2. Type **4321** and press #.

   This is your extension number.
3. Type **1996** and press #.

   This is your password.

Disassociating your home station

To disassociate your home station:

1. Press **Hold** four times.

Related topics

See Configuring Avaya Communication Manager for Telecommuting on page 367 for step-by-step instructions on how to configure your office equipment.

See Training Users on page 385 for step-by-step instructions on how to use your home station.

Setting up Remote Access

Remote Access permits a caller located outside the system to access the server running Avaya Communication Manager through the public or private network and then use the features and services of the system.

Remote Access users can dial into the system using central office (CO), Foreign Exchange (FX), Wide Area Telecommunications trunks (WATS), and Integrated Services Digital Network Primary Rate Interface (ISDN-PRI) trunks. In addition, a dedicated Remote Access Direct Inward Dialing number can be provided.

SECURITY ALERT:

Avaya has designed the Remote Access feature incorporated in this product that, when properly administered by the customer, will enable the customer to minimize the ability of unauthorized persons to gain access to the network. It is the customer's responsibility to take the appropriate steps to properly implement the features, evaluate and administer the various restriction levels, protect access codes and distribute them only to individuals who have been advised of the sensitive nature of the access information. Each authorized user should be instructed concerning the proper use and handling of access codes.

In rare instances, unauthorized individuals make connections to the telecommunications network through use of remote access features. In such an event, applicable tariffs require that the customer pay all network charges for traffic. Avaya cannot be responsible for such charges, and will not make any allowance or give any credit for charges that result from unauthorized access.

If you do not intend to use Remote Access now or in the future, you can permanently disable the feature. If you do decide to permanently disable the feature, it will require Avaya Services intervention to activate the feature again.

Before you start

Configure the **Incoming Destination** and **Night Service** fields on the **CO Trunk** screen. For information about configuring a CO trunk, see Adding a CO, FX, or WATS Trunk Group on page 426.

Verify that the **Authorization Codes** field on the **System Parameters Customer Options** screen is set to **y**.

Verify that the **SVN Authorization Code Violation Notification Enabled** field on the **Security-Related System Parameters** screen is set to **y**.

Instructions

Setting up remote access

In our example, we set up a remote access extension with maximum security. This assists you in blocking unauthorized people from gaining access to your network.

To set up remote access:

1. Type **change remote-access** and press **Enter**.

   The **Remote Access** screen appears.

Figure 155: Remote Access screen

```
                                REMOTE ACCESS
   Remote Access Extension 1234___          Barrier Code Length 7__
Authorization Code Required? y                   Remote Access Dial Tone: y

Barrier Code   COR  TN  COS    Expiration Date   No. of Calls    Calls Used
1:1234567      1__  1_  1__    01/01/99         ______          ______
2:________     1__  1_  1__    __/__/__         ______          ______
3:________     1__  1_  1__    __/__/__         ______          ______
4:________     1__  1_  1__    __/__/__         ______          ______
5:________     1__  1_  1__    __/__/__         ______          ______
6:________     1__  1_  1__    __/__/__         ______          ______
7:________     1__  1_  1__    __/__/__         ______          ______
8:________     1__  1_  1__    __/__/__         ______          ______
9:________     1__  1_  1__    __/__/__         ______          ______
10:_______     1__  1_  1__    __/__/__         ______          ______

Permanently Disable? __ Disable Following A Security Violation? y
(NOTE:  You must logoff to effect permanent disabling of Remote Access)
```

2. Type **1234** in the **Remote Access Extension** field.

   This is the extension specified in the **Incoming Destination** field on the **CO Trunk** screen.

3. Type **7** in the **Barrier Code Length** field.

   This is the number of digits your barrier code must be when entered.

4. Type **y** in the **Authorization Code Required** field.

   This means you must also enter an authorization code when you access the system's Remote Access facilities. For information about setting up access codes, see Setting up Authorization Codes on page 412.

5. Type **y** in the **Remote Access Dial Tone** field.

   This means you hear dial tone as a prompt to enter your authorization code.

6. Type **1234567** in the **Barrier Code** field.

   This is the 7-digit barrier code you must enter to access the system's Remote Access facilities.

7. Type **1** in the **COR** field.

   This is the class of restriction (COR) number associated with the barrier code that defines the call restriction features.

8. Type **1** in the **TN** field.

   This is the Tenant Partition (TN) number.

9. Type **1** in the **COS** field.

   This is the class of service (COS) number associated with the barrier code that defines access permissions for Call Processing features.

10. Type **01/01/04** in the **Expiration Date** field.

    This is the date the barrier code expires. A warning message is displayed on the system copyright screen seven days before the expiration date. The system administrator can modify the expiration date to extend the time interval, if necessary.

11. Type **y** in the **Disable Following A Security Violation** field.

    This disables the remote access feature following detection of a remote access security violation.

12. Press **Enter** to save your work.

Disabling remote access permanently

To disable remote access permanently:

1. Type **change remote-access** and press **Enter**.

   The **Remote Access** screen appears.

2. Type **y** in the **Permanently Disable** field.

   If you permanently disable this feature, it requires Avaya Services intervention to reactivate the feature. There is a charge for reactivation of this feature.

3. Press **Enter** to save your work.

CAUTION:

Your attempt to disable the Remote Access feature will be lost if te server running Communication Manager is rebooted without saving translations. Therefore, execute a **save translation** command after permanently disabling the Remote Access feature.

Using Secure Shell for remote login

You can log in remotely to the following platforms using Secure Shell (SSH) as a secure protocol:

- G350 Media Gateway
- S8300, S8500, S8700, or S8710 Media Server Linux command line
- Communication Manager System Administration Terminal (SAT) interface on a Media Server using port 5022.

The SSH capability provides a highly secure method for remote access. The capability also allows a system administrator to disable Telnet when it is not needed, making for a more secure system. For details on disabling Telnet, see Turning off Telnet for increased security.

Note:

The client device for remote login must also be enabled and configured for SSH.

Refer to your client P.C. documentation for instructions on the proper commands for SSH.

More information

Remote Access provides you with access to the system and its features from the public network. This allows you to make business calls from home or use Recorded Telephone Dictation Access to dictate a letter. If authorized, you can also access system features from any on-site extension.

With Remote Access you can dial into the system using Direct Inward Dialing (DID), Central Office (CO), Foreign Exchange (FX), or 800 Service trunks. When a call comes in on a trunk group dedicated to Remote Access, the system routes the call to the Remote Access extension you have assigned. If DID is provided and the Remote Access extension is within the range of numbers that can be accessed by DID, Remote Access is accessed through DID.

Barrier codes provide your system security and define calling privileges through the administered COR. You can administer up to 10 barrier codes, each with a different COR and COS. Barrier codes can be from 4 to 7 digits, *but all codes must be the same length*. You can also require that users enter an authorization code to use this feature. Both barrier codes and authorization codes are described under Setting up Authorization Codes on page 412.

Related topics

See QSIG to DCS TSC Gateway on page 1353 for information about and field descriptions on the Remote Access screen.

See *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205, for a description of the Remote Access feature.

Training Users

Avaya Communication Manager allows you to associate and disassociate PSA, change the coverage path for your station, change the extension to which you forward your calls, and change your personal station's security code.

Before you start

Configure PSA. For information about configuring PSA, see Setting up Personal Station Access on page 369.

Assign two coverage options for your system. For information on how to assign coverage options, see Assigning Coverage Options on page 377.

Configure call forwarding for your system. For information about configuring call forwarding, see Setting up Call Forwarding on page 375.

Configure security codes for a station. For information about configuring personal station security codes, see Assigning an Extender Password on page 374.

Instructions

Associating PSA

In this example, we associate PSA (preferences and permissions) assigned to your station with another compatible terminal.

To associate PSA:

1. Dial **#4**.

   This is the associate PSA feature access code. You hear dial tone.

2. Type **1234** and press #.

   This is your extension.

3. Type **4321** and press #.

   This is your Station Security Code. You hear a confirmation tone.

Disassociating PSA

In our example, we disassociate PSA from the station you are using.

To disassociate PSA:

1. Dial **#3**.

   This is the disassociate PSA feature access code. You are no longer PSA associated to this station.

Changing a coverage option

In this example, we change the coverage option from path 1 to path 2 from a remote location.

To change a coverage option:

1. Dial **1234**.

   This is the extension you configured for telecommuting. You hear dial tone.

2. Dial **#9** and press #.

   This is the feature access code you set for changing a coverage path. You hear dial tone.

3. Dial **4321** and press #.

   This is the extension for which you want to change the coverage path.

4. Dial **87654321** and press #.

   This is the extension security code.

5. Dial **2**.

   This is the new coverage path. You hear confirmation tone.

Changing call forwarding

In this example, we change call forwarding to extension 1235.

To change call forwarding:

1. Dial **1234**.

   This is the extension you configured for telecommuting.

2. Dial **#8** and press #.

   This is the feature access code you set for activating extended call forward. You hear dial tone.

3. Dial **4321** and press #.

   This is the extension from which you want to forward calls.

4. Dial **87654321** and press #.

   This is the extension security code. You hear dial tone.

5. Dial **1235**.

   This is the extension to which you want to forward calls. You hear the confirmation tone.

Changing your personal station security codes

In this example, we change the security code for extension 1235 from 98765432 to 12345678.

To change your security code:

1. Dial **#5**.

   This is the feature access code you set for changing your security code. You hear dial tone.

2. Dial **1235** and press #.

   This is the extension for which you want to change the security code.

3. Dial **98765432** and press #.

   This is the current security code for the extension. You hear dial tone.

4. Dial **12345678** and press #.

   This is the new security code. Security codes can be 3-8 digits long.

5. Dial **12345678** and press #.

   This is to confirm your new security code. You hear the confirmation tone.

   Note:

   If you cannot change your security code, Manager 1 can clear the problem using the **`Clear Audit Summary`** command.

Interrupting the command sequence for personal station security codes

To interrupt the command sequence for personal station security codes:

1. To interrupt the command sequence before step 3, choose one of these options:

   - Hang up or press the disconnect or recall button before hearing intercept tone in step 3.

     The system does not log an invalid attempt. You must restart the process at step 1.

   - Type ***** before the second # in step 3.

     You must begin the change sequence at the point of entering your extension in step 2. (You should not enter the FAC again.)

   - Type ***** after the FAC has been entered and before the final #.

     You must restart the process at step1.

2. To interrupt the command sequence after step 3, type ***** in steps 4 or 5, you must begin the change sequence at the point of entering the new station security code (SSC) in step 4.

   If you hear intercept tone in any step, the command sequence has been invalidated for some reason and you must restart the process at step 1.

   If you hear intercept tone after step 3, the system logs an invalid attempt via the Security Violations Notification (SVN) feature. This is true even if you attempt to interrupt the change sequence with an asterisk.

Related topics

See *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205, for a description of the Security Violations Notification (SVN) feature.

Chapter 11: Enhancing System Security

Basic Security

Keeping your system secure

The following is a partial list you can use to help secure your system. It is not intended as a comprehensive security checklist. See the *Avaya Toll Fraud and Security Handbook* for more information about these and other security-related features.

- Secure the system administration and maintenance ports and/or logins on Avaya Communication Manager using the Access Security Gateway. This optional password authentication interface program is provided to customers with maintenance contracts.
- Activate Security Violations Notification to report unsuccessful attempts to access the system. Security Violations Notification lets you automatically disable a valid login ID following a security violation involving that login ID and disable remote access following a security violation involving a barrier code or authorization code.
- Use the **`list history`** command to determine if unauthorized changes have been made to the system. To assist in identifying unauthorized use of the system, the History report lists each time a user logs on or off the system. See the *Reports for Avaya Communication Manager* for more information about this report.
- Secure trunks using Automatic Route Selection (ARS), Class of Restriction (COR), Facility Restriction Levels (FRLs) and Alternate Facility Restriction Levels (AFRLs), Authorization Codes, Automatic Circuit Assurance (ACA), and Forced Entry of Account Codes (see "Call Detail Recording" in *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205 for more information).
- You can log in remotely using Secure Shell (SSH) as a secure protocol. The SSH capability provides a highly secure method for remote access. The capability also allows a system administrator to disable Telnet when it is not needed, making for a more secure system.

Activate Enhanced Call Transfer for your voice messaging system, if available. This limits transfers to valid extensions, but you also need to restrict transfers to extensions that may offer dial tone to the caller, such as screen extensions.

Preventing Toll Fraud

Top 15 tips to help prevent toll fraud

Toll Fraud Tips:

1. Protect system administration access

   Make sure secure passwords exist for all logins that allow System Administration or Maintenance access to the system. Change the passwords frequently.

   Set logoff notification and forced password aging when administering logins. You must assign passwords for these logins at setup time.

   Establish well-controlled procedures for resetting passwords.

2. Prevent voice mail system transfer to dial tone

   Activate "secure transfer" features in voice mail systems.

   Place appropriate restrictions on voice mail access/egress ports.

   Limit the number of invalid attempts to access a voice mail to five or less.

3. Deny unauthorized users direct inward system access (screen)

   If you are not using the Remote Access features, deactivate or disable them.

   If you are using Remote Access, require the use of barrier codes and/or authorization codes set for maximum length. Change the codes frequently.

   It is your responsibility to keep your own records regarding who is allowed to use which authorization code.

4. Place protection on systems that prompt callers to input digits

   Prevent callers from dialing unintended digit combinations at prompts.

   Restrict auto attendants and call vectors from allowing access to dial tone.

5. Use system software to intelligently control call routing

   Create Automatic Route Selection or World Class Routing patterns to control how each call is to be handled.

   Use "Time of Day" routing capabilities to limit facilities available on nights and weekends.

   Deny all end-points the ability to directly access outgoing trunks.

6. Block access to international calling capability

   When international access is required, establish permission groups.

   Limit access to only the specific destinations required for business.

7. Protect access to information stored as voice

   Password restrict access to voice mail mailboxes.

   Use non-trivial passwords and change passwords regularly.

8. Provide physical security for telecommunications assets

   Restrict unauthorized access to equipment rooms and wire connection closets.

   Protect system documentation and reports data from being compromised.

9. Monitor traffic and system activity for abnormal patterns

   Activate features that "turn off" access in response to unauthorized access attempts.

   Use Traffic and Call Detail reports to monitor call activity levels.

10. Educate system users to recognize toll fraud activity and react appropriately

    From safely using calling cards to securing voice mailbox password, train your users on how to protect themselves from inadvertent compromises to the system's security.

11. Monitor access to the dial-up maintenance port. Change the access password regularly and issue it only to authorized personnel. Consider activating Access Security Gateway. See "Access Security Gateway" in *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205*,* for more information.

12. Create a system-management policy concerning employee turnover and include these actions:

    a. Delete any unused voice mailboxes in the voice mail system.

    b. Immediately delete any voice mailboxes belonging to a terminated employee.

    c. Immediately remove the authorization code if a terminated employee had screen calling privileges and a personal authorization code.

    d. Immediately change barrier codes and/or authorization codes shared by a terminated employee. Notify the remaining users of the change.

    e. Remove a terminated employee's login ID if they had access to the system administration interface. Change any associated passwords immediately.

13. Back up system files regularly to ensure a timely recovery. Schedule regular, off-site backups.

14. Callers misrepresenting themselves as the "telephone company," "AT&T," "RBOCS," or even known employees within your company may claim to be testing the lines and ask to be transferred to "900," "90," or ask the attendant to do "start 9 release." This transfer reaches an outside operator, allowing the unauthorized caller to place a long distance or international call. Instruct your users to never transfer these calls. Do not assume that if "trunk to trunk transfer" is blocked this cannot happen.

Hackers run random generator PC programs to detect dial tone. Then they revisit those lines to break barrier codes and/or authorization codes to make fraudulent calls or resell their services. They do this using your telephone lines to incur the cost of the call. Frequently these call/sell operations are conducted at public payphones located in subways, shopping malls, or airport locations. See QSIG to DCS TSC Gateway on page 1353 to prevent this happening to your company.

Physical Security

Physical security is your responsibility. Implement the following safeguards as an added layer of security:

1. Unplug and secure attendant console handsets when the attendant position is not in use.
2. Lock wiring closets and media server rooms.
3. Keep a log book register of technicians and visitors.
4. Shred all Communication Manager information or directories you discard.
5. Always demand verification of a technician or visitor by asking for a valid I.D. badge.
6. Keep any reports that may reveal trunk access codes, screen barrier codes, authorization codes, or password information secure.
7. Keep the attendant console and supporting documentation in an office that is secured with a changeable combination lock. Provide the combination only to those individuals who need to enter the office.
8. Keep any documentation pertaining to Communication Manager operation secure.
9. Label all backup tapes or flash cards with correct dates to avoid using an outdated one when restoring data. Be sure that all backup media have the correct generic software load.

System Security Checklist

Here's some of the steps required for indemnification. Use these to analyze your system security.

1. Remove all default factory logins of **cust**, **rcust**, **browse**, **nms**, and **bcms** and assign unique logins with 7-character alphanumeric passwords and a 90-day password aging. Use the **`list logins`** command to find out what logins are there.
2. If you do not use Remote Access, be sure to disable it permanently.

   Tip:

   You can use the **`display remote-access`** command to check the status of your remote access.

   To disable Remote Access, on the **Remote Access** screen, in the **Permanently Disable** field, type **y**.

   Note:

   Avaya recommends that you permanently disable Remote Access using the **`change remote-access`** command. If you do permanently disable Remote Access, the code is removed from the software. Avaya charges a fee to restore the Remote Access feature.
3. If you use Remote Access, but only for internal calls, change announcements or remote service observing.
   a. Use a 7-digit barrier code.
   b. Assign a unique COR to the 7-digit barrier code.

      The unique COR must be administered where the **FRL** is **0**, the **Calling Party Restriction** field is **outward**, the **Calling Permissions** field is **n** on all unique Trunk Group COR.
   c. Assign **Security Violation Notification Remote** to **10** attempts in **2** minutes.
   d. Set the aging cycle to **90** days with **100** call limit per barrier code.
4. If you use Remote Access to process calls off-net or in any way access the public network:
   a. Use a 7-digit barrier code.
   b. Assign a unique COR to the barrier code.
   c. Restrict the COR assigned to each barrier code by FRL level to only the required calling areas to conduct business.
   d. Set the aging cycle to **90** days with **100** call limit per barrier code.
   e. Suppress dial tone where applicable.
   f. Administer Authorization Codes.

g. Use a minimum of 11 digits (combination of barrier codes and authorization codes).

h. Assign **Security Violation Notification Remote** to 10 attempts in 2 minutes.

5. If you use vectors:

a. Assign all Vector Directory Numbers (VDN) a unique COR. See the *Avaya Communication Manager Contact Center Guide to ACD Contact Centers* and the *Avaya Communication Manager Contact Center Call Vectoring and Expert Agent Selection (EAS) Guide* for more information.

Note:

The COR associated with the VDN dictates the calling privileges of the VDN/vector. High susceptibility to toll fraud exists on vectors that have "collect digits" steps. When a vector collects digits, it processes those digits back to Communication Manager and if the COR of the VDN allows it to complete the call off-net, it will do so. For example, the announcement "If you know your party's 4-digit extension number, enter it now" results in 4 digits being collected in step 6. If you input "90##" or "900#", the 4 digits are analyzed and if "9" points towards ARS and "0" or "00" is assigned in the ARS Analysis Tables and the VDN COR allows it, the call routes out of the server to an outside local exchange or long distance operator. The operator then connects the call to the requested number.

b. If vectors associated with the VDN do not require routing the call off-net or via AAR, assign a unique COR where the **FRL** is **0**, the **Calling Party Restriction** field is **outward**, the **Calling Permissions** field is **n** on all unique Trunk Group COR.

c. If the vector has a "route-to" step that routes the call to a remote server via AAR, assign a unique COR with a unique ARS/AAR Partition Group, the lowest FRL to complete an AAR call, and **n** on all unique COR assigned to your public network trunking facilities on the Calling Permissions. Assign the appropriate AAR route patterns on the AAR Partition Group using the **`change aar analysis partition x 2`** command.

Tip:

You can use the **`display aar analysis print`** command to print a copy of your Automatic Alternate Routing (AAR) setup before making any changes. You can use the printout to correct any mistakes.

d. If the vector has a "route-to" step that routes the call to off-net, assign a unique COR with a unique ARS/AAR Partition Group, the lowest FRL to complete an ARS call, and **n** on all unique COR assigned to your public network trunking facilities on the Calling Permissions. Assign the appropriate complete dial string in the "route-to" step of the vector the unique ARS Partition Group using the **`change ars analysis partition x 2`** command.

6. On the **Feature Access Code** screen, **Facility Test Calls Access Code**, the **Data Origination Access Code**, and the **Data Privacy Access Code** fields, change from the default or remove them.

   Note:

   These codes, when dialed, return system dial tone or direct access to outgoing trunking facilities. Transfers to these codes can take place via an unsecured vector with "collect digits" steps or an unsecured voice mail system.

7. Restrict Call Forwarding Off Net on every class of service.

   See Class of Service on page 830 for more information on Class of Service.

   Note:

   You cannot administer loop-start trunks if Call Forwarding Off Net is required.

8. If loop start trunks are administered on Communication Manager and cannot be changed by the Local Exchange Company, block all class of service from forwarding calls off-net. In the **Class of Service** screen, **Restriction Call Fwd-Off Net** field, set to **y** for the 16 (0-15) COS numbers.

   See Class of Service on page 830 for more information.

   Note:

   If a station is call forwarded off-net and an incoming call to the extension establishes using a loop-start trunk, incorrect disconnect supervision can occur at the Local Exchange Central Office when the call terminates. This gives the caller recall or transfer dial tone to establish a fraudulent call.

9. Administer Call Detail Recording on all trunk groups to record both incoming and outgoing calls.

   See Collecting Information About Calls on page 591 for more information.

10. On the **Route Pattern** screen, be careful assigning route patterns with an **FRL** of **0**; these allow access to outgoing trunking facilities. Avaya recommends assigning routes with an FRL of 1 or higher.

    Note:

    An exception might be assigning a route pattern with an FRL of 0 to be used for 911 calls so even restricted users may dial this in emergencies.

    Tip:

    You can use the **`list route-pattern print`** command to print a copy of your FRLs and check their status.

11. On all trunk group screens, set the **Dial Access** field to **n**. If set to **y**, it allows users to dial Trunk Access Codes, thus bypassing all the ARS call screening functions.

    See the Trunk Group section of the Screen Reference for more information.

12. On the AAR and ARS Digit Analysis Table on page 709, set all dial strings not required to conduct business to **den** (deny).

13. If you require international calling, on the AAR and ARS Digit Conversion Table on page 715, use only the 011+ country codes/city codes or specific dial strings.
14. Assign all trunk groups or same trunk group types a unique Class of Restriction. If the trunk group does not require networking through Communication Manager, administer the Class of Restriction of the trunk group where the **FRL** is **0**, the **Calling Party Restriction** field is **outward**, and all unique Class of Restriction assigned to your outgoing trunk groups are **n**. See Class of Restriction on page 813 for more information.

    Tip:

    You can use the **`list trunk-group print`** command to have a printout of all your trunks groups. Then, you can use the **`display trunk-group x`** command (where ***x*** is the trunk group) to check the COR of each trunk group.

15. For your AUDIX, on the **System Appearance** screen, set:
    - the **Enhanced Call Transfer** field to **y**.
    - the **Transfer Type** field to **enhanced**. If set to **basic**, set the **Transfer Restriction** field to **subscribers**. See Feature-Related System Parameters on page 975 for more information.

    Note:

    The COR of the voice mail ports dictates the calling restrictions of the voice mail. If the above settings are not administered correctly, the possibility exists to complete a transfer to trunk access codes or ARS/AAR feature codes for fraudulent purposes. Never assign mailboxes that begin with the digits or trunk access codes of ARS/AAR feature access codes. Require your users to use a mailbox password length greater than the amount of digits in the extension number.

16. Avaya recommends you administer the following on all voice mail ports:
    - Assign all voice mail ports a unique COR. See Class of Restriction on page 813 for more information.
    - If you are not using outcalling, fax attendant, or networking, administer the unique COR where the **FRL** is **0**, the **Calling Party Restriction** field is **outward**, and all unique trunk group COR on the Calling Permissions are **n**. See Class of Restriction on page 813 for more information.

    Note:

    Avaya recommends you administer as many layers of security as possible. You can implement Step 9 and Step 16 as a double layer of security. In the event that the voice mail system becomes unsecured or compromised for any reason, the layer of security on Avaya Communication Manager takes over, and vice versa.

17. Administer all fax machines, modems, and answering machines analog voice ports as follows:
    - Set the **Switchhook Flash** field to **n**.
    - Set the **Distinctive Audible Alert** field to **n**. See Station on page 1409 for more information.
18. Install a Call Accounting System to maintain call records. In the **CDR System Parameters** screen, **Record Outgoing Calls Only** field, set to **y**. See CDR System Parameters on page 797 for more information.
19. Call Accounting Systems produce reports of call records. It detects phones that are being hacked by recording the extension number, date and time of the call, and what digits were dialed.

Adding Logins and Passwords

This section shows you how to add a user and their password. To add a login, you must be a superuser with authority to administer permissions.

When adding logins, remember the following:

- Type the new login name as part of the **add** command. The name must be 3 to 6 alphanumeric characters in length, and can contain the characters 0-9, a-z, A-Z.
- The password must be from 7 to 11 alphanumeric characters in length and contain at least 1 non-alphabetic character.

Instructions

We will add the login **angi3** with the password **b3stm0m**. We also will require the user to change their password every 30 days.

To add new logins and passwords:

1. Type **add login angi3** and press **Enter**.

   The **Login Administration** screen appears.

Figure 156: Login Administration screen

```
                                LOGIN ADMINISTRATION

              Password of Login Making Change:

          LOGIN BEING ADMINISTERED
                                Login's Name: angi3
                                  Login Type:
                               Service Level:
    Disable Following a Security Violation?
                                                        Access to INADS Port? _

          LOGIN'S PASSWORD INFORMATION
                           Login's Password:
                   Reenter Login's Password:
        Password Aging Cycle Length (Days): 30

        LOGOFF NOTIFICATION
          Facility Test Call Notification? y     Acknowledgment Required? y
               Remote Access Notification? y     Acknowledgment Required? y

ACCESS SECURITY GATEWAY PARAMETERS
Access Security Gateway? n
```

The **Login's Name** field shows the name you typed in the **add** command.

2. In the **Password of Login Making Change** field, type your superuser password.
3. In the **Disable Following a Security Violation** field, type **y** to disable this login following a login security violation.

   This field appears only if on the **Security-Related System Parameters** screen, **SVN Login Violation Notification** field is **y**.
4. In the **Login's Password** field, type **b3stm0m**.

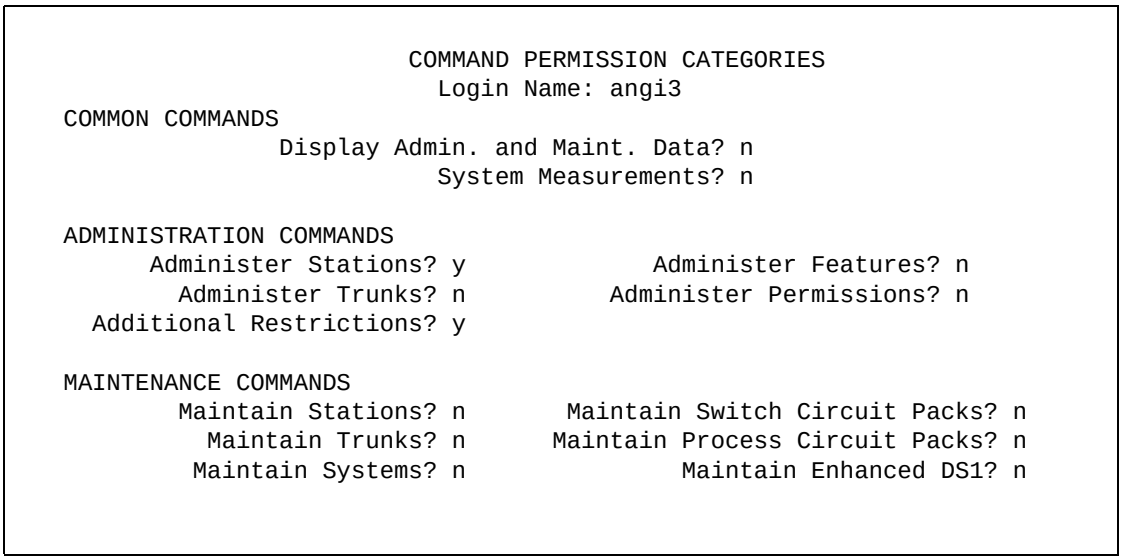
   The password does not appear on the screen as you type.
5. In the **Reenter Login's Password** field, retype **b3stm0m**.
6. In the **Password Aging Cycle Length (Days)** field, type **30**.

   This requires the user to change the password every 30 days.
7. Press **Enter** to save your changes.

   Now you need to set the permissions for this new login.
8. Type **change permissions angi3** and press **Enter**.

   The **Command Permission Categories** screen appears.

Figure 157: Command Permission Categories screen



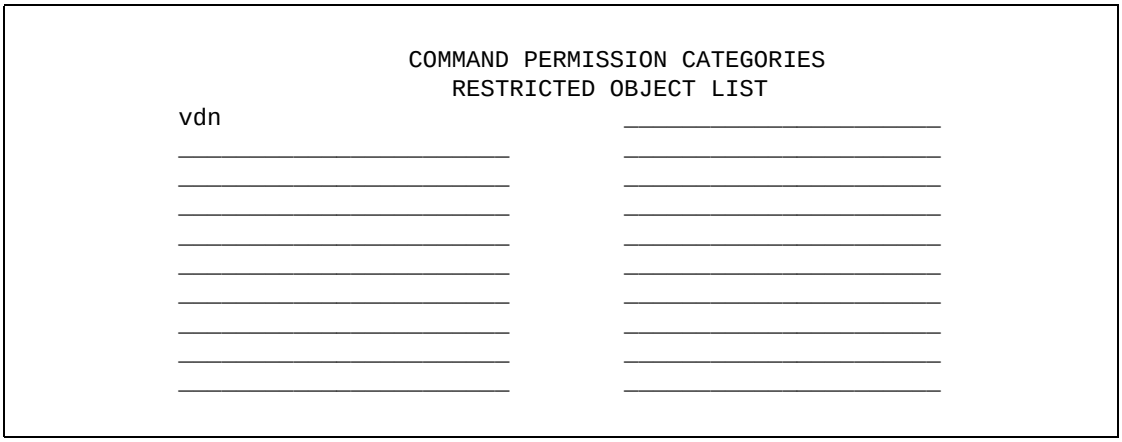
9. In the **Administer Stations** field, type **y**.

   This allows your user to add, change, duplicate, or remove stations, data modules and associated features.

10. In the **Additional Restrictions** field, type **y**.

    A **y** in this field brings up the second and third pages of this screen.

Figure 158: Command Permission Categories screen- Restricted Objects



11. In the first field, type **vdn**.

    This restricts your user from administering a VDN.

12. Press **Enter** to save your changes.

More information

When you add a login, the Security Measurement reports do not update until the next hour.

Password aging is an option you can start while administering logins. The password for each login can be aged starting with the date the password was created or changed and continuing for a specified number of days (1 to 99).

The system notifies the user at the login prompt, 7 days before the password expiration date, their password is about to expire. When the password expires, the user needs to enter a new password into the system before logging in.

Changing a Login

This section shows you how to change a user's login. You may need to change a user's password because it has expired. To change a login's attributes, you must be a superuser with authority to administer permissions.

When changing logins, remember the following:

- Type the new login name as part of the change command. The name must be 3 to 6 alphanumeric characters in length, and can contain the characters 0-9, a-z, A-Z.
- The password must be from 7 to 11 alphanumeric characters in length and contain at least 1 non-alphabetic character.

Instructions

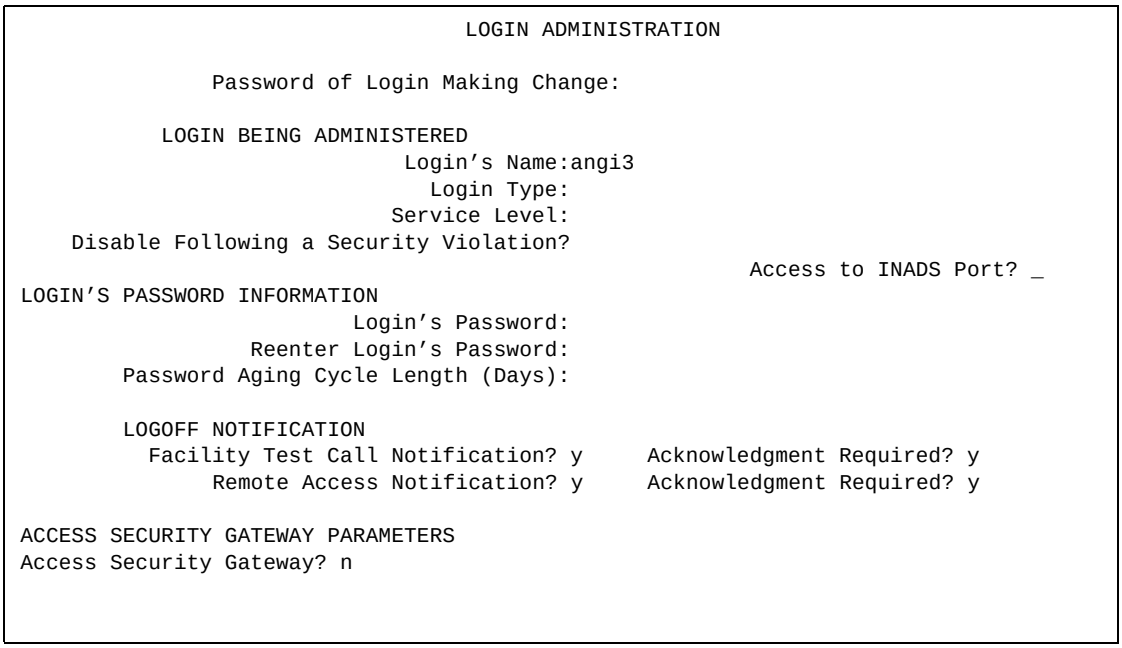
We will change the login **angi3** with the password **b3stm0m**. We also will require the user to change their password every 30 days.

To change logins:

1. Type **`change login angi3`** and press **Enter**.

   The **Login Administration** screen appears.

Figure 159: Login Administration screen



2. In the **Password of Login Making Change** field, type your superuser password.
3. In the **Login's Password** field, type **b3stm0m**.

   This is the login for the password you are changing.
4. In the **Reenter Login's Password** field, retype **b3stm0m**.

   The password does not appear on the screen as you type.
5. In the **Password Aging Cycle Length (Days)** field, type **30**.

   This requires the user to change the password every 30 days.
6. Press **Enter** to save your changes.

Related topics

For more information on logging in, see Logging into the System on page 21.

Displaying a Login

This section shows you how to display a user's login and review their permissions.

Instructions

To display a login such as **angi3**:

1. Type **`display login angi3`**, and press **Enter**.

   The **Login Administration** screen appears and displays all information about the requested login except the password.

Removing a Login

This section shows you how to remove a user's login. To remove a login, you must be a superuser.

Instructions

To remove a login such as **angi3**:

1. Type **`remove login angi3`** and press **Enter**.

   The **Login Administration** screen appears showing information for the login you want to delete.

2. Press **Enter** to remove the login, or press **Cancel** to leave this screen without removing the login.

More information

When you remove a login, the Security Measurement reports do not update until the next hour.

Related topics

For more information on logging in, see Logging into the System on page 21.

Using Access Security Gateway

This section shows you how to use Access Security Gateway (ASG). ASG prevents unauthorized access by requiring the use of the hand-held Access Security Gateway Key for logging into the system.

You need superuser privileges to perform any of the ASG procedures.

Before you start

You need an Access Security Gateway Key.

On the **System-Parameters Customer-Options** screen, verify the **Access Security Gateway** field is **y**. If not, contact your Avaya representative.

Instructions

To set up access security gateway:

1. Type **`change login`** ***`xxxx`***, where ***`xxxx`*** is the alphanumeric login ID.

   The **Login Administration** screen appears.

Figure 160: Login Administration screen

```
                                LOGIN ADMINISTRATION

               Password of Login Making Change:

          LOGIN BEING ADMINISTERED
                                Login's Name:xxxxxxx
                                  Login Type:
                               Service Level:
    Disable Following a Security Violation?
                                                          Access to INADS Port? _

          LOGIN'S PASSWORD INFORMATION
                           Login's Password:
                  Reenter Login's Password:
        Password Aging Cycle Length (Days):

        LOGOFF NOTIFICATION
          Facility Test Call Notification? y     Acknowledgment Required? y
               Remote Access Notification? y     Acknowledgment Required? y

ACCESS SECURITY GATEWAY PARAMETERS
Access Security Gateway? n
```

2. In the **Password of Login Making Change** field, type your password.
3. In the **Access Security Gateway** field, type **y**.

   When set to **y**, the **Access Security Gateway Login Administration** screen (page 2) appears automatically.
4. Either:
   - Set the **System Generated Secret Key** field to:
     - **y** for a system-generated secret key, or
     - **n** for a secret key to be entered by the administrator, *or*
   - In the **Secret Key** field, enter your secret key.

     Be sure to remember your secret key number.
5. All other fields on page 2 are optional.
6. Press **Enter** to save your changes.
7. Type **`change system-parameters security`** and press **Enter**.

   The **Security-Related System Parameters** screen appears.

Figure 161: Security-Related System Parameters screen

```
                                                              Page 2 of 2
             SECURITY-RELATED SYSTEM PARAMETERS

  SECURITY VIOLATION NOTIFICATION PARAMETERS

    SVN Station Security Code Violation Notification Enabled? y
           Originating Extension: ________   Referral Destination: ________  Station
Security Code Threshold: 10                Time Interval: 0:03
          Announcement Extension: ________

  STATION SECURITY CODE VERIFICATION PARAMETERS

                      Minimum Station Security Code Length: 4
    Security Code for Terminal Self Administration Required? y

  ACCESS SECURITY GATEWAY PARAMETERS

    SYSAM-LCL? n      SYSAM-RMT? y
        MAINT? n       SYS-PORT? n
```

8. In the **ACCESS SECURITY GATEWAY PARAMETERS** section, you determine which of the following necessary port type fields to set to **y**.

   Note:

   Avaya recommends that you protect the SYSAM-RMT port since it is a dial-up port and therefore is more susceptible to compromise.

   In this example, in the **SYSAM-RMT** field, type **y**.

9. Press **Enter** to save your changes.

Disabling Access Security Gateway

To temporarily disable ASG while users are on vacation or travel:

1. Type **`change login xxxx`**, where ***xxxx*** is the alphanumeric login ID.

   The **Login Administration** screen appears.

Figure 162: Login Administration

```
                    LOGIN ADMINISTRATION

              Password of Login Making Change:

          LOGIN BEING ADMINISTERED
                              Login's Name:xxxxxxx
                                Login Type:
                              Service Level:
    Disable Following a Security Violation?
                                                      Access to INADS Port? _

          LOGIN'S PASSWORD INFORMATION
                          Login's Password:
                  Reenter Login's Password:
        Password Aging Cycle Length (Days):

        LOGOFF NOTIFICATION
          Facility Test Call Notification? y     Acknowledgment Required? y
               Remote Access Notification? y     Acknowledgment Required? y

ACCESS SECURITY GATEWAY PARAMETERS
Access Security Gateway? n
```

2. On the **Access Security Gateway Login Administration** page (page 2), set the **Blocked** field to **y**.

   Setting the **Blocked** field to **y** does not remove the login from the system, but temporarily disables the login.

3. Press **Enter** to save your changes.

   Note:

   A superuser can disable and restart access for another superuser.

Restarting Access Security Gateway

To restart temporarily disabled access security gateway access for login:

1. Type **change login *xxxx*** and press **Enter,** where ***xxxx*** is the alphanumeric login ID.

   The **Login Administration** screen appears.

2. On the **Access Security Gateway Login Administration** page (page 2), set the **Blocked** field to **n**.

3. Press **Enter** to save your changes.

Loss of an ASG key

If a user loses their Access Security Gateway Key:

1. Modify any logins associated with the lost Access Security Gateway Key. See the *Access Security Gateway (ASG) Key Release 1.0 User's Guide* for more information on changing your PIN.

2. If the login is no longer valid, type **remove login *xxxx*** and press **Enter,** to remove the invalid login from the system, where xxxx is the alphanumeric login ID.

3. To keep the same login, change the Secret Key associated with the login to a new value.

4. Using the new secret key value, re-key devices that generate responses and interact with the login.

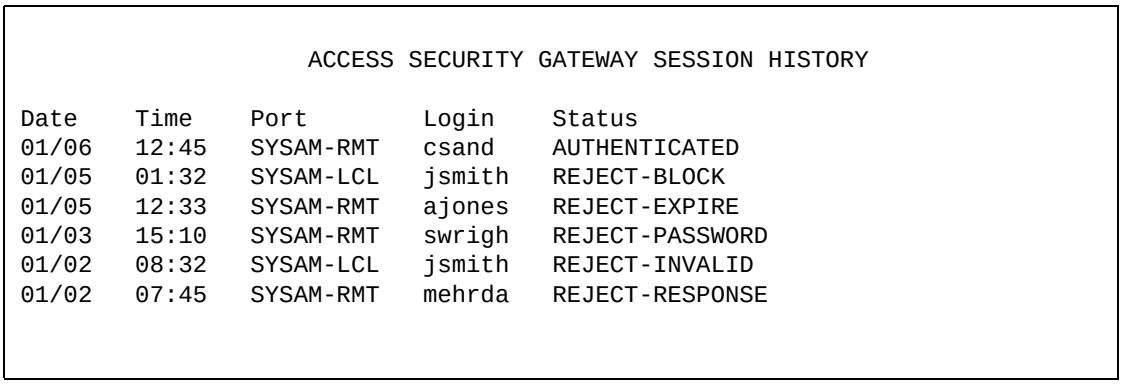
Monitoring the Access Security Gateway History Log

The Access Security Gateway Session History Log records all ASG session establishment and session rejection events except when, on the **Login Administration** screen, the **Access to INADS Port** field is **y**. You must be a superuser to use the **list asg-history** command.

1. Type **list asg-history** and press **Enter**.

   The **Access Security Gateway** screen appears.

Figure 163: Access Security Gateway



```
                    ACCESS SECURITY GATEWAY SESSION HISTORY

Date    Time    Port        Login    Status
01/06   12:45   SYSAM-RMT   csand    AUTHENTICATED
01/05   01:32   SYSAM-LCL   jsmith   REJECT-BLOCK
01/05   12:33   SYSAM-RMT   ajones   REJECT-EXPIRE
01/03   15:10   SYSAM-RMT   swrigh   REJECT-PASSWORD
01/02   08:32   SYSAM-LCL   jsmith   REJECT-INVALID
01/02   07:45   SYSAM-RMT   mehrda   REJECT-RESPONSE
```

This screen contains the following fields:

- **Date** — Contains the date of the session establishment or rejection. For example, the date displays in the mm/dd format where mm = month and dd = day.
- **Time** — Contains the time of the session establishment or rejection. For example, the time displays in the hh/mm format where hh = hour and mm = minute.
- **Port** — Contains the port mnemonic associated with the port on which the session was established or rejected. The port mnemonics for G3r systems are SYSAM-LCL, SYSAM-RMT, MAINT, and SYS-PORT. For G3si systems, they are MRG1, INADS, NET, and EPN.
- **Login** — Contains the alphanumeric login string entered by the user and associated with the session establishment or rejection.
- **Status** — Contains a code that indicates whether the session was established or rejected and, if rejected, the reason for the rejection.

Related topics

For more information on ASG, see "Access Security Gateway" in *Feature Description and Implementation for Avaya Communication Manager*, 555-245-205.

For ASG login procedures, see "Logging in" in *Feature Description and Implementation for Avaya Communication Manager*, 555-245-205.

For more information on SVN, see "Security Violations Notification" in *Feature Description and Implementation for Avaya Communication Manager*, 555-245-205.

Changing Login Permissions

This section shows you how to change login permissions.

Once you have created a login, you can modify the permissions associated with the login. The system maintains default permissions for each level of login, but you may want to further restrict the login, or at least make sure the defaults are appropriate for the user. The default values for these fields vary based on the login type.

Instructions

We will change the login permissions of **angi3**.

To change login permissions:

1. Type **change permissions angi3** and press **Enter**.

   The **Command Permission Categories** screen appears.

Figure 164: Command Permissions Categories screen

```
               COMMAND PERMISSION CATEGORIES
                         Login Name: angi3
COMMON COMMANDS
               Display Admin. and Maint. Data? n
                         System Measurements? n

ADMINISTRATION COMMANDS
      Administer Stations? y             Administer Features? n
        Administer Trunks? n           Administer Permissions? n
  Additional Restrictions? y

MAINTENANCE COMMANDS
        Maintain Stations? n       Maintain Switch Circuit Packs? n
          Maintain Trunks? n      Maintain Process Circuit Packs? n
         Maintain Systems? n               Maintain Enhanced DS1? n
```

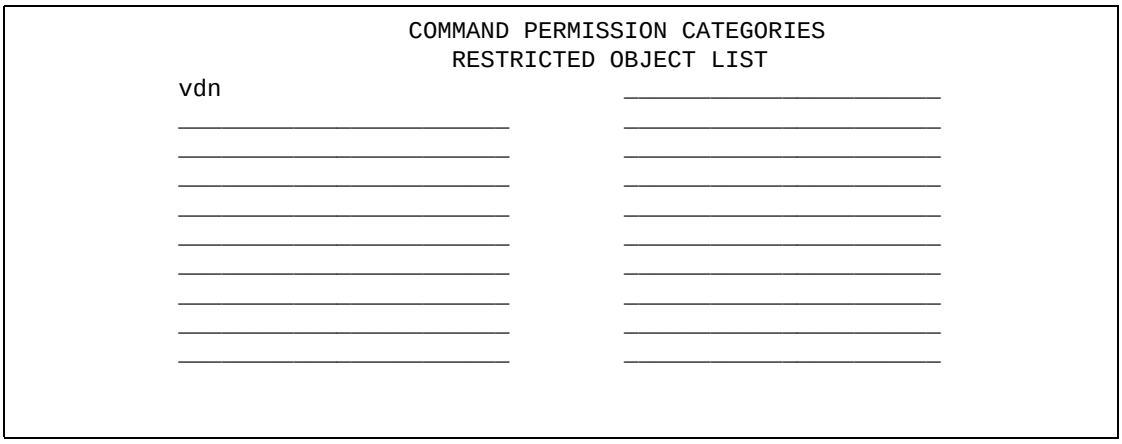
2. In the **Administer Stations** field, type **y**.

   This allows your user to add, change, duplicate, or remove stations, data modules and associated features.

3. In the **Additional Restrictions** field, type **y**.

   A **y** in this field brings up the second and third pages of this screen.

Figure 165: Command Permissions Categories screen - Restricted Objects



4. In the first field, type **vdn**.

   This restricts your user from administering a VDN.

5. Press **Enter** to save your changes.

Changing Passwords

This section shows you how to change a user's password.

Instructions

We will change the password for login **angi3** to **g3or5e**.

To change passwords:

1. Type **`change password angi3`** and press **Enter**.

   The **Password Administration** screen appears.

Figure 166: Password Administration screen

```
                                PASSWORD ADMINISTRATION

Password of Login Making Change: angi3

LOGIN BEING CHANGED
                        Login Name:

LOGIN'S PASSWORD INFORMATION
                  Login's Password:
          Reenter Login's Password:
```

2. In the **Password of Login Making Change** field, type your password to change any field on this screen.

   Type **angi3**.

3. In the **Login's Password** field, type the initial password for this login.

   Type **g3or5e**.

   Notify the owner of the login to change their password immediately. The password does not appear on the screen as you type.

   A password must be from 4 to 11 characters in length and contain at least 1 alphabetic and 1 numeric symbol.

4. In the **Reenter Login's Password** field, retype the login's password as above, for verification.

   Type **g3or5e**.

   The password does not appear on the screen as you type.

5. Press **Enter** to save your changes.

Using Busy Verify

This section shows you how to use Busy Verify (also known as Busy Verification) to help find fraud problems.

When you suspect toll fraud, you can interrupt the call on a specified trunk group or extension number and monitor the call in progress. Callers will hear a long tone to indicate the call is being monitored.

SECURITY ALERT:

Listening to someone else's calls may be subject to federal, state, or local laws, rules, or regulations. It may require the consent of one or both of the parties on the call. Familiarize yourself with all applicable laws, rules, and regulations and comply with them when you use this feature.

Before you start

On the **Trunk Group** screen, verify the **Dial Access** field is **y**. If not, contact your Avaya technical support representative.

Instructions

To use busy verify:

1. Type **change station *xxxx***, where ***xxxx*** is the station to be assigned the busy verify button. Press **Enter**.

   For this example, enter extension **1014**.

   The **Station** screen appears.

Figure 167: Station screen

```
                                                          Page 3 of X
                              STATION
    SITE DATA
         Room: _______                                  Headset? n
         Jack: ___                                      Speaker? n
        Cable: ___                                     Mounting: d
        Floor: _______                              Cord Length: 0_
     Building: _______                                Set Color: _____


ABBREVIATED DIALING
     List1: _________          List2: _________          List3: _________

BUTTON ASSIGNMENTS
 1: call-appr                  4: verify__
 2: call-appr                  5: _________
 3: call-appr
```

2. In the **BUTTON ASSIGNMENTS** area, type **verify**.
3. Press **Enter** to save your changes.
4. To activate the feature, press the **Verify** button on the telephone and then enter the Trunk Access Code and member number to be monitored.

Setting up Authorization Codes

Authorization codes provide the means for extending control of system users' calling privileges. They extend calling-privilege control and provide an extra level of security for remote-access callers.

Note:

To maintain system security, Avaya recommends you use authorization codes.

See the *Avaya Toll Fraud and Security Handbook* for more information.

Before you start

On the **System-Parameters Customer-Options** screen, verify the **Authorization Codes** field is **y**. If not, contact your Avaya representative. This field turns on the feature and permits you to selectively specify levels of calling privileges that override in-place restrictions.

Instructions

To set up authorization codes:

1. Type **`change system-parameters features`** and press **Enter**.

   The **Feature-Related System Parameters** screen appears. Click **Next** until you find the **Authorization Code Enabled** field.

Figure 168: Feature-Related System Parameters screen

```
                                                          Page 3 of x
                     FEATURE-RELATED SYSTEM PARAMETERS

                      Reserved Slots for Attendant Priority Queue: 5_
                                       Time Before Off-Hook Alert: 10__
                            Emergency Access Redirection Extension: _____
             Number of Emergency Calls Allowed in Attendant Queue: __
                                            Call Pickup Alerting? n
                      Temporary Bridged Appearance on Call Pickup? y
                                   Call Pickup on Intercom Calls? y
                                            Directed Call Pickup? n
                Deluxe Paging and Call Park Timeout to Originator? n
Controlled Outward Toll Restriction Intercept Treatment: tone _______
    Controlled Termination Restriction (Do Not Disturb): tone _______
             Controlled Station to Station Restriction: tone _______
AUTHORIZATION CODE PARAMETERS           Authorization Code Enabled? y
                                         Authorization Code Length: 7
                           Authorization Code Cancellation Symbol? #
                                           Attendant Time Out Flag? n
                                       Display Authorization Code? _
                   Controlled Toll Restriction Replaces: station-station
         Controlled Toll Restriction Intercept Treatment: extension     3000
```

2. In the **Authorization Code Enabled** field, type **y**.

   This enables the Authorization Codes feature on a system-wide basis.

3. In the **Authorization Code Length** field, type **7**.

   This defines the length of the Authorization Codes your users need to enter. To maximize the security of your system, Avaya recommends you make each authorization code the maximum length allowed by the system.

4. In the **Authorization Code Cancellation Symbol** field, leave the default of **#**.

   This is the symbol a caller must dial to cancel the 10-second wait period during which your user can enter an authorization code.

5. In the **Attendant Time Out Flag** field, leave the default of **n**.

   This means a call is not to be routed to the attendant if a caller does not dial an authorization code within 10 seconds or dials an invalid authorization code.

6. In the **Display Authorization Code** field, type **n**.

   This prevents the authorization code from displaying on telephone sets thus maximizing your security.

7. Press **Enter** to save your changes.

8. Type **change authorization-code *nnnn***, where **nnnn** is the authorization code. Press **Enter**.

   The **Authorization Code — COR Mapping** screen appears.

Figure 169: Authorization Code - COR Mapping screen

```
                                                        Page 1 of 1
                    Authorization Code - COR Mapping
  Note:    XX codes administered.  Use "list" to display all codes.
 AC     COR   AC     COR   AC     COR   AC     COR   AC     COR    AC     COR
4285193 1_  _______ __  _______ __  _______ __  _______ __  _______ __  _______ __
_______ __  _______ __  _______ __  _______ __  _______ __  _______ __  _______ __
_______ __  _______ __  _______ __  _______ __  _______ __  _______ __  _______ __
_______ __  _______ __  _______ __  _______ __  _______ __  _______ __  _______ __
_______ __  _______ __  _______ __  _______ __  _______ __  _______ __  _______ __
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_______ __  _______ __  _______ __  _______ __  _______ __  _______ __  _______ __
_______ __  _______ __  _______ __  _______ __  _______ __  _______ __  _______ __
_______ __  _______ __  _______ __  _______ __  _______ __  _______ __  _______ __
```

9. In the **AC** field, enter the authorization code your users must dial.

   In this example, type **4285193**. The number of digits entered must agree with the number assigned in the **Feature-Related System Parameters** screen, **Authorization Code Length** field.

   Note:

   Remember, all authorization codes used in the system must be the same length.

10. In the **COR** field, enter the desired Class of Restriction number from 0 through 95.

    In our example, type **1**.

11. Press **Enter** to save your changes.

Related topics

See "Class of Restiction" in *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205 for more information on setting up dialing out restrictions.

See *Administration for Network Connectivity for Avaya Communication Manager* for more information on using trunk access codes.

See "Facility Restriction Levels and Traveling Class Marks" *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205, and Route Pattern on page 1367 for more information on assigning Facility Restriction Levels.

See "Call Detail Recording" in *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205, and Station on page 1409 for more information on using Call Detail Recording (CDR) on station phones.

See Class of Restriction on page 813 and Station on page 1409 for more information on using Class of Restriction (COR) on station phones.

See "Remote Access" in *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205, for more information on allowing authorized callers to access the system from remote locations.

See "Barrier Codes" in *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205, or QSIG to DCS TSC Gateway on page 1353 for information on barrier codes.

Setting up Security Violations Notification

This section shows you how to use Security Violations Notification (SVN) to set security-related parameters and to receive notification when established limits are exceeded. You can run reports related to invalid access attempts. You also can disable a login ID or remote access authorization that is associated with a security violation.

When a security violation has occurred, there are steps that you can take to be sure that this same attempt is not successful in the future. See the *BCS Products Security Handbook* for more information.

Instructions

To set up Security Violations Notification:

1. Type **`change system-parameters security`** and press **Enter**.

   The system displays the **Security-Related System Parameters** screen.

Figure 170: Security-Related System Parameters screen

```
change system-parameters security                                Page 1 of 2
                    SECURITY-RELATED SYSTEM PARAMETERS

SECURITY VIOLATION NOTIFICATION PARAMETERS

  SVN Login Violation Notification Enabled? y
       Originating Extension: _____ Referral Destination: _____
             Login Threshold: 5_          Time Interval: 0:03
      Announcement Extension: _____

  SVN Remote Access Violation Notification Enabled? y
       Originating Extension: _____ Referral Destination: _____
      Barrier Code Threshold: 10          Time Interval: 0:03
      Announcement Extension: _____

  SVN Authorization Code Violation Notification Enabled? y
       Originating Extension: _____ Referral Destination: _____
 Authorization Code Threshold: 10          Time Interval: 0:03
      Announcement Extension: _____
```

2. In the **SVN Login Violation Notification Enabled** field, type **y**.

   This sets Security Violations Notification login violation notification.

   Note:

   If you are not using Security Violation Notification for logins, type **n** in the **SVN Login Violation Notification Enabled** field and go to Step 11.

3. In the **Originating Extension** field, type **3040**.

   This becomes the telephone extension for the purpose of originating and identifying SVN referral calls for login security violations.

4. In the **Referral Destination** field, type **attd** to send all calls to the attendant.

   This is the telephone extension that receives the referral call when a security violation occurs.

5. In the **Login Threshold** field, type **3**.

   This is the minimum number of login attempts that are permitted before a referral call is made. More than 3 attempts causes a security violation notification.

6. In the **Time Interval** field, type **0:03**.

   This the time interval in which the threshold, or number of violations, must occur.

7. Press **Enter** to save your changes.

8. (Optional) Type **change login** ***nnnn***, where ***nnnn*** is your login ID.

   The system displays the **Login Administration** screen.

Figure 171: Login Administration screen

```
change login nnnn                                              Page 1 of 2
                              LOGIN ADMINISTRATION

               Password of Login Making Change:

           LOGIN BEING ADMINISTERED
                              Login's Name:xxxxxxx
                                Login Type:
                              Service Level:
    Disable Following a Security Violation?
                                                    Access to INADS Port? _

          LOGIN'S PASSWORD INFORMATION
                          Login's Password:
                   Reenter Login's Password:
        Password Aging Cycle Length (Days):

        LOGOFF NOTIFICATION
          Facility Test Call Notification?     Acknowledgment Required?
               Remote Access Notification?     Acknowledgment Required?

ACCESS SECURITY GATEWAY PARAMETERS
Access Security Gateway?
```

9. (Optional) In the **Disable Following A Security Violation** field, type **y**.

   This disables a login following detection of a login security violation for the login you are administering.

10. Press **Enter** to save your changes.

    Note:

    If you are not using Remote Access, go to Step 14.

11. (Optional) Type **change remote-access** and press **Enter**.

    The **Remote Access** screen appears.

Figure 172: Remote Access screen

```
                                REMOTE ACCESS
   Remote Access Extension_________          Barrier Code Length____
Authorization Code Required? y                    Remote Access Dial Tone: n

Barrier Code   COR  TN  COS    Expiration Date   No. of Calls    Calls Used
1:_________    1__  1_  1__    __/__/__          ______          ______
2:_________    1__  1_  1__    __/__/__          ______          ______
3:_________    1__  1_  1__    __/__/__          ______          ______
4:_________    1__  1_  1__    __/__/__          ______          ______
5:_________    1__  1_  1__    __/__/__          ______          ______
6:_________    1__  1_  1__    __/__/__          ______          ______
7:_________    1__  1_  1__    __/__/__          ______          ______
8:_________    1__  1_  1__    __/__/__          ______          ______
9:_________    1__  1_  1__    __/__/__          ______          ______
10:________    1__  1_  1__    __/__/__          ______          ______

Permanently Disable? __ Disable Following A Security Violation? y
(NOTE:  You must logoff to effect permanent disabling of Remote Access)
```

12. (Optional) In the **Disable Following A Security Violation** field, type **y**.

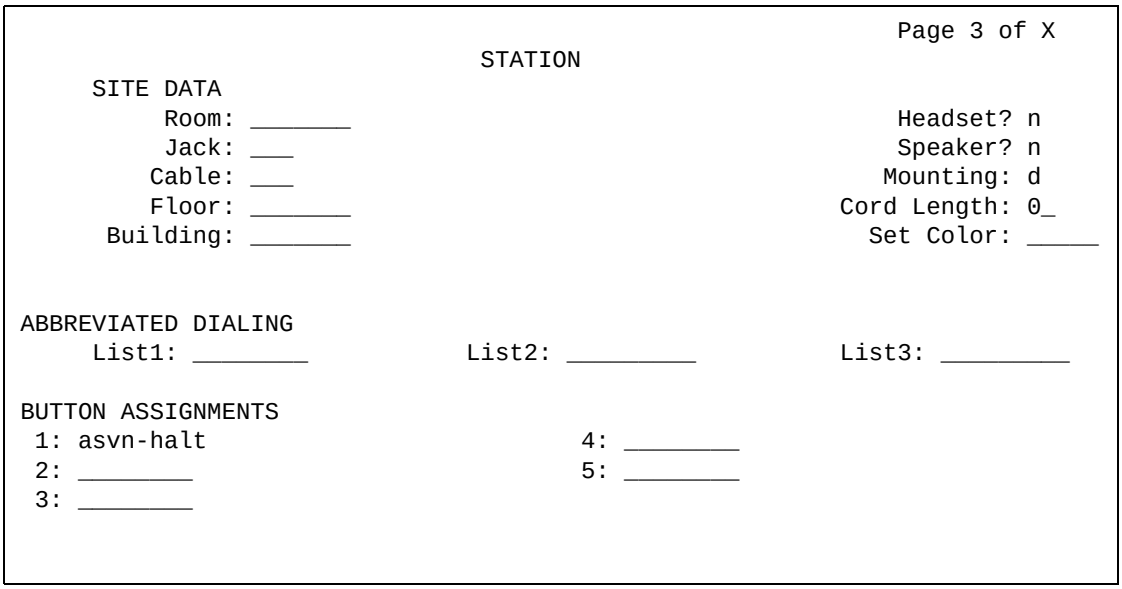
    This disables Remote Access following detection of a remote access security violation.

13. (Optional) Press **Enter** to save your changes.

14. Type **change station *xxxx***, where ***xxxx*** is the station to be assigned the notification halt button. Press **Enter**.

    The **Station** screen appears.

Figure 173: Station screen



15. In the Feature Button Assignments section, type one of the following:
    - **asvn-halt** — The Authorization Code Security Violation Notification call is activated when an authorization code security violation is detected. This applies only if you are using authorization codes.
    - **lsvn-halt** — The Login Security Violation Notification call is activated a referral call when a login security violation is detected.
    - **rsvn-halt** — The Remote Access Barrier Code Security Violation Notification call is activated as a call referral. This applies only if you are using Remote Access barrier codes.
    - **ssvn-halt** — The Station Code Security Violation Notification call is activated when a station code security violation is detected. This applies only if you are using station codes.

    Note:

    Any of the above 4 security violations will cause the system to place a notification call to the designated telephone. The call continues to ring until answered. To stop notification of any further violations, press the button associated with the type of violation.
16. Press **Enter** to save your changes.

Using Station Lock

Station Lock allows only authorized users to make external telephone calls from a particular station.

Note:

Avaya recommends digital phones use a Station Lock button rather than a feature access code.

Setting Station Lock

We will set Station Lock to allow authorized users to access the system through a particular station.

Before you start

- Be sure the **Station Lock COR** field on the **Class of Restriction** screen has the COR the user is using to define the calling restrictions.

Instructions

We will set Station Lock on a digital telephone (extension 7262):

1. Type **change station 7262** and press **Enter**.
2. In the **Security Code** field, enter a security code of up to 8 digits.

   In the **COR** field, leave the default at **1**.
3. In the **BUTTON ASSIGNMENTS** section, type **sta-lock**.
4. Press **Enter** to save your changes.
5. Type **change cor 1** and press **Enter**.
6. In the **Calling Party Restriction** field, type **none**.

   This means that no calling party restrictions exist on extension 7262.
7. In the **Station Lock COR** field, type **2**.
8. Press **Enter** to save your changes.
9. Type **change cor 2** and press **Enter**.

10. In the **Calling Party Restriction** field, verify it is **outward**.
11. Press **Enter** to save your changes.

    Now when extension 7262 activates Station Lock, calling restrictions are determined by the Station Lock COR, COR 2. Based on the administration of COR 2, extension 7262 is not allowed to call outside the private network. When Station Lock is not active on extension 7262, calling restrictions are determined by the COR administered on the **Station** screen, COR 1. In this example, when extension 7262 is unlocked, calls outside the private network are allowed.

Set Station Lock on an analog, x-mobile, or digital telephone without a Station Lock button (extension 7262 and use a feature access code of 08):

1. Type **change station 7262** and press **Enter**.
2. In the **Security Code** field, enter a security code of up to 8 digits.

   In the **COR** field, leave the default at 1. This means that anyone can call outside on extension 7262.
3. Press **Enter** to save your changes.
4. Type **change system-parameters features** and press **Enter**.

   The **Feature-Related System-Parameters** screen appears.
5. In the **Special Dial Tone** field, type **y** for an audible tone indicating the station is locked.
6. Press **Enter** to save your changes.
7. Type **change feature-access-codes** and press **Enter**.

   The **Feature Access Codes** screen appears.
8. Move the cursor to the **Station Lock Activation** field.
9. In the **Activation** field, type ***08**.
10. In the **Deactivation** field, type **#08**.
11. Press **Enter** to save your changes.

    Now when a user activates Station Lock, no one can call outside from extension 7262.

Dealing with Security Violations

When a security violation occurs, there are steps that you can take to be sure that this same attempt is not successful in the future.

Disabling a login ID

There may be occasions when you have to disable a login for one of your users because of a security violation.

1. Log in to Avaya Communication Manager using a login ID with the correct permissions.
2. Type **disable login *nnnnn***, where ***nnnnn*** is the login you want to disable, and press **Enter**.

Enabling a login ID

You may have to enable a login ID that has been disabled by a security violation, or disabled manually with the disable login command.

1. Log in to Avaya Communication Manager using a login ID with the correct permissions.
2. Type **enable login *nnnnn***, where ***nnnnn*** is the login you want to enable, and press **Enter**.

Enabling remote access

You may have to enable Remote Access that has been disabled following a security violation, or disabled manually.

1. Log in to Avaya Communication Manager using a login ID with the correct permissions.
2. Type **enable remote-access** and press **Enter**.

Disabling remote access

There may be occasions when you have to disable remote access for one of your users because of a security violation.

1. Log in to Avaya Communication Manager using a login ID with the correct permissions.
2. Type **disable remote-access** and press **Enter**.

Chapter 12: Managing Trunks

Tips for Working with Trunk Groups

You'll find detailed procedures for administering specific trunk groups elsewhere in this chapter. However, there's more to working with trunks than just administering trunk groups.

Following a process

Trunking technology is complex. Following a process can prevent mistakes and save you time. To set up new trunks and trunk groups, Avaya recommends following the process below (some steps may not apply to your situation):

1. Install the necessary circuit packs and perform any administration the circuit pack requires.
2. Connect the appropriate ports to your network service provider's trunks.
3. Administer a trunk group to control the operation of the trunks.
4. Assign the ports you're using to the trunk group.
5. For outgoing or 2-way trunks, administer Automatic Route Selection so Avaya Communication Manager knows which outgoing calls to route over this trunk group.
6. Test your new trunk group. Using the trunk access code, place a variety of calls.

This chapter provides instructions for steps 3 and 4 in this process.

Working with your network service provider

Depending on the type of trunk you want to add, the vendor may be your local telephone company, a long distance provider, or some other service provider. Key settings on Avaya Communication Manager must be identical to the same settings on the provider's equipment for your trunks to work. Clear, frequent communication with your provider is essential — especially since some providers may use different terms and acronyms than Avaya does!

Once you decide that you want to add a new trunk, contact your vendor. The vendor should confirm the type of signal you want and provide you with a circuit identification number for the new trunk. Be sure to record any vendor-specific ID numbers or specifications in case you ever have any problems with this trunk.

Keeping records

In addition to recording vendor-specific information such as ID numbers, you should record the following information about every trunk group you have.

The questions you need to answer	The kind of information you need to get
What type of trunk group is it?	You need to know what kind of trunks these are (central office (CO), foreign exchange (FX), etc.) and whether they use any special services (such as T1 digital service). You also need to know what kind of signaling the group uses. For example, you might have a CO trunk group with ground-start signaling running on a robbed-bit T1 service.
Which telephone numbers are associated with each trunk group?	For incoming or two-way trunk groups: • What number or numbers do outside callers use to call into your server over this group? • What is the destination extension to which this trunk group delivers calls? Does it terminate at an attendant or a voice-mail system? For outgoing trunk groups: • What extensions can call out over this trunk group?
Is the service from your network service provider sending digits on incoming calls?	Direct Inward Dial and Direct Inward/ Outward Dial trunks send digits to Avaya Communication Manager. Tie trunks may send digits, depending on how they're administered. You need to know: • How many digits is your service provider sending? • Are you inserting any digits? What are they? • Are you absorbing any digits? How many? • What range of numbers has your service provider assigned you?

Helpful tips for setting common fields

The procedures in this section cover the specific fields you must administer when you create each type of trunk group. Here are some tips for working with common fields that are available for most trunk groups.

- **Dial Access** — Typing **y** in this field allows users to route calls through an outgoing or two-way trunk group by dialing its trunk access code.

  SECURITY ALERT:
  Calls dialed with a trunk access code over Wide Area Telecommunications Service (WATS) trunks are not validated against the ARS Digit Analysis Table, so users can dial anything they wish. For security, you may want to leave the field set to **n** unless you need dial access to test the trunk group.

- **Outgoing Display** — Typing **y** in this field allows display phones to show the name and group number of the trunk group used for an outgoing call. This information may be useful to you when you're trying to diagnose trunking problems.
- **Queue Length** — Don't create a queue for two-way loop-start trunks, or you may have a problem with glare (the interference that happens when a two-way trunk is seized simultaneously at both ends).
- **Trunk Type** — Use ground-start signaling for two-way trunks whenever possible: ground-start signaling avoids glare and provides answer supervision from the far end. Try to use loop-start signaling only for one-way trunks.

Related topics

See the *Hardware Guide for Avaya Communication Manager,* 555-245-207, for information on the types of circuit packs available and their capacities.

See your server's Installation manual for circuit-pack installation instructions.

See Modifying Call Routing on page 271 for detailed information on Automatic Route Selection.

Adding a CO, FX, or WATS Trunk Group

Basic administration for Central Office (CO), Foreign Exchange (FX), and WATS trunk groups is identical, so we've combined instructions for all 3 in the following procedure. In most cases, Avaya recommends leaving the default settings in fields that aren't specifically mentioned in the following instructions. Your Avaya representative or network service provider can give you more information. Your settings in the following fields *must* match your provider's settings:

- Direction
- Comm Type
- Trunk Type

CAUTION:

Use the list above as a starting point and talk to your service provider. Depending on your particular application, you may need to coordinate additional administration with your service provider.

Before you start

Before you can administer any trunk group, you must have one or more circuit packs of the correct type with enough open ports to handle the number of trunks you need to add. To find out what circuit packs you need, see the *Hardware Guide for Avaya Communication Manager.*

Instructions

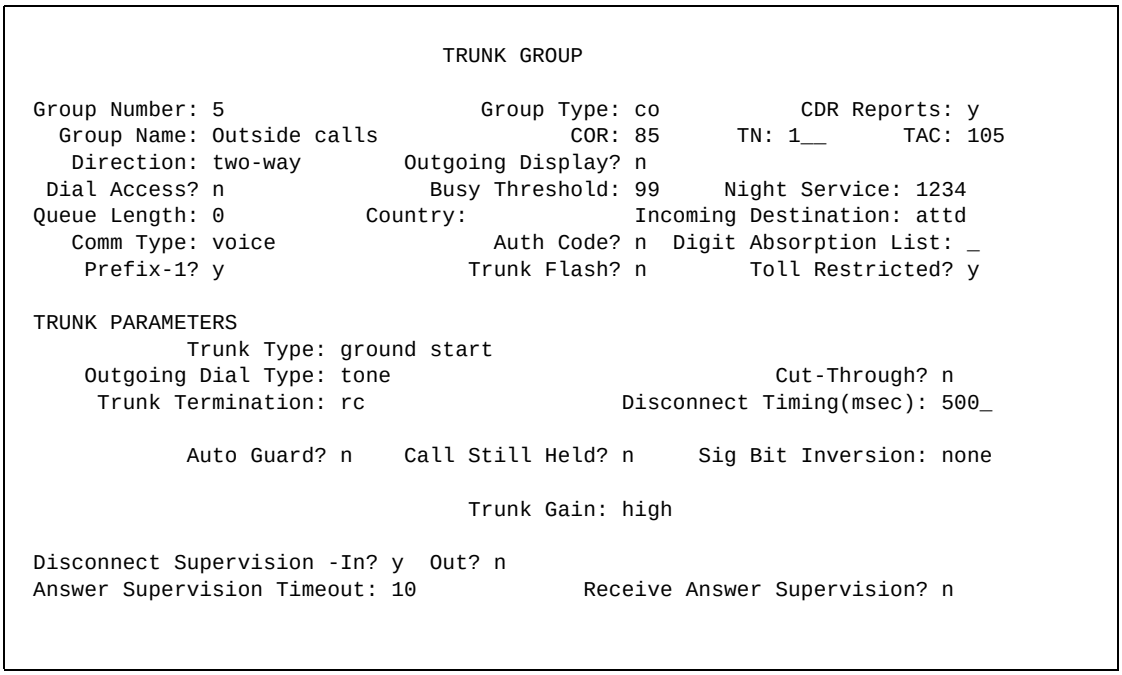
As an example, we will set up a two-way CO trunk group that carries voice and voice-grade data only. Incoming calls terminate to an attendant during business hours and to a night service destination the rest of the time

To add the new CO trunk-group:

1. Type **`add trunk-group next`** and press **Enter**.

   The **Trunk Group** screen appears. The system assigns the next available trunk group number to this group. In our example, we're adding trunk group 5.

Figure 174: Trunk Group screen



```
                                TRUNK GROUP

Group Number: 5                     Group Type: co           CDR Reports: y
  Group Name: Outside calls               COR: 85      TN: 1__      TAC: 105
   Direction: two-way        Outgoing Display? n
 Dial Access? n                Busy Threshold: 99     Night Service: 1234
Queue Length: 0           Country:              Incoming Destination: attd
   Comm Type: voice                 Auth Code? n  Digit Absorption List: _
    Prefix-1? y                   Trunk Flash? n        Toll Restricted? y

TRUNK PARAMETERS
            Trunk Type: ground start
    Outgoing Dial Type: tone                              Cut-Through? n
     Trunk Termination: rc                    Disconnect Timing(msec): 500_

            Auto Guard? n    Call Still Held? n     Sig Bit Inversion: none

                                 Trunk Gain: high

Disconnect Supervision -In? y  Out? n
Answer Supervision Timeout: 10             Receive Answer Supervision? n
```

2. In the **Group Type** field, type **co**.

   This field specifies the kind of trunk group you're creating.

3. In the **Group Name** field, type **Outside calls**.

   This name will be displayed, along with the group number, for outgoing calls if you set the **Outgoing Display?** field to **y**. You can type any name up to 27 characters long in this field.

4. In the **COR** field, type **85**.

   This field controls which users can make and receive calls over this trunk group. Assign a class of restriction that's appropriate for the COR calling permissions administered on your system.

5. In the **TAC** field, type **105**.

   This field defines a unique code that you or your users can dial to access this trunk group. The code also identifies this trunk group in call detail reports.

6. In the **Direction** field, type **two-way**.

   This field defines the direction of traffic flow on this trunk group.

7. In the **Night Service** field, type **1234**.

   This field assigns an extension to which calls are routed outside of business hours.

8. In the **Incoming Destination** field, type **attd**.

   This field assigns an extension to which incoming calls are routed during business hours. By entering attd in this field, incoming calls go to the attendant and the system treats the calls as Listed Directory Number calls.

9. In the **Comm Type** field, type **voice**.

   This field defines whether a trunk group can carry voice, data, or both. Analog trunks only carry voice and voice-grade data.

10. In the **Trunk Type** field, type **ground-start**.

    This field tells the system what kind of signaling to use on this trunk group. To prevent glare, Avaya recommends ground start signaling for most two-way CO, FX, and WATS trunk groups.

11. In the **Outgoing Dial Type** field, type **tone**.

    This field tells Communication Manager how digits are to be transmitted for outgoing calls. Entering tone actually allows the trunk group to support both dual-tone multifrequency (DTMF) and rotary signals, so Avaya recommends that you always put tone in this field.

12. In the **Trunk Termination** field, type **rc**.

    Use **rc** in this field when the distance to the central office or the media server at the other end of the trunk is more than 3,000 feet. Check with your service provider if you're not sure of the distance to your central office.

13. Press **Enter** to save your changes.

Now you are ready to add trunks to this trunk group. See Adding Trunks to a Trunk Group on page 443.

Adding a DID Trunk Group

In most cases, Avaya recommends leaving the default settings in fields that aren't specifically mentioned in the following instructions. Your Avaya representative or network service provider can give you more information. For Direct Inward Dialing (DID) trunk groups, settings in the following fields *must* match your provider's settings:

- Direction
- Comm Type
- Trunk Type
- Expected Digits (only if the digits your provider sends *do not* match your dial plan)

CAUTION:

Use the list above as a starting point and talk to your service provider. Depending on your particular application, you may need to coordinate additional administration with your service provider.

Before you start

Before you can administer any trunk group, you must have one or more circuit packs of the correct type with enough open ports to handle the number of trunks you need to add. To find out what circuit packs you need, see the *Hardware Guide for Avaya Communication Manager.*

Tip:

In the **DID/Tie/ISDN Intercept Treatment** field on the **Feature-Related System Parameters** screen, enter **attd**. Incoming calls to invalid extensions will be routed to the attendant.

Instructions

To add the new Direct Inward Dialing (DID) trunk-group:

1. Type **`add trunk-group next`** and press **Enter**.

   The **Trunk Group** screen appears. The system assigns the next available trunk group number to this group. In our example, we're adding trunk group 5.

Figure 175: Trunk Group screen

```
                                TRUNK GROUP

Group Number: 5               Group Type: did             CDR Reports: y
  Group Name: Incoming calls         COR: 85          TN: 1       TAC: 105

                     Country: 1
                                Auth Code? n


TRUNK PARAMETERS
         Trunk Type: wink-start   Incoming Rotary Timeout(sec): 5
                                            Incoming Dial Type: tone
  Trunk Termination: rc                Disconnect Timing(msec): 500
    Digit Treatment: insertion                          Digits: 6
    Expected Digits: 4                       Sig Bit Inversion: none
  Analog Loss Group: ___                    Digital Loss Group: ___
Extended Loop Range? n      Trunk Gain: high    Drop Treatment: silence

 Disconnect Supervision - In? y
```

2. In the **Group Type** field, type **did**.

   This field specifies the kind of trunk group you're creating.

3. In the **Group Name** field, type **Incoming calls**.

   You can type any name up to 27 characters long in this field.

4. In the **COR** field, type **85**.

   This field controls which users can receive calls over this trunk group. Assign a class of restriction that's appropriate for the COR calling permissions administered on your system.

5. In the **TAC** field, type **105**.

   This code identifies the trunk group on CDR reports.

6. In the **Trunk Type** field, type **wink-start**.

   This field tells the system what kind of signaling to use on this trunk group. In most situations, use wink start for DID trunks to minimize the chance of losing any of the incoming digit string.

7. In the **Incoming Dial Type** field, type **tone**.

   This field tells Communication Manager how digits are transmitted for incoming calls. Entering tone actually allows the trunk group to support both DTMF and rotary signals, so Avaya recommends that you always put tone in this field.

8. In the **Trunk Termination** field, type **rc**.

   Use rc in this field when the distance to the central office or the media server at the other end of the trunk is more than 3,000 feet. Check with your service provider if you're not sure of the distance to your central office.

9. Press **Enter** to save your changes.

Now you're ready to add trunks to this trunk group. See Adding Trunks to a Trunk Group on page 443.

Related topics

See Inserting and Absorbing Digits on page 446 for instructions on matching modifying incoming digit strings to match your dial plan.

Adding a PCOL Trunk Group

In most cases, when administering Personal Central Office Line (PCOL) trunk groups, Avaya recommends leaving the default settings in fields that aren't specifically mentioned in the following instructions. Your Avaya representative or network service provider can give you more information. Your settings in the following fields *must* match your provider's settings:

- Trunk Type
- Trunk Direction

CAUTION:

Use the list above as a starting point and talk to your service provider. Depending on your particular application, you may need to coordinate additional administration with your service provider.

Before you start

Before you can administer any trunk group, you must have one or more circuit packs of the correct type with enough open ports to handle the number of trunks you need to add. To find out what circuit packs you need, see the *Hardware Guide for Avaya Communication Manager,* 555-245-207.

Instructions

As an example, we will set up a new PCOL group and administer the group as a CO trunk for two-way voice traffic.

To add the new PCOL group:

1. Type **add personal-co-line next** and press **Enter**.

   The **Personal CO Line Group** screen appears.

Figure 176: Personal CO Line Group screen

```
                                                              Page 1 of X
                            PERSONAL CO LINE GROUP
 Group Number: 1                    Group Type: co            CDR Reports: y
   Group Name: OUTSIDE CALL                                         TAC: ____
Security Code: ___               Coverage Path: ____     Data Restriction? _
                              Outgoing Display? _

TRUNK PARAMETERS
                 Trunk Type: ground-start            Trunk Direction: two-way
                 Trunk Port: 01D1901_        Disconnect Timing(msec): 500
                 Trunk Name: __________             Trunk Termination: rc____
         Outgoing Dial Type: tone_____              Analog Loss Group: 7
                   Prefix-1? y                    Digital Loss Group: _
 Answer Supervision Timeout: ___          Receive Answer Supervision? _
                 Trunk Gain: high                           Country: 1
           Charge Conversion: 1____
               Decimal Point: none__
             Currency Symbol: ___
                 Charge Type: units__
```

2. In the **Group Type** field, type **co**.

   This field specifies the kind of trunk group you're creating. PCOL groups can be administered as CO, FX, or WATS trunks.

3. In the **Group Name** field, type **Outside calls**.

   This name will be displayed, along with the group number, for outgoing calls if you set the **Outgoing Display?** field to **y**. You can type any name up to 27 characters long in this field. (You may want to put the telephone number here that's assigned to this trunk.)

4. In the **TAC** field, type **111**.

   This field defines a unique code that you or your users can dial to access this trunk group. The code also identifies this trunk group in call detail reports.

5. In the **Trunk Type** field, type **ground start**.

   This field tells the system what kind of signaling to use on this trunk group. To prevent glare, Avaya recommends ground start signaling for most two-way CO, FX, and WATS trunk groups.

6. In the **Trunk Port** field, type **01D1901**.

   This is the port to which the trunk is connected.

7. In the **Trunk Termination** field, type **rc**.

   Use rc in this field when the distance to the central office or the media server at the other end of the trunk is more than 3,000 feet. Check with your service provider if you're not sure of the distance to your central office.

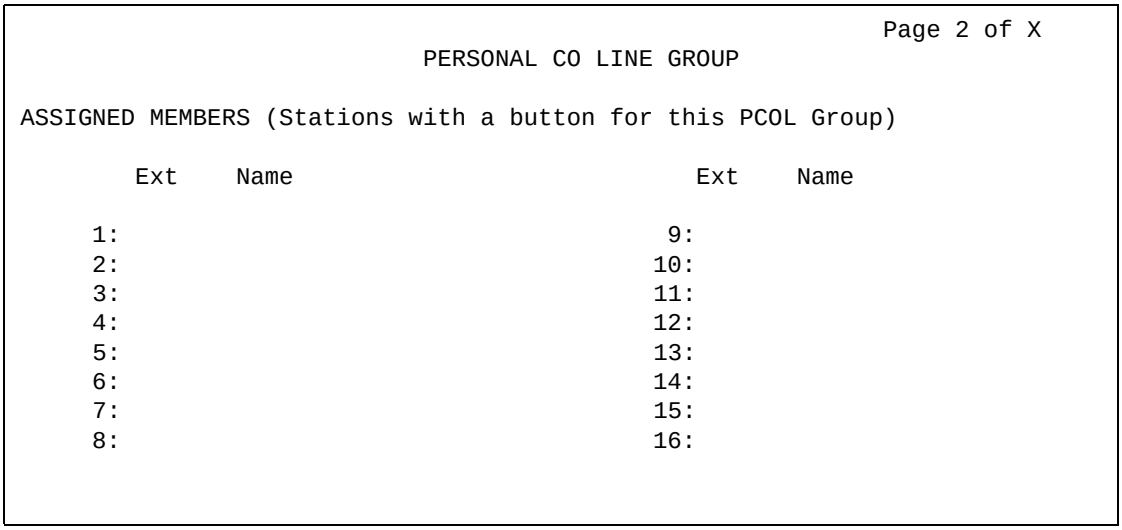
8. In the **Outgoing Dial Type** field, type **tone**.

   This field tells Communication Manager how digits are to be transmitted for outgoing calls. Entering tone actually allows the trunk group to support both DTMF and rotary signals, so Avaya recommends that you always put tone in this field.

9. Press **Enter** to save your changes.

You assign phones to a PCOL group by administering a CO Line button on each telephone. Once assigned, the **Assigned Members** page of the **Personal CO Line Group** screen displays member phones:

Figure 177: Personal CO Line Group screen



```
                                                          Page 2 of X
                          PERSONAL CO LINE GROUP

ASSIGNED MEMBERS (Stations with a button for this PCOL Group)

        Ext    Name                              Ext    Name

     1:                                        9:
     2:                                       10:
     3:                                       11:
     4:                                       12:
     5:                                       13:
     6:                                       14:
     7:                                       15:
     8:                                       16:
```

More information

Call Detail Recording

Call detail recording (CDR) can be activated for calls on a personal CO line, but the CDR record does not specifically identify the call as PCOL. Calls over personal CO lines can, however, be identified by the trunk access code used on the call. The call is recorded to the extension number assigned to the telephone where the call was originated or answered.

Restrictions

- Abbreviated Dialing can be used with a personal CO line, but the accessed lists are associated with the individual phones.
- Auto Hold and Leave Word Calling do not work with calls on a personal CO line.
- Send All Calls cannot be activated for a personal CO line.
- INTUITY AUDIX cannot be in the coverage path of a PCOL group.
- Only phones in the same PCOL group can bridge onto calls on the personal CO line. If a user is active on his or her primary extension number on a PCOL call, bridged call appearances of that extension number cannot be used to bridge onto the call.
- When a user puts a call on hold on a personal CO line, the status lamp associated with the PCOL button does not track the busy/idle status of the line.

Adding a Tie or Access Trunk Group

In most cases, Avaya recommends leaving the default settings in fields that aren't specifically mentioned in the following instructions. Your Avaya representative or network service provider can give you more information. Your settings in the following fields *must* match your provider's settings (or the setting on the far-end media server, if this is a private network trunk group):

- Direction
- Comm Type
- Trunk Type

CAUTION:

Use the list above as a starting point and talk to your service provider. Depending on your particular application, you may need to coordinate additional administration with your service provider.

Before you start

Before you can administer any trunk group, you must have one or more circuit packs of the correct type with enough open ports to handle the number of trunks you need to add. To find out what circuit packs you need, see the *Hardware Guide for Avaya Communication Manager,* 555-245-207.

Tip:

In the **DID/Tie/ISDN Intercept Treatment** field on the **Feature-Related System Parameters** screen, enter **attd**. Incoming calls to invalid extensions will be routed to the attendant.

Instructions

As an example, we will add a two-way tie trunk group that supports voice and voice-grade data.

To add the new tie trunk-group:

1. Type **`add trunk-group next`** and press **Enter**.

   The **Trunk Group** screen appears. The system assigns the next available trunk group number to this group. In our example, we're adding trunk group 5.

Figure 178: Trunk Group screen

```
                                                          Page 1 of X
                                TRUNK GROUP

Group Number: 5__                  Group Type: tie           CDR Reports: y
  Group Name: Outside calls              COR: 85_      TN: 1__      TAC: 105_
   Direction: two-way_       Outgoing Display? n    Trunk Signaling Type: ____
 Dial Access? n                Busy Threshold: 99_         Night Service: 1234
Queue Length: 0__                                  Incoming Destination: _______
   Comm Type: voice                 Auth Code? n
                                  Trunk Flash? n

        BCC: 0
TRUNK PARAMETERS
   Trunk Type (in/out): wink/wink___     Incoming Rotary Timeout(sec): 5_
    Outgoing Dial Type: tone_____                 Incoming Dial Type: tone____
                                        Disconnect Timing(msec): 500_
       Digit Treatment: __________                        Digits: ____
       Expected Digits:__                       Sig Bit Inversion: none
     Analog Loss Group: ___                    Digital Loss Group: ___
    Incoming Dial Tone? y

 Disconnect Supervision - In? y  Out? n
  Answer Supervision Timeout: 0__           Receive Answer Supervision? y
```

2. In the **Group Type** field, type **tie**.

   This field specifies the kind of trunk group you're creating.

3. In the **Group Name** field, type **Outside calls**.

   This name will be displayed, along with the group number, for outgoing calls if you set the **Outgoing Display?** field to **y**. You can type any name up to 27 characters long in this field.

4. In the **COR** field, type **85**.

   This field controls which users can make or receive calls over this trunk group. Assign a class of restriction that's appropriate for the COR calling permissions administered on your system.

5. In the **TAC** field, type **105**.

   This field defines a unique code users can dial to access this trunk group.

6. In the **Direction** field, type **two-way**.

   This field defines the direction of traffic flow on this trunk group.

7. In the **Night Service** field, type **1234**.

   This field assigns an extension to which calls are routed outside of business hours.

8. In the **Comm Type** field, type **voice**.

   This field defines whether a trunk group can carry voice, data, or both. Analog trunks only carry voice and voice-grade data. If you're administering a T1 connection in North America, type **rbavd** in this field.

9. In the **Trunk Type** field, type **wink/wink**.

   This field tells the system what kind of signaling to use on this trunk group. Because we're receiving and sending digits over this trunk group, we're using wink/wink signaling to minimize the chance of losing part of the digit string in either direction.

10. Type **tone** in both the **Outgoing Dial Type** and **Incoming Dial Type** fields.

    These fields tell Communication Manager how digits are transmitted for incoming calls. Entering tone actually allows the trunk group to support both DTMF and rotary signals, so Avaya recommends that you always put tone in this field.

11. Press **Enter** to save your changes.

Now you're ready to add trunks to this trunk group. See Adding Trunks to a Trunk Group on page 443.

Adding a DIOD Trunk Group

Administration for Direct Inward and Outward Dialing (DIOD) trunk groups varies from country to country. See your local Avaya representative for more information. Remember that the central office serving your switching system may be emulating another country's network protocol. If so, you'll have to administer your circuit packs and trunk groups to match the protocol used by your central office.

If you're using Incoming Caller ID (ICLID) on analog trunks connected to a DIOD Central Office trunk circuit pack, DO NOT put these trunks in an outgoing AAR or ARS route pattern. Since the loop-start trunks supported on the DIOD Central Office trunk circuit pack do not provide answer supervision, the potential for toll fraud exists.

Setting up Digital Trunks

Any of the common trunks, except for PCOL trunks, can be analog or digital. (PCOL trunks can only be analog.) Administering a digital trunk group is very similar to administering its analog counterpart, but digital trunks must connect to a DS1 circuit pack and this circuit pack must be administered separately. The example in this section shows you how to do this.

In most cases, Avaya recommends leaving the default settings in fields that aren't specifically mentioned in the following instructions. Your Avaya representative or network service provider can give you more information.

Your settings in the following fields *must* match your provider's settings:

- Bit Rate
- Line Coding (unless you're using a channel service unit to convert between your line coding method and your provider's)
- Framing Mode
- Signaling Mode
- Interface Companding

CAUTION:

Use the list above as a starting point and talk to your service provider. Depending on your particular application, you may need to coordinate additional administration with your service provider.

Before you start

Assign the DS1 circuit pack before you administer the members of the associated trunk groups.

CAUTION:

If enhanced DS1 administration is not enabled, you cannot make changes to the DS1 Circuit Pack screen before you remove related member translations of all trunks from the trunk group. See Enhanced DS1 administration on page 441.

Before you can administer a digital trunk group, you must have one or more circuit packs that support DS1 with enough open ports to handle the number of trunks you need to add. To find out what circuit packs you need, see the *Hardware Guide for Avaya Communication Manager,* 555-245-207.

Instructions

The following example shows a DS1 circuit pack configured for T1 service. The circuit pack is supporting a two-way CO trunk group that carries only voice and voice-grade data.

To configure a new DS1 circuit pack:

1. Type **add ds1 07A19** and press **Enter**.

   The **DS1 Circuit Pack** screen appears. You must enter a specific port address for the circuit pack.

Figure 179: DS1 Circuit Pack screen

```
add ds1 xxxxxx                                              Page 1 of 2
                                DS1 CIRCUIT PACK

             Location: 07A19                         Name: two-way CO
             Bit Rate: 1.544                  Line Coding: b8zs
    Line Compensation: 1                      Framing Mode: esf
       Signaling Mode: ____________              D-Channel: __
              Connect: __________                 Interface: ____________
         Interconnect: ___                     Peer Protocol: ________
    TN-C7 Long Timers? _                    Country Protocol: ____
                                            Protocol Version: _
 Interface Companding: mulaw                          CRC? _
           Idle Code: 11111111
                                DCP/Analog Bearer Capability: ________
    MMI Cabling Board: _____  MMI Interface: ESM

 MAINTENANCE PARAMETERS

       Slip Detection? y                Near-end CSU Type: ________
```

2. In the **Name** field, type **two-way CO**.

   Use this name to record useful information such as the type of trunk group associated with this circuit pack or its destination.

3. In the **Bit Rate** field, type **1.544**.

   This is the standard for T1 lines.

4. In the **Line Coding** field, type **b8zs**.

   Avaya recommends you use b8zs whenever your service provider supports it. Since this trunk group only carries voice traffic, you could also use ami-zcs without a problem.

5. In the **Framing Mode** field, type **esf**.

   Avaya recommends you use esf whenever your service provider supports it.

6. In the **Signaling Mode** field, type **robbed-bit**.

7. In the **Interface Companding** field, type **mulaw**.

   This is the standard for T1 lines in North America.

8. Press **Enter** to save your changes.

More information

T1 recommended settings

The table below shows recommended settings for standard T1 connections to your local exchange carrier.

Field	Value	Notes
Line Coding	b8zs	Use ami-zcs if b8zs is not available.
Signaling Mode	robbed-bit	Robbed-bit signaling gives you 56K bandwidth per channel. If you need a 64K clear channel for applications like asynchronous data transmission or remote administration access, use common channel signaling.
Framing	esf	Use d4 if esf is not available.

If you use b8zs line coding and esf framing, it will be easier to upgrade your T1 facility to ISDN should you want to. You can upgrade without reconfiguring external channel service units, and your service provider won't have to reconfigure your network connection.

E1 recommended settings

DS1 administration for E1 service varies from country to country. See your local Avaya technical support representative for more information.

Note:

Remember that the central office serving your switching system may be emulating another country's network protocol. If so, you'll have to administer your circuit packs and trunk groups to match the protocol used by your central office.

Enhanced DS1 administration

Normally, you can't change the **DS1 Circuit Pack** screen unless you remove all related trunks from their trunk group. However, if the **DS1 MSP** field on the **System-Parameters Customer-Options** screen is **y**, and you are assigned the associated login permissions, you can change some of the fields on the **DS1 Circuit Pack** screen without removing the related trunks from their trunk group.

The following enhanced DS1 administration login permissions must be assigned on the **Command Permission Categories** screen:

- The **Maintain Enhanced DS1** field must be **y**.
- The **Maintain Trunks** field must be **y**.
- The **Maintain Switch Circuit Packs** field must be **y**.

If you busy out the DS1 circuit pack, you can change the following fields: **CRC**, **Connect**, **Country Protocol**, **Framing Mode**, **Interface**, **Interconnect**, **Line Coding**, and **Protocol Version**. After changing these fields, you may also have to change and resubmit associated screens.

Matching field settings on different screens

For enhanced DS1 administration, some field values on the **DS1 Circuit Pack** screen must be consistent with those on other screens as shown in the table below. If you change field values on the **DS1** screen, you must change the related fields on the other screens and resubmit them.

DS1 Circuit Pack field	Affected screens
Line Coding	**Route Pattern** **Access Endpoint** **Signaling Group** **Tone Generation**
Connect	**Signaling Group**
Protocol Version	**Signaling Group**
Interface	**Signaling Group**
Interconnect	**Tone Generation**
Country Protocol	**Signaling Group** **Tone Generation**

Specific combinations of settings for some of these fields are shown below.

ITC, Bit Rate, and Line Coding

The **ITC (Information Transfer Capability)** field appears on the **Route Pattern** screen, **Trunk Group** screen, and **Access Endpoint** screen. The **Line Coding** and the **Bit Rate** fields appear on the **DS1 Circuit Pack** screen. The settings for these fields on all the screens must be coordinated as shown in the following tables.

ITC field	Bit Rate	Line Coding field
restricted	1.544 Mbps	ami-zcs
	2.048 Mbps	ami-basic
unrestricted	1.544 Mbps	b8zs
	2.048 Mbps	hdb3

Interconnect and corresponding Group Type entries.

The **Interconnect** field appears on the **DS1 Circuit Pack** screen. The **Group Type** field appears on the **Trunk Group** screen. Set these fields as shown in the following table.

Interconnect field	Group Type field
co	co, did, diod, fx, or wats
pbx	access, aplt, isdn-pri, tandem, or tie

Related topics

See DS1 Circuit Pack on page 928 for information on administering DS1 service.

See "DS1 Trunk Service" in *Feature Description and Implementation for Avaya Communication Manager, 555-245-205,* for detailed information on DS1 service.

Adding Trunks to a Trunk Group

Use this procedure to add new trunks or to change the assignment of existing trunks. To change the assignment of existing trunks, remove them from their current trunk group and add them to the new group.

Before you start

You must add a trunk group before you can assign and administer individual trunks. To add a new trunk group, see the instructions in this chapter for the type of group you want to add.

Instructions

As an example, we will assign 5 trunks to a new tie trunk group, trunk group 5. We'll use ports on several circuit packs for members of this group.

To assign trunks to trunk group 5:

1. Type **`change trunk-group 5`** and press **Enter**.

   The **Trunk Group** screen appears.

2. Move to the **Group Member Assignments** page.

   Some of the fields on this screen won't appear for every trunk group.

Figure 180: Trunk Group screen

```
                                TRUNK GROUP
                                        Administered Members(min/max): xxx/yyy
                                           Total Administered Members: xxx
GROUP MEMBER ASSIGNMENTS
       Port     Code  Sfx  Name         Night          Mode     Type    Ans Delay
    1: 1B1501_ TN464   F   5211_______  _____          e&m_____ t1-comp ____
    2: 1B1502_ TN464   F   5212_______  _____          e&m_____ t1-comp ____
    3: 1B1601_ TN464   F   5213_______  _____          e&m_____ t1-comp ____
    4: 1B1602_ TN464   F   5214_______  _____          e&m_____ t1-comp ____
    5: 1B1701_ TN464   F   5215_______  _____          e&m_____ t1-comp ____
    6: _______            ___________  _____          ________ _______ ____
    7: _______            ___________  _____          ________ _______ ____
    8: _______            ___________  _____          ________ _______ ____
    9: _______            ___________  _____          ________ _______ ____
   10: _______            ___________  _____          ________ _______ ____
   11: _______            ___________  _____          ________ _______ ____
   12: _______            ___________  _____          ________ _______ ____
   13: _______            ___________  _____          ________ _______ ____
   14: _______            ___________  _____          ________ _______ ____
   15: _______            ___________  _____          ________ _______ ____
```

3. In the **Port** field in row 1, type **1B1501**.

   This field assigns the first member of the trunk group to a port on a circuit pack.

4. In the **Name** field in row 1, type **5211**.

   This is the extension assigned to this trunk. In general, type the circuit ID or telephone number for each trunk in this field. The information is helpful for tracking your system or troubleshooting problems. Update these fields whenever the information changes.

5. In the **Mode** field, type **e&m**.

   CAUTION:

   An entry in this field is only required for some circuit packs. Dip switch settings on the circuit pack control the signalling mode used on the trunk group, so the entry in the Mode field must correspond to the actual setting on the circuit pack.

6. In the **Type** field, type **t1-comp**.

   An entry in this field is only required for some circuit packs.

7. Repeat steps 3 to 6, as appropriate, for the remaining trunks.

   Notice that you can assign trunks in the same trunk group to ports on different circuit packs.

8. Press **Enter** to save your changes.

Removing Trunk Groups

There's more to removing a trunk group than just executing the remove trunk-group command. If you're using Automatic Route Selection (ARS), you must remove an outgoing or two-way trunk group from any route patterns that use it. If you've administered **Trunk-Group Night Service** buttons for the trunk group on any phones, those buttons must be removed or assigned to another trunk group.

Instructions

As an example, we will remove trunk group 5. This two-way group is used in ARS route pattern 2. In addition, a **Trunk-Group Night Service** button on extension 8410 points to this group.

To remove trunk group 5:

1. In the **Route Pattern** screen for route pattern 2, clear the entries for trunk group 5.

   If you're replacing trunk group 5 with another trunk group, just type the information for the new trunk group over the old entries. Remember to press **Enter** to save your changes.

2. In the **Station** screen for extension 8410, clear the entry in the **Button Assignments** field for the **Trunk-Group Night Service** button.

   Remember to press **Enter** to save your changes.

3. In the **Trunk Group** screen for trunk group 5, remove all member trunks from the group.

   See Adding Trunks to a Trunk Group on page 443 for instructions.

4. Type **`remove trunk-group 5`** and press **Enter**.

   The **Trunk Group** screen appears.

5. Press **Enter** to remove the trunk group.

Inserting and Absorbing Digits

Use this procedure to modify the incoming digit string on DID and tie trunks by inserting (adding) or absorbing (deleting) digits. You'll need to do this if the number of digits you receive doesn't match your dial plan.

Instructions

As an example, let us say you have a DID trunk group. It's group number is 5. Your service provider can only send 4 digits, but your dial plan defines 5-digit extensions beginning with 6:

1. Type **`change trunk-group 5`** and press **Enter**.

   The **Trunk Group** screen appears.

Figure 181: Trunk Group screen

```
                              TRUNK GROUP

Group Number: 5               Group Type: did             CDR Reports: y
  Group Name: Incoming calls         COR: 85         TN: 1        TAC: 105

                     Country: 1
                                Auth Code? n


TRUNK PARAMETERS
         Trunk Type: wink-start   Incoming Rotary Timeout(sec): 5
                                             Incoming Dial Type: tone
  Trunk Termination: rc                Disconnect Timing(msec): 500
    Digit Treatment: insertion                          Digits: 6
    Expected Digits: 4                        Sig Bit Inversion: none
Extended Loop Range? n      Trunk Gain: high    Drop Treatment: silence

 Disconnect Supervision - In? y
```

2. In the **Digit Treatment** field, type **insertion**.

   This field tells Communication Manager to add digits to the incoming digit string. These digits are always added at the beginning of the string.

3. In the **Digits** field, type **6**.

   For insertion, this field defines the specific digits to insert. Communication Manager will add a "6" to the front of the digit strings delivered with incoming calls. For example, if the central office delivers the string "4444," Avaya Communication Manager will change it to "64444," an extension that fits your dial plan.

4. In the **Expected Digits** field, type **4**.

   This field tells Communication Manager how many digits the central office sends.

   Note:

   The **Expected Digits** field does not appear on the screen for tie trunk groups.

5. Press **Enter** to save your changes.

If your service provider sends 7 digits but you only need 5, you need to absorb the first 2 digits in the digit string. To absorb digits:

1. Type **change trunk-group 5** and press **Enter**.

   The **Trunk Group** screen appears.

2. In the **Digit Treatment** field, type **absorption**.

   This field tells Communication Manager to remove digits from the incoming digit string. These digits are always removed from the beginning of the string.

3. In the **Digits** field, type **2**.

   For absorption, this field defines *how many* digits will be absorbed. Communication Manager will remove the first 2 digits from the digit strings delivered with incoming calls. For example, if the central office delivers the string "556-4444," Avaya Communication Manager will change it to "64444," an extension that fits your dial plan.

4. In the **Expected Digits** field, type **7**.

   This field tells Communication Manager how many digits the central office sends.

   Note:

   The **Expected Digits** field does not appear on the screen for tie trunk groups.

5. Press **Enter** to save your changes.

Related topics

See Adding a DID Trunk Group on page 429 for instructions on administering a DID trunk group.

See Adding a Tie or Access Trunk Group on page 435 for instructions on administering a tie trunk group.

Administering Trunks for Listed Directory Numbers

Listed directory numbers (LDN) are the telephone numbers given for an organization in public telephone directories. You can administer Avaya Communication Manager so that calls to different listed directory numbers go to the same attendant group. How you administer your system for LDN calls depends on whether the calls are coming in over DID and tie trunks or over CO and FX trunks.

Instructions

As an example, let us say that one attendant group answers calls for 3 different businesses, each with its own listed directory number:

- Company A — 855-2020
- Company B — 855-1000
- Company C — 855-1111

DID trunks and some tie trunks transmit part or all of the dialed digit string to Communication Manager. If you want these calls to different numbers to go to one attendant group, you must identify those numbers for Communication Manager on the Listed Directory Numbers screen.

We will take the 3 businesses listed above as an example. We will assume your server receives 4 digits from the central office on a DID trunk group and that you're not using Tenant Partitioning. To make these calls to different listed directory numbers terminate to your attendant group:

1. Type **`change listed-directory-numbers`** and press **Enter**.

   The **Listed Directory Numbers** screen appears.

Figure 182: Listed Directory Numbers screen

```
                                                    Page   1 of   2
                        LISTED DIRECTORY NUMBERS

                          Night Destination:

      Ext        Name                          TN
 1: 2020       Company A                     1
 2: 1000       Company B                     1
 3: 1111       Company C                     1
 4:                                          1
 5:                                          1
 6:                                          1
 7:                                          1
 8:                                          1
 9:                                          1
10:                                          1
```

2. In the **Ext 1** field, type **2020**.

   This is the LDN for Company A.

3. In the **Name** field, type **Company A**.

   This name will appear on the console display so the attendant knows which business the call is for and how to answer it.

4. Repeat steps 2 and 3 for the other two businesses.

   You can enter up to 20 different listed directory numbers on this screen.

5. Press **Enter** to save your changes.

To make LDN calls over a CO or FX trunk group terminate to an attendant group, you must type attd in the **Incoming Destination** field on the **Trunk Group** screen for that group.

When you use the **Listed Directory Number** screen to assign some extensions to the attendant group, or when you enter **attd** in the **Incoming Destination** field on the **Trunk Group** screen for CO or FX trunks, Avaya Communication Manager treats these calls as LDN calls.

Related topics

See Listed Directory Numbers on page 1249 for detailed information about this feature.

Administering Answer Detection

Use this procedure to administer an outgoing or two-way trunk group for network answer supervision or answer supervision by timeout. If your network supplies answer supervision to a trunk group, you can administer Avaya Communication Manager to recognize and respond to that signal. If your network does not supply answer supervision, you can set a timer for all calls on that group. When the timer expires, Communication Manager assumes the call has been answered and call detail recording starts (if you are using CDR).

For information about answer detection by call classification, contact your Avaya technical support representative or see "Answer Detection" in *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205 for an introduction.

Before you start

Determine whether the trunk group receives answer supervision from your service provider or private network. For example, most loop-start CO, FX, and WATS trunks do not provide answer supervision.

Instructions

As an example, we will administer trunk group 5 for both types of answer detection.

To administer trunk group 5 for answer supervision from the network:

1. On the **Trunk Group** screen for group 5, type **y** in the **Receive Answer Supervision** field.
2. Press **Enter** to save your change.

Now we will administer answer supervision by timeout. We'll set the timer to 15 seconds. To administer trunk group 5 for answer supervision by timeout:

1. On the **Trunk Group** screen for group 5, type **15** in the **Answer Supervision Timeout** field.
2. Press **Enter** to save your change.

Related topics

See :Answer Detection" in *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205*,* for detailed information about this feature.

Administering ISDN Trunk Groups

Integrated Services Digital Network (ISDN) trunk groups support the ISDN and Call-by-Call Service Selection service selection features. The trunk group provides end-to-end digital connectivity and supports a wide range of services including voice and non-voice services, to which users have access by a limited set of CCITT-defined, standard multipurpose interfaces.

The ISDN trunk group may contain ISDN-PRI or ISDN-BRI interfaces. However, it is not possible to use the two types of interfaces in the same trunk groups. The type of interface is chosen when the trunk members are assigned to the trunk group.

When ISDN-PRI interfaces are used on ISDN trunk groups, they may also be used to support the Wideband Switching feature. This is intended to work with the H0 (384 Kbps), H11 (1536 Kbps), H12 (1920 Kbps), and NXDS0 (128 to 1984 Kbps) data services, and to support high-speed video conferencing and data applications.

When an ISDN trunk connects two servers or switches, set the trunk options identically at both ends of the connection, with the exception of the **Trunk Hunt** fields. When ISDN-PRI interfaces are used, it is acceptable for both ends to have the **Trunk Hunt** fields administered as **cyclical**, but if one end is administered as **ascend**, the other end must be administered as **descend**. This helps avoid the possibility of glare conditions. When ISDN-BRI is used, the **Trunk Hunt** field has to be **cyclical**.

Hardware requirements

ISDN-BRI interfaces are supported by the TN2185 Trunk-side BRI circuit pack (for implementing the user side of the BRI interface) and by the TN556B or TN556C ISDN-BRI Line circuit pack or the TN2198 ISDN BRI (U-LT) Line circuit pack (for the network side of the BRI interface). The TN2464 circuit supports T1 and E1 digital facilities.

ISDN-PRI interfaces are supported by the TN767 circuit pack (for assignment of a T1 signaling link and up to 24 ISDN-PRI trunk group members), or the TN464C or later circuit pack (for assignment of a T1 or E1 signaling link and up to 24 or 31 ISDN-PRI trunk group members, respectively). The TN2464 and TN2207 circuit pack can also be used with ISDN-PRI.

- The D-channel for ISDN-PRI interfaces switches through either the TN765 Processor Interface (PI) circuit pack or the TN778 Packet Control (PACCON) circuit pack. The D-channel for ISDN-BRI interfaces only switches through the TN778 Packet Control (PACCON) circuit pack.

  Note:

  You cannot use the TN765 circuit pack with ISDN-BRI interfaces.

- For DEFINITY Server R configurations, the D-channel switches through the TN1655 Packet Interface (PKTINT) circuit pack.

- A TN780 or TN2182 Tone Clock circuit pack provides synchronization for the DS1 circuit pack.

  Note:

  The TN767 cannot be used to carry the D-channel if either the TN778 (PACCON) or TN1655 (PKTINT) circuit packs are used to switch the D-channel. However, in these circumstances, the TN767 can be used for NFAS interfaces carrying only B-channels.

Screens used to administer ISDN trunk groups

The table below shows the screens used to administer the TN765 Processor Interface (PI) for DEFINITY SI configurations.

Screen	Field
System-Parameters Customer-Options	**Version** **ISDN-PRI** **QSIG Optional Features**
Feature-Related System Parameters	**Send Non-ISDN Trunk Group Name as Connected Name** **Display Connected Name/Number for ISDN DCS Calls**
DS1 Circuit Pack	**All**
Signaling Group	**All**
Synchronization Plan	**All** (for more informationon this screen see Maintenance Procedures for Avaya Communication Manager 2.1, Media Gateways and Servers, 03-300192
Data Module	**All**
Processor Channel Assignment for DEFINITY SI	**All**
Interface Links	**All**
Trunk Group (ISDN)	**All**
ISDN Numbering - Public/Unknown	**All**

1 of 2

Screen	Field
ISDN Numbering - Private	**All**
Route Pattern	**All**
Hunt Groups	**ISDN Caller Display**
Terminating Extension Group	**ISDN Caller Display**

2 of 2

The table below shows the screens used to administer the TN778 Packet Control (PACCON) for an DEFINITY SI configurations.

Screen	Field
Feature-Related System Parameters	**Send Non-ISDN Trunk Group Name as Connected Name?** **Display Connected Name/ Number for ISDN DCS Calls?**
System-Parameters Customer-Options	**Version** **ISDN-BRI Trunks** **ISDN-PRI** **QSIG Optional Features**
Maintenance-Related System Parameters	**Packet Bus Maint**
Synchronization Plan	**All**
Trunk Group (ISDN)	**All**
ISDN Numbering - Public/Unknown	**All**
ISDN Numbering - Private	**All**
ISDN-BRI Circuit Pack Screen (if using ISDN-BRI interfaces) or **DS1 Circuit Pack Screen** (if using ISDN-PRI interfaces)	**All** **All**
Route Pattern	**All**
Hunt Groups	**ISDN Caller Display**

1 of 2

Screen	Field
Signaling Group (if using ISDN-PRI interfaces)	**All**
Terminating Extension Group	**ISDN Caller Display**
	2 of 2

The table below shows the screens used to administer the TN778 Packet Control (PACCON) for DEFINITY R, S8300 Media Server, S8700 or S8710 IP-Connect, and S8700 or S8710 Multi-Connect.

Screen	Field
Feature-Related System Parameters	**Send Non-ISDN Trunk Group Name as Connected Name?** **Display Connected Name/Number for ISDN DCS Calls?**
System-Parameters Customer-Options	**Version** **ISDN-BRI Trunks** **ISDN-PRI** **QSIG Optional Features**
Synchronization Plan	**All**
Trunk Group (ISDN)	**All**
ISDN Numbering - Public/ Unknown	**All**
ISDN-BRI Circuit Pack screen (if using ISDN-BRI interfaces) or **DS1 Circuit Pack screen** (if using ISDN-PRI interfaces)	**All** **All**
ISDN Numbering - Private	**All**
Route Pattern	**All**
Hunt Groups	**ISDN Caller Display**
Signaling Group (if using ISDN-PRI interfaces)	**All**
Terminating Extension Group	**ISDN Caller Display**

Table Notes

- **System-Parameters Customer-Options** — The **ISDN-BRI Trunks** or **ISDN-PRI** fields must be set to **y**. For a TN778 and if using ISDN-PRI interfaces, the **PRI Over PACCON** field must be set to **y**. If not, a new license file is needed. **QSIG Optional Features** fields may be enabled to allow appropriate administration for Supplementary Service Protocol.
- **Feature-Related System-Parameters** — Set the **Send Non-ISDN Trunk Group Name** as **Connected Name** and **Display Connected Name/Number for ISDN DCS Calls** fields.
- **ISDN-BRI Trunk Circuit Pack** — This screen is required if using ISDN-BRI trunk interfaces. Assign all fields as required.
- **DS1 Circuit Pack** — This screen is required if using ISDN-PRI interfaces.
  - DS1 (T1) Circuit Pack

    Assign all fields as required. For **Facility Associated Signaling**, up to 23 ports are available for administration as trunk members in an associated ISDN-PRI trunk group. The 24th port is used as a signaling channel. For **Non-Facility Associated Signaling**, all 24 ports may be used on certain DS1 circuit packs. The D-channel signaling function for these packs must be provided by a designated DS1 pack on its 24th channel.
  - E1 Circuit Pack

    Assign all fields as required. For **Facility Associated Signaling**, up to 30 ports are available for administration as trunk members in an associated ISDN-PRI trunk group. Port number 16 is used as a signaling channel.
- **Maintenance-Related System-Parameters** — Use this screen only for a TN778. Set the **Packet Bus Maint** field to **y**.
- **ISDN Trunk Group** — Enter information in all the fields except the trunk group members. When using ISDN-PRI interfaces, enter the members after you establish the signaling links.
- **Signaling Group** — This screen is required if ISDN-PRI interfaces are used. Complete all fields. This screen identifies groups of ISDN-PRI DS1 interface B-channels for which a given D-channel (or D-channel pair) will carry the associated signaling information (supports the Facility and Non-Facility Associated Signaling feature). Each DS1 board that is required to have a D-channel must be in a different signaling group by itself (unless D-channel backup is needed, in which case a second DS1 is administered as a backup D-channel). You are not required to select a channel for a trunk group, but if you do, you must have already defined the trunk group as type ISDN.

  Note:

  The following three screens, **Processor Interface Data Module**, **Communication Interface Links**, and **Communication Processor Channel Assignment** are used only to support the ISDN-PRI interfaces using PI TN765.

- **Processor Interface Data Module** — Use this screen only for a TN765. Assign up to 8 interface links using 8 **Processor Interface Data Module** screens for multi-carrier cabinet systems, and up to 4 links for single-carrier cabinet systems. One **Processor Interface Data Module** screen must be completed for each interface link to be assigned.
- **Communication Interface Links** — Use this screen only for a TN765. Assign link numbers 01 to 08 for a multi-carrier cabinet system or links 01 to 04 for a single-carrier cabinet system as required. When first administering this screen for ISDN in Avaya Communication Manager, do not administer the **Enable** field.
- **Communication Processor Channel Assignment** — Use this screen only for a TN765. Enter assigned link numbers and assign associated channel numbers to each link. Complete all fields of the screen as required. When first administering this screen for ISDN in Avaya Communication Manager, you need to:
  - First, administer the **Interface Links** screen, except the **Enable** field.
  - Second, administer the **ISDN** fields on the **Processor Channel** screen.
  - Last, go back to the **Interface Links** screen and administer the **Enable** field.
- **ISDN Numbering - Public/Unknown** — Complete all fields. This screen supports the ISDN Call Identification Display.
- **ISDN Numbering - Private** — Complete all fields. This screen supports the ISDN Call Identification Display.
- **Routing Pattern** — Complete all fields including the **Supplemental ISDN Routing Information** fields as required.
- **Hunt Group** — Complete the **ISDN Caller Display** field by entering either **grp-name** or **mbr-name** to specify whether the hunt group name or member name, respectively, is sent to the originating user (supports the ISDN Call Identification Display feature).
- **Terminating Extension Group** — Complete the **ISDN Caller Display** field by entering either **grp-name** or **mbr-name** to specify whether the group name or member name, respectively, is sent to the originating user (supports the ISDN Call Identification Display feature).
- **Synchronization Plan** — Assigns primary and secondary external synchronization sources for the ISDN-BRI Trunk or DS1 circuit pack. Complete all screen fields as required.

  Note:

  ISDN-BRI and ISDN-PRI interfaces cannot be mixed in the same trunk group. Therefore, consider the following:

  - The earliest trunk member (the lowest numbered one) administered is considered correct.
  - If an offending member is subsequently found (meaning the first member was BRI and a later member was PRI, or vice versa), the cursor positions on the offending member, and the following error message appears: **You cannot mix BRI and PRI ports in the same trunk group.**

Chapter 13: Managing Announcements

Understanding Announcements

An announcement is a recorded message a caller may hear while the call is in a queue, or if a call receives intercept treatment for some reason. An announcement is often used in conjunction with music.

Announcements can be integrated or external. Integrated announcements reside on a circuit pack in the carrier. External announcements are stored and played back from adjunct equipment. For more information on external announcements, see *Avaya Communication Manager Contact Center Guide to ACD Contact Centers.*

Adding Announcement Data Modules

Note:

This task is only required if you are using the TN750-series announcement circuit pack.

Your system uses a data module to save and restore the announcements from the integrated announcement circuit pack and your system's memory. You need to set up the data module that is built into the announcement circuit pack.

Before you start

You need to record your announcement on a special announcement circuit pack on your system, or on an external device like an answering machine. For circuit pack information, see *Hardware Guide for Avaya Communication Manager,* 555-245-207.

Instructions

To set up the announcement data module, determine the port location of the Announcement circuit pack. You can find circuit pack information on the Integrated Announcement Board screen.

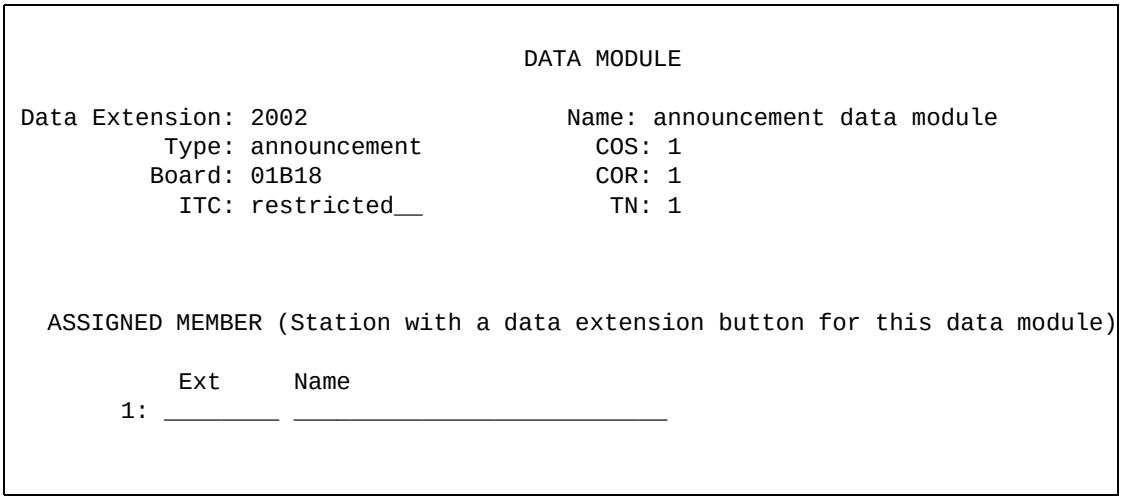
To set up the data module on the Announcement circuit pack located at 01B18:

1. Type **add data-module next** and press **Enter**.

   The **Data modules** screen appears.

   The system automatically fills in the number of the next available extension. Make sure the extension conforms to your dial plan. If you want to assign a specific extension, type that extension in the command instead of "next."

Figure 183: Data Module screen



```
                                DATA MODULE

Data Extension: 2002                  Name: announcement data module
          Type: announcement            COS: 1
         Board: 01B18                   COR: 1
           ITC: restricted__             TN: 1



  ASSIGNED MEMBER (Station with a data extension button for this data module)

           Ext     Name
      1: ________ ____________________________
```

2. In the **Name** field, type **announcement data module**.
3. In the **Type** field, type announcement and press **Enter**.

   The **Port** field automatically changes to **Board**.
4. In the **Board** field, type **01B18**.

   This is the address of the announcement circuit pack.
5. Press **Enter** to save your changes.

Adding Announcement Extensions

You need to assign an extension for each announcement that you want to record. After you define an announcement extension, you use it to record and access the announcement.

Announcements can be stored on a TN2501AP circuit pack or a TN750 circuit pack.

Instructions

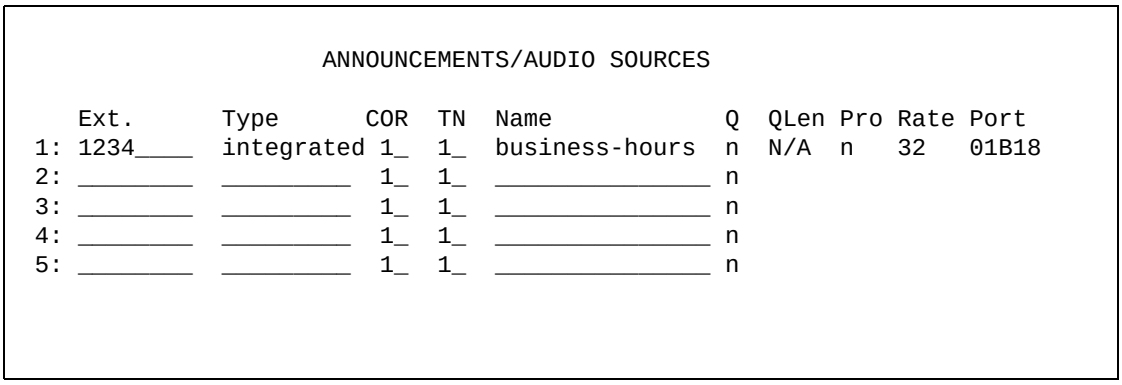
We will use extension 1234 for an announcement about business hours. We will use the integrated announcement circuit pack located on 01B18 (for VVAL, we use 12V9 if referring to gateway 12).

To add an announcement extension 1234:

1. Type **change announcements** and press **Enter**.

   The **Announcements/Audio Sources** screen appears.

Figure 184: Announcements/Audio Sources screen



```
                    ANNOUNCEMENTS/AUDIO SOURCES

   Ext.      Type      COR  TN  Name            Q  QLen Pro Rate Port
1: 1234____  integrated 1_  1_  business-hours  n  N/A  n   32   01B18
2: _________ __________  1_  1_  ________________ n
3: _________ __________  1_  1_  ________________ n
4: _________ __________  1_  1_  ________________ n
5: _________ __________  1_  1_  ________________ n
```

2. In the **Ext** field, type **1234**.
3. In the **Type** field, type **integrated**.
4. In the **Name** field, type **business hours**.

Note:

If you are using either the TN2501AP or the G700 Media Gateway VVAL, then you must enter a name in the **Name** field. This name becomes the filename for the announcement file. See Announcements/Audio Sources on page 750 for more information on requirements for this field.

5. In the **Q** field, type **y**.

   This sets up an announcement queue so calls wait in a queue to hear an announcement if all the ports on the announcement circuit pack are busy.

   If you leave this field **n**, there is no announcement queue. When all the announcement ports are busy the caller hears busy or other feedback, depending upon how the announcement was accessed.

6. **N/A** appears in the **QLen** (queue length) field. You cannot change this field for integrated announcements because they have a pre-set queue length.
7. In the **Pro** (protected) field, type **n**.

   If you enter **n**, users with console permissions can change the announcement. If you enter **y**, the announcement cannot be changed.

   This field appears only if the **Type** field is integrated.

8. In the **Port** field, type:
   - the location of the announcement circuit pack (TN2501AP or TN750). In this example, we use **01B18**.
   - **ggv9** for G700 VVAL (where **gg** is the gateway number of the G700 Media Gateway).
9. Repeat steps 2 -8 with the correct information for each announcement you want to record.
10. Press **Enter** to save your work.

    Note:

    These steps have only created the administered name for the announcement file. You fill the file space when you record an announcement or transfer an announcement file to the circuit pack through a file transfer protocol (FTP) session.

To check that the announcement administration is correct:

1. At the SAT, type **`list integrated-annc-boards`** and press **Enter**.

Recording Announcements

You can record an announcement for callers to hear when they dial a specific extension. You can use the same steps to change an existing announcement.

For instructions on how to record an announcement at a computer or from a telephone, to a TN2501AP, see Recording VAL announcements at a system telephone.

If you want to change an announcement that is recorded on a circuit pack without memory, you have to delete the original message and record a new announcement. See Deleting and Erasing Announcements on page 467.

Announcement sessions always use port 0, which also is used for playing announcements, on the TN750 circuit packs. Announcement sessions always use port 1, which is dedicated for telephone access, on the TN2501AP circuit packs. With the TN2501AP circuit pack or VVAL, the port is only busy if another telephone access session is active to the same circuit pack.

To begin an announcement session, the user must dial the administered feature access code (FAC) followed by the announcement session. If an announcement session is already in progress, or if a save or restore command is in progress, then the user hears reorder tone (fast busy) and the system drops the call.

If the telephone session port to an integrated circuit pack is in use, then the user hears reorder tone followed by silence. This indicates that the port will be reserved for an announcement session. The user should redial the FAC and extension every 45 seconds to gain access to the port.

Note:

For the 1.1 and later releases of Avaya Communication Manager, multiple telephone sessions are allowed, with one session associated with each active integrated announcement circuit pack.

Once an end user accesses an announcement session, the user can dial **1** to record an announcement, **2** to play an announcement, or **3** to delete the announcement. If the circuit pack memory is more than 90% full, then Communication Manager gives stutter dial tone when the user gains access to an announcement session. Even if the user hears stutter tone, the user should begin speaking to record the announcement.

Before you start

- You need to have a telephone or console with a class of service (COS) that provides console permissions to record announcements.

  See Command Permission Categories on page 835 for more information on COS permissions.

Instructions

To record an announcement to extension 1234, or change the announcement already recorded there, we'll use a telephone with console permissions. In our example, the announcement feature access code is *56.

To record or change the announcement:

1. Dial ***56** from a telephone or console.
   - If you hear dial tone, go to step 2.
   - If you hear a fast busy signal, hang up and redial the FAC and extension every 45 seconds until you hear dial tone.
2. Dial the announcement extension **1234**.

   You hear dial tone.
3. Dial **1** to begin recording.
   - If you hear a beep or stutter tone, begin speaking. If the circuit pack memory becomes full during recording, the system drops your connection and does not retain the announcement.
   - If you hear intercept tone, hang up and record your announcement on another extension that is assigned to a different circuit pack.
4. To end the recording:
   - If you are using a digital telephone, press #. You hear dial tone allowing you to continue your session (for example, dial 2 to hear the announcement just recorded).
   - If you are using an analog telephone, hang up. If your analog telephone is not connected through lineside DS1, the system records an electrical click at the end of the recording. You have to redial the announcement feature access code to continue your session.
5. To listen to the announcement you just recorded:
   - If you are using a digital telephone, do not hang up. Dial 2. The recording plays back through the handset.
6. If you are not satisfied with the announcement,
   - press **1** to re-record the announcement.
   - press **3** to delete the announcement and end the recording session.
7. If you want to listen to the announcement after you have hung up, dial the extension from any telephone or console. In this example, dial **1234**. The announcement plays through the handset. When dialing announcement extensions directly, if the announcement does not exist (i.e., has not been recorded or FTP'd to the board), the caller hears silence instead of a busy tone with the VAL type sources (TN2501 or G700 Gateway). With TN750 boards, a busy tone is provided in this case.

Note:

You have to wait 15 seconds after you record the announcement before you can dial the extension to hear your announcement. During this 15-second window, you cannot record a new announcement and no one can play this announcement. You can re-record the announcement. Dial the feature access code, dial the extension, and press **2** before the 15-second timer expires.

Saving Announcements

Note:

This task only applies if you are using a TN750-series announcement circuit pack.

If you are using the TN2501AP (VAL) announcement circuit pack or VVAL announcements, see Recording new VAL announcements at a computer on page 474.

You can save and back-up announcements from an announcement circuit pack to system memory. Your system memory is a tape, disk, or memory card, depending on your system. Use this procedure primarily for announcement circuit packs without built-in memory. VAL and VVAL announcements are backed up internally to non-volatile flash memory.

CAUTION:

Announcements that are recorded on announcement circuit packs without built-in memory are lost if they are not saved to system memory before power is shut down or the circuit pack is reset or removed.

For extra security, you may also want to save announcements from announcement circuit packs that have their own built-in memory to another announcement circuit pack with built-in memory or to tape.

CAUTION:

Do not copy, save, or restore announcements from an announcement circuit pack that has built-in memory to one without built-in memory because it may corrupt the announcement data.

Instructions

To save or backup announcements when you have only one announcement circuit pack:

1. Type **save announcements** and press **Enter** to save the changes.

   This process can take up to 40 minutes. You cannot administer your system while it is saving announcements.

To save or backup announcements when you have more than one announcement circuit pack, specify the circuit pack address in your command.

For example to save the announcements from the circuit pack at 01B18:

1. Type **save announcements from 01B18** and press **Enter**.

   Note:

   If you have announcement circuit packs with and without built-in memory, save the circuit pack without built-in memory.

If you have a duplicated system with an active processor and a stand-by processor, you can save announcements from the announcement circuit pack to both processors. Simply add system memory for both processors to the save announcements command.

If the save announcements procedure fails on one of the processors, it continues on the other processor, which results in inconsistent announcement data between the two processors.

To verify that the save announcements command was successful:

1. Type **display integrated announcement boards** and press **Enter**.

   The **Integrated Announcement Boards** screen appears.

This screen shows the date, time, and location of the announcement circuit pack most recently saved. If the save announcement command failed or the save translations were not run before rebooting, **N/A** appears in these fields.

Copying Announcements

Note:

This task only applies if you are using a TN750-series announcement circuit pack.

If you are using the TN2501AP (VAL) announcement circuit pack or VVAL announcements, see Moving announcements from the VAL circuit pack on page 480.

You can copy recorded announcements between back-up disks and tapes when you want to store your recorded announcements in more than one place.

Instructions

We will copy the announcements stored in our weekly disk back-up to the tape back-up we take off company premises for a month. To copy announcements between your back-up disk and tape:

1. Type **copy announcements** and press **Enter** to save the changes.

   This process can take up to 40 minutes. You cannot administer your system while it is copying announcements.

Restoring Announcements

Note:

This task only applies if you are using a TN750-series announcement circuit pack.

If you are using the TN2501AP (VAL) announcement circuit pack or VVAL announcements, see Moving announcements to a VAL circuit pack or to another LAN device on page 482.

You can restore announcements from system memory to an announcement circuit pack. Your system memory is a tape, disk, or memory card, depending on your system.

If you have a duplicated system, you system always restores the announcements located on the active processor.

Instructions

We will restore announcements from our system memory (stored on a disk) to the integrated announcement circuit pack on our system.

To restore announcements from system memory to the integrated announcement circuit pack:

1. Type **`restore announcements disk`** and press **Enter**.

We will restore announcements from system memory to an announcement circuit pack that has built-in memory. We know that the announcement circuit pack is located at 01B18:

1. Type **`restore announcements from cabinet 01. carrier B, slot 18`** and press **Enter**.
2. Press **Enter** to restore announcements.

Fixing problems

Problem	Possible causes and solutions
You receive error code E28	You do not have an integrated announcement circuit pack in the system.
You receive error code E31	A call is connected to the announcement on the circuit pack and the port is busy. Wait for the call to disconnect and try restore announcements again.

If the system crashes or there is a processor interchange during restore announcements, the restore fails and there is no valid announcement on the circuit pack. Repeat the process when the system is operating properly.

Deleting and Erasing Announcements

You can use a system telephone to delete recorded announcements from an announcement circuit pack that does not have stored memory. When you delete the announcement from the circuit pack, you do not delete the announcement from your system backup tape or disk.

You can also erase the announcements stored on an integrated announcement circuit pack that has stored memory.

Note:

The system denies any attempt to delete an announcement while it is playing, being transferred, or being backed up to FLASH (amber LED is flashing), regardless of whether the attempt is from a system telephone, the SAT, or through an FTP session.

Before you start

- Look up the announcement extension, which is mapped to a specific TN2501AP circuit pack or VVAL gateway by location.
  1. At the SAT, type **`list integrated-annc-boards`** and press **Enter**.
  2. Determine the extension(s) for the announcement(s) that you want to delete.
- Record the feature access code (FAC) for an announcement session.

Instructions

We will use a telephone with console permissions to delete the recorded announcement assigned to extension 1234. In our example, we know that the announcement feature access code is *56.

1. Dial ***56** from a telephone or console.

   You hear dial tone.

2. Dial **1234**.

   You hear dial tone.

3. Dial **3** to delete the announcement from the announcement circuit pack.

   You hear a confirmation tone.

   If the announcement is protected or is playing at the time of the command, you hear a fast busy signal (reorder tone) and the system does not delete the announcement.

4. Hang up the telephone.
5. To ensure that an announcement was deleted, dial **1234**.

   You hear a busy signal if the announcement was deleted.

6. Repeat Steps 1-5 for each announcement that you want to delete.

   You can delete only one announcement at a time.

You may also want to remove the announcement extension from the system. To remove the extension, use your system administration terminal to complete the following steps:

1. Type **change announcements** and press **Enter**.

   The **Announcements/Audio Sources** screen appears.

2. Delete the information in the **Ext** and **Type** fields.
3. Press **Enter** to save your work.

Finally, to erase announcements on the announcement circuit pack located at 01B18:

1. Type **erase announcements 01B18**.

   A warning message appears.

2. Press **Enter** to erase the announcements on the circuit pack.

Setting up Continuous-Play Announcements

You can set up announcements to keep on repeating while callers are connected to the announcement, so a caller listens until the system plays the entire announcement.With a "barge-in" queue, you do not need a separate port for each announcement.

For example, you can set up an Automatic Wakeup announcement to repeat and use a barge-in queue. When guests pick up the telephone to hear an announcement at a particular time, they use only one port and the message repeats on that port until the last guest goes off-hook and the message ends.

Before you start

You must use an integrated, multiple-integrated, or external announcement for barge-in announcements.

Instructions

We will set up a continuous-play announcement for our integrated announcement on extension 1234:

1. Type **change announcements** and press **Enter**.

   The **Announcements/Audio Sources** screen appears.

Figure 185: Announcements/Audio Sources screen

```
                    ANNOUNCEMENTS/AUDIO SOURCES
   Ext    .   Type       COR  TN  Name            Q  QLen Pro Rate Port
1: 1234____  integ-rep   1_   1_  business hours   b  N/A  n   32   01B18
2: _________ _________   1_   1_  _______________ n
3: _________ _________   1_   1_  _______________ n
4: _________ _________   1_   1_  _______________ n
5: _________ _________   1_   1_  _______________ n
6
```

2. Type **b** in the **Q** field on the same line as extension 1234.
3. Leave **business hours** in the **Name** field, or enter a new description for the announcement.
4. Press **Enter** to save your work.

Getting Started with the TN2501AP or G700 Virtual VAL

Before you can use the capabilities of the TN2501AP or G700 VVAL announcement circuit pack, it must be properly installed and configured. These instructions are contained in other documents in the Avaya Communication Manager documentation library. For more information about these and other tasks related to using the TN2501AP, see the documents listed in the following table.

Task	Information source
Installing the TN2501AP Administering IP Connections Adding IP Routes Testing the IP Connections	*Made Easy Tool for DEFINITY Media Server Configurations* *Installation, Upgrades and Additions for the Avaya CMC1 Media Gateway*
Installing VVAL for a G700 Media Gateway using the Media-Gateway screen and the **`enable announcement`** command Administering IP Connections Adding IP Routes Testing the IP Connections **Note:** G700 Media Gateway embedded VAL announcements (VVAL) must have the gateway(s) that will provide announcements enabled in order for announcement extensions assigned to that gateway to be played.	Each G700 Media Gateway that will be used to provide announcements through the embedded VAL circuitry on the Gateway processor circuit pack must be assigned on the **Media-Gateway** screen and enabled using the **`enable announcements`** command before announcements can be recorded using the telephone or played from that gateway. Announcements can be administered to a gateway and files can be FTPed to that gateway even though it is not enabled. However, the G700 Media Gateway first must be assigned on the **Media-Gateway** screen so as to be used for gateway announcements. Each G700 Media Gateway when enabled is counted as a VAL circuit pack towards the system limit of either 1 VAL circuit pack (if the **VAL Maximum Capacity** field is **n**) or 10 circuit packs (for the S8700, S8710, or S8300 Media Servers) if the **VAL Maximum Capacity** field is **y**. First the G700 Media Gateway must have the **V9** field assigned to **gateway-announcements** on the **Media-Gateway** screen before the G700 embedded VAL (VVAL) can be enabled. Then the G700 Media Gateway embedded VAL is enabled using the **`enable announcement-board ggV9`** command (where **gg** is the gateway number assigned on the **Media-Gateway** screen).

1 of 2

Task	Information source
	The G700 Media Gateway embedded VAL also can be disabled using the **`disable announcement-board ggV9`** command. This removes that gateway from the VAL circuit pack count but announcements already assigned and recorded/FTPed on that circuit pack remain but will not play.
Administering Announcements (recording, copying, deleting, etc.)	Managing VAL Announcements Using the SAT on page 472 and Managing VAL Announcements Using FTP on page 478.
Viewing announcement usage measurements (**`list measurements announcement`** command)	*Reports for Avaya Communication Manager.*
Troubleshooting announcements	Troubleshooting VAL Announcements on page 485.
Troubleshooting VAL hardware	*Maintenance for Avaya Communication Manager and DEFINITY Server* for your model(s).
	2 of 2

Managing VAL Announcements Using the SAT

These are the tasks that you can perform from a System Access Terminal (SAT):

- Administering an announcement using the SAT
- Recording announcements (with an option to use a system telephone)
- Deleting announcements (with an option to use a system telephone)

To administer an announcement using the SAT, see Adding Announcement Extensions on page 459.

Recording VAL announcements

You can record an announcement for callers to hear when they dial a specific extension or as part of call vectoring. You can use the same steps to change an existing announcement.

You can record announcements in many ways:

- Professional or computer recordings
- Recording new announcements at a computer
- Recording announcements at a system telephone

Before you start

Ensure that the announcement administration is complete before proceeding. You must have assigned a name before you can record an announcement (see Adding Announcement Extensions on page 459).

If you are replacing a TN750C announcement circuit pack with the TN2501AP,

- get a list and description of the announcements stored on the TN750C circuit pack.
- re-record the announcements on a computer or at a professional recording studio as .wav files (CCITT µ-Law or A-Law, 8KHz, 8-bit mono), so that they are ready to transfer to the new announcement circuit pack after it is installed and administered.

Replacing old announcement circuit packs with the new TN2501AP circuit pack requires that you

- remove previous announcement administration
- record new announcements for the TN2501AP
- re-record any announcements currently resident on the TN750 circuit packs that you are replacing. You cannot transfer or restore TN750 announcements from flash card, tape, or optical disk to the TN2501AP.

CAUTION:

When you change a VAL announcement, the TN2501AP is activated to perform a FLASH auto-save backup 5 minutes later. If you try to record a new announcement during this backup process, the new recording fails, and a denial event is logged to the Event Report screen. For information on viewing denial events, see Viewing the Denial Events Log on page 485.

Announcement file format requirements

In order to be compatible with the TN2501AP circuit pack and Avaya Communication Manager, announcement recordings must have the following parameters:

- CCITT A-Law or CCITT µ-Law companding format (do not use PCM)
- 8KHz sample rate
- 8-bit resolution (bits per sample)
- Mono (channels = 1)
- The µ-Law (Mu-Law) companding is used in the United States and A-Law is used internationally. Use the companding format specified on the **Location Parameters** screen.
- File names for the TN2501 (VAL) boards cannot contain blank spaces nor any of the following characters: ‘.’, ‘,’, ‘/’,’:’, ‘*’,’?’, ‘<’, ‘>’, ‘\’. The file name itself is case sensitive, but the *.wav* file extension must be lower case.

Announcements that are recorded in this format occupy 8K bytes per second of file space for every second of recorded speech. For example, a 10-second announcement creates an 80K bytes .wav file.

Professional or computer recordings

In order to be compatible with the TN2501AP circuit pack and Avaya Communication Manager, announcement recordings must meet the Announcement file format requirements described above.

Recording new VAL announcements at a computer

To record an announcement at a computer:

1. At the computer, open the application that you use to record .wav files.
2. Set the recording parameters.
3. Record the announcement by speaking into a microphone connected to the computer.
4. Play the announcement back at the computer before transferring the file to the VAL circuit pack.

Recording VAL announcements at a system telephone

To record an announcement at a system telephone, see Recording Announcements.

Converting announcement files to VAL format

If you are sharing recordings in a multi-site environment with Avaya Communication Manager and CONVERSANT systems, you must convert announcement files for use on either system. If you need to convert an announcement file to the required format, you can use a sound recording utility application to do so.

To convert a previously-recorded announcement, or a file compatible with a CONVERSANT system, to a format compatible with Avaya Communication Manager:

1. Open the sound recording application on your computer (for example, Microsoft Windows Sound Recorder).
2. Open the file you want to convert.
3. Check the file properties to see if you need to change the parameters.
4. If you need to change the recording parameters, look for a conversion tool (some have a **Convert Now** option, others use **Save As**).
5. Change the file parameters to those listed in Announcement file format requirements.

   Note:

   In some applications, assigning the format (for example, CCITT µ-Law) sets the remainder of the default parameters. Check each parameter carefully, perhaps changing default settings to match the parameters listed above. CCITT µ-Law or A-Law can be referred to as ITU G.711 µ-Law or ITU G.711 A-Law, respectively.

Deleting VAL announcements

Before you start

Look up the announcement information:

1. At the SAT, type **list directory board** and press **Enter**.
2. Determine which announcement(s) that you want to delete, either by extension or filename.
3. Decide whether you are
   - Deleting individual VAL announcement files using the SAT
   - Deleting all VAL announcements on a circuit pack using the SAT
   - Deleting and Erasing Announcements

Note:

The system denies any attempt to delete an announcement while it is playing, being transferred, or backed up to FLASH (amber LED flashes), regardless of whether the attempt is from a system telephone, the SAT, or through an FTP session.

Deleting individual VAL announcement files using the SAT

To delete the announcement named Closed:

1. At the SAT, type **remove file board board-location /annc/filename.wav** and press **Enter**.

   For example, to delete the announcement, type:

   remove file board 01A11 /annc/Closed.wav

   Note:

   Filenames are case-sensitive and require the .wav file extension.

   The **/annc** portion of the command directs the system to the announcement subdirectory on the VAL circuit pack, and **/Closed.wav** indicates to delete the filename "Closed.wav."

Deleting all VAL announcements on a circuit pack using the SAT

To delete all of the announcement files on the VAL circuit pack:

1. At the SAT, type **busyout board board-location** and press **Enter**.

   Ensure that the command is successful.

   Note:

   When the VAL board is busied out,

   - both the RSCL and ethernet ports are busied out.
   - firmware takes down the ethernet link.
   - FTP is disabled because the ethernet link is down.
   - announcements on that circuit pack cannot play.

2. At the SAT, type **erase announcements board board-location** and press **Enter**.

   CAUTION:

   This command deletes the specified announcement file in both RAM and FLASH memory. The board firmware ignores the protect flag (Pro field) when erasing the announcement files.

3. At the SAT, type **list directory board** and press **Enter**.
4. Verify that there are no files listed.

   Note:

   The announcement directory on the TN2501AP or G700 Media Gateway is **/annc**.

5. Type **list integrated-annc-boards** and press **Enter**.

   Check the list to see that the announcement was deleted. The Length in Seconds field should show 0.

Deleting VAL announcements at a system telephone

See Deleting and Erasing Announcements.

Changing integrated announcement board locations

You can change announcement boards that are currently administered on the **Announcements/Audio Sources** screen to a new board location.

To change an announcement:

1. At the SAT, type **change integ-annc-brd-loc** and press **Enter**.
   The **Change Integrated Announcement Translations** screen appears.
2. In the **Change all board location translations from board** field, type the board location currently administered on the **Announcement/Audio Sources** screen.
3. In the **to** field, type the board location to which you want to move the integrated announcement translations.
4. Press **Enter** to save your changes.

Displaying integrated announcement boards

You can display announcement boards that are currently administered on the **Announcements/Audio Sources** screen.

To display integrated announcement boards:

1. At the SAT, type **display integrated-annc-boards** and press **Enter**.
   The **Integrated Announcement Boards** screen appears.

Figure 186: Integrated Announcement Boards screen

```
display integrated-annc-boards                                  Page 1 of x

    Last Board Loction Saved:

  Board                 Time                    Number of
  Location       Sfx    Remaining     Rate      Recordings        Checksum ID

1:01D09          C      206           32              7           047a
2:02817          C      248           32              1           00db
3:
4:
5:
```

Managing VAL Announcements Using FTP

This section includes information on setting up and terminating a file transfer protocol (FTP) session and outlines tasks that you can do in an FTP session.

There are 3 basic components to an FTP session:

- Setting up an FTP session
- Performing tasks in an FTP session
- Ending an FTP session

SECURITY ALERT:
Be sure to read and observe all of the Security Alerts regarding enabling and disabling the VAL (TN2501AP or embedded G700 Gateway circuit packs) filesystem and FTP sessions into it.

Setting up an FTP session

Setting up an FTP session into the VAL circuit pack involves:

1. Preparing the VAL circuit pack for the FTP session, which
   - allows an FTP session on an individual VAL circuit pack.
   - creates an ftp-login and ftp-password for that session.
2. Starting an FTP session from a computer or network management terminal. Before you can start the FTP session, you need to know
   - the VAL circuit pack's IP address from Step 1.
   - the VAL circuit pack's ftp-login and ftp-password from Step 1.

Preparing the VAL circuit pack for the FTP session

To prepare the VAL circuit pack for the FTP session, including setting the username and password:

1. At the SAT, type **enable filesystem board board-location login ftp-username [3-6 characters] ftp-password [7-11 characters]** and press **Enter**.

   For example, the command:

   enable filesystem board 01A11 login romeo shakespeare

   enables an FTP session into the VAL circuit pack in Cabinet 1, carrier A, slot 11. The ftp-username (3-6 characters) for this session is **romeo**, and the ftp-password (7-11 characters) is **shakespeare**.

   When the FTP session on the circuit pack is enabled, the announcement and firmware files are available to anyone who knows the VAL circuit pack's IP address, the ftp-username, and the ftp-password.

   SECURITY ALERT:
   Avaya recommends using a unique ftp-login and ftp-password for each FTP session.

Starting an FTP session

If you are unfamiliar with FTP client application software, contact your network administrator for information about access to an FTP session.

The following points apply to FTP sessions into the VAL circuit pack:

- In FTP sessions, filenames are case-sensitive and require the .wav" file extension.
- Only one FTP session can be active at a time. If an FTP session is already active for a particular VAL circuit pack, the system denies a second attempt to establish an FTP session from some other remote host.
- The VAL circuit pack has two user-accessible directories:
  - /annc for playable announcements
  - / (root) for temporary storage of embedded software updates. Use this directory only for software updates.
- FTP sessions time out after 10 minutes of inactivity.

Instructions

1. At the FTP client, type **ftp val-ip-address** and press **Enter**. The IP address must match the administered IP address (see **change node-names ip**).
2. At the username prompt, type **romeo** and press **Enter**.

3. At the password prompt, type **shakespeare** and press **Enter**.

   The system responds with **User logged in**.

   Note:

   Once you are logged in you are in the announcements directory (/annc).

4. If you are moving files to or from the VAL circuit pack, you must set the system to binary mode. At the FTP client, type **bin** and press **Enter**.

   The system responds with **Types** set to **I**, binary mode.

   CAUTION:

   If you do not transfer announcement files in binary mode, they can be corrupted and the FTP session can fail.

Performing tasks in an FTP session

You are now ready to perform any of these tasks in the FTP session

- Moving announcements from the VAL circuit pack
- Deleting announcements
- Moving announcements to a VAL circuit pack or to another LAN device
- Combining tasks

Moving announcements from the VAL circuit pack

When you move a file from the VAL circuit pack, you are either

- backing up (archiving) an announcement file.
- copying an announcement to another VAL circuit pack (restoring).

Moving a file in an FTP session means copying the file from the VAL circuit pack to the FTP client's default directory. If you want to move the file to another circuit pack or LAN device, see Moving announcements to a VAL circuit pack or to another LAN device.

Before you start

- Ensure that the steps in Setting up an FTP session are complete.
- Ensure that you have adequate storage space for backup on the remote host computer. Depending on the number of announcements and file type, 60 minutes of recording may require up to 32 megabytes of storage space for backup on the remote host computer.
- Know the IP address and location of the TN2501AP circuit pack as well as the filename (**`list directory board`**) for the announcement that you want to move.

  Note:

  The announcement directory on the TN2501AP circuit pack is **`/annc`**.

Instructions

To backup or save an announcement from the VAL board to the client computer through an FTP session:

1. Ensure that the steps in Setting up an FTP session are complete.
2. At the FTP client, type **`get filename.wav`** and press **Enter**.

   Example: **`get Closed.wav`**

   The announcement file is written to the directory from which you initiated the FTP session.

   Note:

   FTP upload or download of announcement files does not preserve the created timestamp. The file receives the current date and time when it is written to the circuit pack or on the computer.
3. List the FTP client directory contents and ensure that the announcement file is among those listed.
4. Terminate the FTP session (see Ending an FTP session).

Deleting VAL announcements using FTP

You can delete an announcement from a TN2501AP circuit pack or from a LAN device.

Note:

The system denies any attempt to delete an announcement while it is playing, being transferred, or backed up to FLASH (amber LED flashes), regardless of whether the attempt is from a system telephone, the SAT, or through an FTP session.

Before you start

- Know the IP address, the announcement filename that you are deleting, and the VAL circuit pack location (**`list directory board)`**.

Instructions

To delete an announcement on a TN2501AP circuit pack through an FTP session:

1. Ensure that the steps in Setting up an FTP session are complete.
2. At the computer client, type **`delete filename.wav`** and press **Enter**.

   Example: **`delete Closed.wav`**

   Note:

   The announcement file is only removed from volatile RAM memory. Approximately 5 minutes later, the file is removed from nonvolatile ROM flash memory.

3. List the contents of the announcement directory and ensure that the file is not listed.

   Note:

   The .wav file extension on the announcement files are visible when you view the announcement directory from the FTP client.

4. Terminate the FTP session (see Ending an FTP session).
5. At the SAT, type **change announcements** and press **Enter**.

   The **Announcements/Audio Sources** screen appears.

6. Remove the announcement administration by deleting the entire line associated with the announcement.
7. Press **Enter** to save your changes.

Moving announcements to a VAL circuit pack or to another LAN device

You can copy an announcement file to the VAL circuit pack to another LAN device.

Before you start

- Know the announcement filename and its location on the client computer.
- Know the destination IP address, the filename, and circuit pack location of the announcement and VAL circuit pack to which you are moving the announcement (**list directory board**).
- Ensure that you *have not just* administered the announcement on the Announcements/Audio Sources screen. If announcement administration precedes the file transfer, then
  - the announcement appears with a zero (0) length on the **list integrated-annc-boards** screen.
  - The **Time Remaining** fields on the **Integrated Announcement Boards** screen (using either the **list** or **display** commands) do not refresh to reflect the presence of the new announcement file on the circuit pack.

  Use this procedure to ensure that the announcement length is accurate:

  1. Administer the announcement on Communication Manager (**change announcements**), using the identical filename in the **Name** field *without spaces or the .wav file extension*.
  2. Attempt to play the announcement that was administered first and transferred second.

     Communication Manager returns a busy signal at the first play attempt.

  3. Attempt to play the announcement that was administered first and transferred second in a telephone access session.

     Again, a busy signal is returned at the first play attempt.

  4. Re-record this announcement with the same filename at a telephone.

Instructions

To copy an announcement to a VAL circuit pack or to another LAN device:

1. Ensure that the steps in Setting up an FTP session are complete.
2. At the FTP client, type **put filename.wav** and press **Enter**.

   Example: **put Closed.wav**
3. List the contents of the VAL announcement directory or LAN device and look for the announcement file among those listed.

   Note:

   FTP upload or download of announcement files does not preserve its timestamp. The file receives the current date and time when it is written to the circuit pack or to a computer.
4. After you are sure that the announcement is on the VAL circuit pack, administer the announcement on Communication Manager (**change announcements**), using the identical filename in the **Name** field *without spaces or the .wav file extension*.
5. Terminate the FTP session (see Ending an FTP session).

Combining tasks

When you combine copying (the **get** command) and moving (the **put** command) announcement files, you can rearrange VAL announcements.

Before you start

- Know the IP address, the filename, and location of the destination VAL circuit pack to which you are moving an announcement (**list directory board**).

Instructions

To move an announcement to a VAL circuit pack from another VAL circuit pack in an FTP session:

1. Ensure that the steps in Setting up an FTP session are complete.

   Note:

   You must first establish an FTP session into the circuit pack *from which you are restoring an announcement*.
2. List the directory contents and ensure that the announcement file is among those listed.
3. At the FTP client, type **get filename.wav** and press **Enter**.

   Example: **get Closed.wav**

   A copy of the file is written to the directory from which you initiated the FTP session.

4. List the FTP client directory contents to ensure that the announcement is among those listed.
5. Terminate the FTP session (see Ending an FTP session) to the circuit pack *from which you copied the announcement file*.
6. Set up a new FTP session into the destination VAL circuit pack (see Setting up an FTP session).
7. At the FTP client, type **`put filename.wav`** and press **Enter**.

   Example: **`put Closed.wav`**
8. List the VAL announcement directory contents to ensure that the announcement is among those listed.
9. Terminate the FTP session to the circuit pack *to which you copied the announcement file*.

Ending an FTP session

FTP sessions to a VAL circuit pack originate at the FTP client end. You terminate an FTP session by

- logging out from the FTP client (type **`bye`** or **`quit`** and press **Enter**)

  and

- typing **`disable filesystem board board-location`** at the SAT and press **Enter**. (This clears the ftp-username and ftp-password.)

  or

- you can effectively terminate the session from the Avaya Communication Manager end by letting the system time out (10 minutes of inactivity).

SECURITY ALERT:

Both logging out of the FTP session and disabling the VAL circuit pack filesystem provide a higher degree of security.

Note:

If you only disable the circuit pack filesystem, you can continue your FTP session. However, new FTP session logins are not allowed.

VAL Manager

VAL Manager is a standalone application that allows you to copy announcement files and Avaya Communication Manager announcement information to and from Avaya Communication Manager over a LAN connection.

VAL Manager offers the following basic features:

- Simplified administration to add, change, and remove Avaya Communication Manager announcement information.
- The ability to back up and restore announcement files and information to and from Avaya Communication Manager.
- The ability to view the status of announcements on the VAL circuit pack in any Communication Manager.

See your Avaya representative for more information about VAL Manager.

Troubleshooting VAL Announcements

If a working announcement file is deleted via FTP, the next attempt to play the announcement fails, and the system adds a software event to the Denial Events Log. You can view this log to see if the announcement has been deleted, and to see if other events have occurred related to announcements.

Viewing the Denial Events Log

To view the Denial Events Log:

1. At the SAT, type **`display events`** and press **Enter**.

   The **Events Report** screen appears. This input screen helps you focus the report on events of a certain type or from a certain time period.

Figure 187: Events Report

```
display events                                Page  1 of  1   SPE B

                          EVENT REPORT

  The following options control which events will be displayed.

         EVENT CATEGORY

            Category: denial

         REPORT PERIOD

            Interval: ___    From: __/__/__:__ To: __/__/__:__


         SEARCH OPTIONS

                           Vector Number: ___
                              Event Type: _____
```

2. In the **Category** field, select or type **denial**.
3. You can further limit the report by setting the Interval field to one of the following selections (select from the help list or type the first letter):
   - **all**
   - **month**
   - **day**
   - **hour**
   - **minute**
4. Press **Enter**.

   The **Events Report** screen appears.

Figure 188: Events Report screen

```
display events                                              Page   1  of  1

                                EVENTS REPORT

Event Event                     Event     Event      First       Last     Evnt
Type  Description               Data 1    Data 2     Occur       Occur     Cnt

2028  Annc file has bad format  8C0009    0        05/14/11:55 05/14/12:04  2
2027  Annc not found on board   8C0007    0        05/14/12:09 05/14/12:09  1
```

5. Look at the 2027 entry in the **Event Type** field.
   - The **Event Description** field explains that the announcement is not on the circuit pack.
   - The **Event Data 1** field contains the announcement number (hexadecimal in the lower three digits).

   Note:

   This denial event only appears once in the Denial Events Log.

Troubleshooting poor sound quality

If you played an announcement files back in another environment and it sounded great, but when you play it back in an Avaya Communication Manager environment the sound quality is poor, ensure that the file formats are compatible. A good announcement file format must be:

- 8Kbps sample rate
- 8-bit resolution (bits per sample)
- A-law or Mu-law companding format
- Mono (channels = 1)

You must also have the same companding mode administered on page 1 of the **Location Parameters** screen (**`change location-parameters`**).

The system records a software denial event in the Denial Events Log each time it plays an announcement with a bad file format. Refer back to Events Report screen on page 487 and find the **2028** entry in the **Event Type** field:

- The **Event Description** field explains that the announcement has a bad file format.
- The **Event Data 1** field contains the announcement number (hexadecimal in the lower three digits).

Viewing announcement measurements

You can view a report of announcement measurements, including how many times an announcement was queued to play, how many callers dropped while in queue, and how many times all announcement ports were busy during the report period.

For more information about this report, see *Reports for Avaya Communication Manager,* 555-233-505*,* on your documentation CD.

Announcement capacities and load balancing

Understanding how Communication Manager handles annoucements helps with questions of capacity the load-balancing of announcement traffic. The Avaya S8700, S8710, and S8300 media servers support up to 3,000 announcements across the Communication Manager system. Each of as many as 10 integrated announcement boards (and each G700 virtual VAL source) is limited to 256 announcements. All Avaya G700 Media Gateways in the system can be assigned as VAL sources, for a maximum of 250 G700 VAL sources in a large S8700 or S8710 media server configuration also with up to 10 TN2501 VAL boards, providing an ultimate capacity of 260 sources for the Avaya DEFINITY R and S8700 or S8710 servers.

Wach TN2501 has 31 playback ports (with 60 minutes of storage capacity) and each G700 VAL source has 15 playback ports (with 20 minutes storage capacity). Initially each caller that is to hear a non-barge-in announcement on that source is connected to an available port until each port has one caller connected. Initially 31 callers can be hearing an announcement from a single TN2501 board (15 with a G700 VAL source). The same announcement can be playing from multiple ports, each starting at a different time. Once all ports are busy playing announcements, subsequent callers that are to hear an announcement residing on that board/source are queued for that source. Communication Manager queue size over all integrated announcement sources is 4,000 callers (DEFINITY Server R & S8700/S8710/S8300 media servers). When a port becomes available, all callers (up to 1,000) waiting in the queue for a specific announcement on that source are connected to the available port to hear the announcement play from the beginning on a first-in, first-out basis. At any given time you can have as many as 1,000 callers hearing the same announcement on a given port of a board/source, yielding a (theoretical, system-wide) capacity of 31,000 callers connected to a VAL board (or 15,000 callers connected to a G700 Media Gateway virtual VAL source). Factor in barge-in announcements which continuously play from a single port (that can have as many as 1,000 callers connected at the same), and this provides a very, very large capacity for playing announcements. The Measurements Announcements commands can be used to monitor the system and help determine burst conditions, allowing for better load balancing of the traffic.

Related topics

For more information, see "Announcements" in *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205.

Non-Volatile Storage of Announcements on VAL

The VAL board has a RAM based file system on which the announcements are stored. This is where a newly recorded or FTP'd announcement resides. The VAL constantly checks to determine whether changes have been made to the file system (file added or deleted or renamed). Once a change has been detected, the VAL waits for 5 minutes of stability. After such a stable period, the VAL backs up the RAM based file system to its FLASH. The FLASH version of the file system is non-volatile and is what is restored upon reset/reseat.

If you record a new announcement and immediately reset the board, the announcement will be lost. The yellow LED blinks while a backup from RAM to FLASH occurs. All of the above is true regardless which (if any) of the .wav files on the board are administered announcements. It is just the way the board (firmware) operates. Note that only files with a .wav extension in the /annc directly are backed up to FLASH.

Avaya Communication Manager announcement synchronicity

As mentioned above, there is a separation between the actual files on the VAL board and the set of administered announcements. You can have 100 .wav files on a VAL and only be using (administered) 5 of them. The best way to visualize this is to look at the difference between a **`list integrated`** and the **`list directory board <brdloc>/annc`** commands.

By going into the **Change Announcement** screen and changing the name of an announcement, what is happening? You are changing only the administration of the announcement. This means when the extension is called, the new file name is requested. The board still has a file with the old file name but not one with the new file name. You then could run **`change announcement`** again and switch the name back to the old one and the announcement will play fine. The VAL board is completely unaware that you change the **`change announcement`** command.

Changing the name of the announcement

To change the name of an announcement you must do two things.

1. Change the administered name (using the **change announcement** command)
2. Change the name of the file on the VAL board

Changing the name of the file on the VAL board

There are two ways to accomplish this.

Note:

For the following steps, do not use the FTP **rename** command since the FTP server on the VAL board does not support it.

To change the name of the file:

1. Change the name of the existing file using the PC.
   a. Access the file on the PC if already there or FTP the announcement file from the VAL board (in binary mode) to the PC.
   b. Rename it locally on the PC. The administered name given to the file must be the same as the filename contained on the TN2501 (VAL) board.
   c. FTP the file with the new name back to the VAL board.
   d. Assign the new name to the existing extension on the announcement screen
   e. Delete the old file on the board using the **remove file** command, if you wish, or
2. Re-record the announcement with the new file name.
   a. FTP the new announcement to the VAL board.
   b. Assign the new name to the existing extension on the announcement screen (replacing the old file name).
   c. Delete the old announcement file using the **remove file** command (if desired).

Chapter 14: Managing Group Communications

Setting up Voice Paging Over Loudspeakers

Use this procedure to allow users to make voice pages over an external loudspeaker system connected to Avaya Communication Manager. If you're using an external paging system instead of an auxiliary trunk circuit pack, don't use this procedure. External systems typically connect to a trunk or station port and are not administered through the **Loudspeaker Paging** screen.

Before you start

Your server running Communication Manager must have one or more auxiliary trunk circuit packs with enough available ports to support the number of paging zones you define. Each paging zone requires 1 port. For information on specific circuit packs, see the *Hardware Guide for Avaya Communication Manager*, 555-245-207.

Instructions

As an example, we will set up voice paging for an office with 5 zones. We'll allow users to page all 5 zones at once, and we'll assign a class of restriction of 1 to all zones.

1. Type **`change paging loudspeaker`** and press **Enter**.

   The **Loudspeaker Paging** screen appears.

Figure 189: Loudspeaker Paging screen

```
                              LOUDSPEAKER PAGING

                                    CDR? y
              Voice Paging Timeout (sec): 30
             Code Calling Playing Cycles:

PAGING PORT ASSIGNMENTS
                   Voice Paging     Code Calling
 Zone  Port        TAC  COR  TN     TAC  COR  TN     Location:
   1:  01C0501     301   1   1                1      Reception area
   2:  01C0502     302   1   1                1      Shipping & receiving
   3:  01C0503     303   1   1                1      Staff offices
   4:  01C0504     304   1   1                1      Management suite
   5:  01C0601     305   1   1                1      Breakroom
   6:                        1                1
   7:                        1                1
   8:                        1                1
   9:                        1                1
 ALL:              310   1   1                1
```

2. In the **Voice Paging Timeout** field, type **30**.

   This field sets the maximum number of seconds a page can last. In our example, the paging party will be disconnected after 30 seconds.

3. In the **Port** field for **Zone 1**, type **01C0501**.

   Use this field to assign a port on an auxiliary trunk circuit pack to this zone.

4. In the **Voice Paging — TAC** field type **301**.

   Use this field to assign the trunk access code users dial to page this zone. You cannot assign the same trunk access code to more than one zone.

5. In the **Voice Paging — COR** field type **1**.

   Use this field to assign a class of restriction to this zone. You can assign different classes of restriction to different zones.

6. On the **Zone 1** row, type **Reception area** in the **Location** field.

   Give each zone a descriptive name so you can easily remember the corresponding physical location.

7. Repeat steps 4 through 6 for zones 2 to 5.

8. In the **ALL** row, type **310** in the **Voice Paging — TAC** field and **1** in the **Voice Paging — COR** field.

   By completing this row, you allow users to page all zones at once. You do not have to assign a port to this row.

9. Press **Enter** to save your changes.

You can integrate loudspeaker voice paging and call parking. This is called "deluxe paging." You enable deluxe paging by entering **y** in the **Deluxe Paging and Call Park Timeout to Originator** field on the **Feature-Related System Parameters** screen. To allow paged users the full benefit of deluxe paging, you should also enter a code in the **Answer Back Access Code** field on the **Feature Access Code** screen if you haven't already: paged users will dial this code + an extension to retrieve calls parked by deluxe paging.

Fixing problems

Problem	Possible causes	Solutions
Users report that they can't page.	The attendant has taken control of the trunk group.	Deactivate attendant control.
Calls to an extension are heard over the loudspeakers.	The extension may have been forwarded to a trunk access code used for paging.	Deactivate call forwarding or change the extension calls are forwarded to.

More information

Users page by dialing the trunk access code assigned to a zone and speaking into their handset. For your users' convenience, you may also want to consider the following options:

- Add the paging trunk access codes to an abbreviated dialing list and allow users to page using the list.
- Assign individual trunk access codes to Autodial buttons.
- Assign individual trunk access codes to Busy buttons. The status lamp tells the user whether or not the trunk is busy.
- For attendants, you can provide one-button paging access by assigning trunk access codes for paging zones to the Direct Trunk Group Select buttons on the attendant console.

With an appropriate class of restriction, remote callers can also make loudspeaker pages.

When deluxe paging is enabled, if a user with an active call dials the trunk access code for a paging zone the active call is automatically parked.

- Users dial the trunk access code + "#" to page and park an active call on their own extensions.
- Users with console permission can park a call on any extension by dialing the trunk access code + the extension.
- Attendants or users with console permissions may park calls to common shared extensions.
- Parked calls can be retrieved from any telephone. Paged users simply dial the answer back feature access code + the extension where the call is parked.

Related topics

See Paging Over Speakerphones on page 498 for another way to let users page.

See Loudspeaker Paging" in *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205, for detailed information on voice paging over loudspeakers.

Setting up Chime Paging Over Loudspeakers

Use this procedure to allow users to make chime pages over an external loudspeaker system connected to your media server. Users page by dialing a trunk access code and the extension of the person they want to page. The system plays a unique series of chimes assigned to that extension. This feature is also known as Code Calling Access.

Before you start

Your server running Communication Manager must have one or more auxiliary trunk circuit packs with enough available ports to support the number of paging zones you define. Each paging zone requires 1 port. For information on specific circuit packs, see the *Hardware Guide for Avaya Communication Manager*, 555-245-207.

Instructions

To set up chime paging, you fill out the necessary fields on the **Loudspeaker Paging** screen and then assign chime codes to individual extensions on the **Code Calling IDs** screen.

As an example, we will set up chime paging for a clothing store with 3 zones. We'll allow users to page all zones at once, and we will assign a class of restriction of 1 to all zones.

1. Type **`change paging loudspeaker`** and press **Enter**.

   The **Loudspeaker Paging** screen appears.

Figure 190: Loudspeaker Paging screen

```
                              LOUDSPEAKER PAGING

                                    CDR? y
              Voice Paging Timeout (sec):
             Code Calling Playing Cycles: 2

PAGING PORT ASSIGNMENTS
                   Voice Paging      Code Calling
 Zone  Port        TAC  COR  TN      TAC  COR  TN     Location:
   1:  01A0301                1      80    1   1      Men's Department
   2:  01A0302                1      81    1   1      Women's Department
   3:  01A0303                1      82    1   1      Children's
   4:                         1                 1
   5:                         1                 1
   6:                         1                 1
   7:                         1                 1
   8:                         1                 1
   9:                         1                 1
 ALL:                         1      89    1   1
```

2. In the **Code Calling Playing Cycles** field, type **2**.

   This field sets the number of times a chime code plays when someone places a page.

3. In the **Port** field for **Zone 1**, type **01A0301**.

   Use this field to assign a port on an auxiliary trunk circuit pack to this zone.

4. In the **Code Calling — TAC** field type **80**.

   Use this field to assign the trunk access code users dial to page this zone. You cannot assign the same trunk access code to more than one zone.

5. In the **Code Calling — COR** field type **1**.

   Use this field to assign a class of restriction to this zone. You can assign different classes of restriction to different zones.

6. On the **Zone 1** row, type **Men's Department** in the **Location** field.

   Give each zone a descriptive name so you can easily remember the corresponding physical location.

7. Repeat steps 4 through 6 for zones 2 and 3.

8. In the **ALL** row, type **89** in the **Code Calling — TAC** field and **1** in the **Code Calling — COR** field.

   By completing this row, you allow users to page all zones at once. You do not have to assign a port to this row.

9. Press **Enter** to save your changes.

To assign chime codes to individual extensions:

1. Type **`change paging code-calling-ids`** and press **Enter**.

   The **Code Calling IDs** screen appears.

Figure 191: Code Calling IDs screen

```
                                CODE CALLING IDs
ID ASSIGNMENTS
          Id   Ext        Id   Ext     Id   Ext     Id   Ext     Id   Ext
          111: 2130       141: _______ 221: _______ 251: _______ 331: _______
          112: 2131       142: _______ 222: _______ 252: _______ 332: _______
          113: 2149       143: _______ 223: _______ 253: _______ 333: _______
          114: 2150       144: _______ 224: _______ 254: _______ 334: _______
          115: 2152       145: _______ 225: _______ 255: _______ 335: _______
          121: 2153       151: _______ 231: _______ 311: _______ 341: _______
          122: 2160       152: _______ 232: _______ 312: _______ 342: _______
          123: 2167       153: _______ 233: _______ 313: _______ 343: _______
          124: _________  154: _______ 234: _______ 314: _______ 344: _______
          125: _________  155: _______ 235: _______ 315: _______ 345: _______
          131: _________  211: _______ 241: _______ 321: _______ 351: _______
          132: _________  212: _______ 242: _______ 322: _______ 352: _______
          133: _________  213: _______ 243: _______ 323: _______ 353: _______
          134: _________  214: _______ 244: _______ 324: _______ 354: _______
          135: _________  215: _______ 245: _______ 325: _______ 355: _______
```

2. Type the first extension, **2130**, in the **Ext** field for **Id 111**.

   Each code Id defines a unique series of chimes.

3. Assign chime codes to the remaining extensions by typing an extension number on the line following each code Id.

   You can assign chime codes to as many as 125 extensions.

4. Press **Enter** to save your changes.

Fixing problems

Problem	Possible causes	Solutions
Users report that they can't page.	The attendant has taken control of the trunk group.	Deactivate attendant control.

More information

Users page by dialing the trunk access code assigned to a zone. For your users' convenience, you may also want to consider the following options:

- Add the paging trunk access codes to an abbreviated dialing list and allow users to page using the list.

  Note:

  Don't use special characters in abbreviated dialing lists used with chime paging.

- Assign individual trunk access codes to Autodial buttons.
- Assign individual trunk access codes to Busy buttons. The status lamp tells the user whether or not the trunk is busy.
- For attendants, you can provide one-button paging access by assigning trunk access codes for paging zones to the **Direct Trunk Group Select** buttons on the attendant console.

With an appropriate class of restriction, remote callers can also make loudspeaker pages.

Related Topics

See Paging Over Speakerphones below for another way to let users page.

See "Loudpeaker Paging" in *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205, for detailed information on chime paging over loudspeakers.

Paging Over Speakerphones

Use this procedure to allow users to make an announcement over a group of digital speakerphones. By dialing a single extension that identifies a group, users can page over all the speakerphones in that group. Speakerphone paging is one-way communication: group members hear the person placing the page but cannot respond directly.

Before you start

You must have DCP set speakerphones or IP set speakerphones.

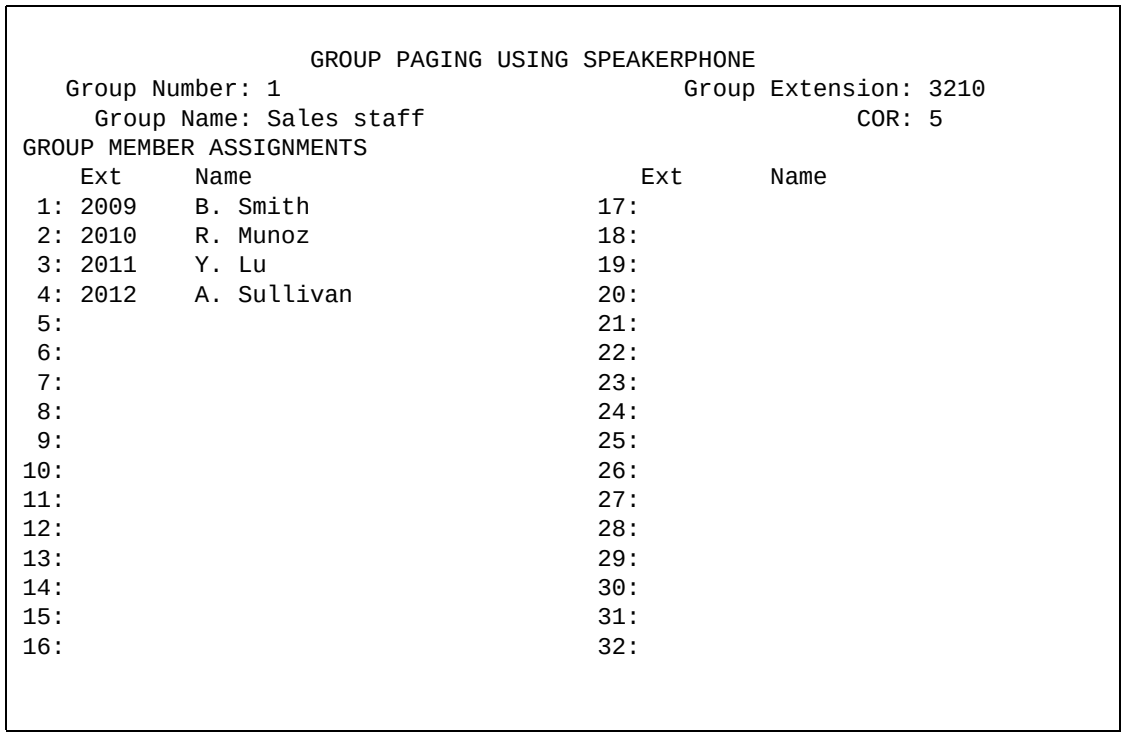
Instructions

To set up speakerphone paging, you create a paging group and assign phones to it. In the following example, we'll create paging group 1 and add 4 members.

1. Type **add group-page 1** and press **Enter**.

   The **Group Paging Using Speakerphone** screen appears.

Figure 192: Group Paging Using Speakerphone screen



```
                    GROUP PAGING USING SPEAKERPHONE
   Group Number: 1                              Group Extension: 3210
     Group Name: Sales staff                                 COR: 5
GROUP MEMBER ASSIGNMENTS
    Ext     Name                              Ext      Name
 1: 2009    B. Smith                     17:
 2: 2010    R. Munoz                     18:
 3: 2011    Y. Lu                        19:
 4: 2012    A. Sullivan                  20:
 5:                                      21:
 6:                                      22:
 7:                                      23:
 8:                                      24:
 9:                                      25:
10:                                      26:
11:                                      27:
12:                                      28:
13:                                      29:
14:                                      30:
15:                                      31:
16:                                      32:
```

2. In the **Group Extension** field, type **3210**.

   This field assigns the extension users dial to page the members of this group.

3. In the **Group Name** field, type **Sales staff**.

   This name appears on callers' telephone displays when they page the group.

4. In the **COR** field, type **5**.

   Any user who wants to page this group must have permission to call COR 5.

5. In the **Ext** field in row 1, type **2009**.

6. Enter the remaining extensions that are members of this group.

   Communication Manager fills in the **Name** fields with the names from the **Station** screen when you save your changes.

7. Press **Enter** to save your changes.

Fixing problems

Problem	Possible causes	Solutions
Users get a busy signal when they try to page.	All phones in the group are busy or off-hook.	Wait a few minutes and try again.
	All phones in the group have Send All Calls or Do Not Disturb activated.	Group members must deactivate these features in order to hear a page.
Some group members report that they don't hear a page.	Some phones in the group are busy or off-hook.	Wait a few minutes and try again.
	Some phones in the group have Send All Calls or Do Not Disturb activated.	Group members must deactivate these features in order to hear a page.

More information

- You can create up to 32 paging groups on Avaya Communication Manager.
- Each group can have up to 32 extensions in it.
- One telephone can be a member of several paging groups.

Related topics

See "Group Paging" in *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205, for detailed information on paging over speakerphones.

Paging Users Who are on Active Calls

Use this procedure to allow one user to interrupt another user's call and make a private announcement. This is called whisper paging. The paging user dials a feature access code or presses a feature button, then dials the extension they want to call. All 3 users can hear the tone that signals the page, but only the person on the paged extension can hear the pager's voice: other parties on the call cannot hear it, and the person making the page cannot hear anyone on the call.

Before you start

Before you administer whisper paging:

- Your Communication Manager server must have a circuit pack that supports whisper paging. For information on specific models, see the *Hardware Guide for Avaya Communication Manager*, 555-245-207.
- Users must have 6400-, 7400-, 8400-, or 9400-series DCP (digital) phones.

Instructions

You give users the ability to use whisper paging by administering feature buttons or feature access codes.

You can give users feature buttons that make, answer, or block whisper pages. Using the **Station** screen, you can administer these buttons in any combination as appropriate:

- Whisper Page Activation — allows this user to place a whisper page
- Answerback — allows this user to answer a whisper page

  Pressing the answerback button automatically puts any active call on hold and connects the paged user to the paging user.
- Whisper Page Off— allows this user to block whisper pages

  If possible, assign this function to a button with a lamp so the user can tell when blocking is active. You cannot administer this button to a soft key.

To allow users to make a whisper page by dialing a feature access code, you simply need to enter a code in the **Whisper Page Activation Access Code** field on the **Feature Access Code** screen.

Related topics

See "Whisper Paging" in *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205, for detailed information on whisper paging.

Using Telephones as Intercoms

Use this procedure to make communications quicker and easier for users who frequently call each other. With the intercom feature, you can allow one user to call another user in a predefined group just by pressing a couple of buttons. You can even administer a button that always calls a predefined extension when pressed.

Instructions

Administering the intercom feature is a 2-step process. First, you create an intercom group and assign extensions to it. Then, to allow group members to make intercom calls to each other, you administer feature buttons on their phones for automatic intercom, dial intercom, or both. This section also provides instructions for allowing one user to pick up another user's intercom calls.

In this example, we'll create intercom group 1 and add extensions 2010 to 2014.

1. Type **`add intercom-group 1`** and press **Enter**.

   The **Intercom Group** screen appears.

Figure 193: Intercom Group screen

```
                                INTERCOM GROUP
                                             Group Number: 1
                                    Length of Dial Code: 1

GROUP MEMBER ASSIGNMENTS
              Ext      DC   Name
           1: 2010     1    B. Smith
           2: 2011     2    L. Yu
           3: 2012     3    R. Munoz
           4: 2013     4    K. Mancetti
           5: 2014     9    N. Mitchell
           6:
           7:
           8:
           9:
          10:
          11:
          12:
          13:
          14:
          15:
          16:
```

2. Type **1** in the **Length of Dial Code** field.

   Dial codes can be 1 or 2 digits long.

3. On row 1, type **2010** in the **Ext** field.
4. On row 1, type **1** in the **DC** field.

   This is the code a user will dial to make an intercom call to extension 2010. The length of this code must exactly match the entry in the **Length of Dial Code** field.

5. Repeat steps 3 and 4 for the remaining extensions.

   Dial codes don't have to be in order. Communication Manager fills in the **Name** field with the name from the **Station** screen when you save changes.

6. Press **Enter** to save your changes.

To allow users to make intercom calls, you must administer feature buttons on the phones in the intercom group. You can administer buttons for dial intercom, automatic intercom, or both on multi-appearance phones. You can't administer either intercom feature on single-line phones, but you can assign single-line phones to intercom groups so those users can receive intercom calls.

As an example, we will set up automatic intercom between extensions 2010 (dial code = 1) and 2011 (dial code = 2) in intercom group 1.

To set up automatic intercom between extensions 2010 and 2011:

1. Type **change station 2010** and press **Enter**.

   The **Station** screen appears.

Figure 194: Station screen

```
                                    STATION
SITE DATA
      Room: _______                                  Headset? n
      Jack: _____                                    Speaker? n
     Cable: _____                                   Mounting: d
     Floor: _______                               Cord Length: 0
  Building: _______                                 Set Color: ______


ABBREVIATED DIALING
     List1: ________           List2: ________           List3: ________

BUTTON ASSIGNMENTS
  1: call-appr_                          6: ___________
  2: call-appr_                          7: ___________
  3: call-appr_                          8: ___________
  4: auto-icom_  Grp: 1  DC: 2           9: ___________
  5: dial-icom_  Grp: 1                 10: ___________
```

2. Move to the page with the BUTTON ASSIGNMENTS fields.
3. In BUTTON ASSIGNMENTS field **4**, type **auto-icom** and press **Tab**.

   The **Grp** and **DC** fields appear.
4. In the **Grp** field, type **1**.

   This is the number of the intercom group. Since an extension can belong to more than one intercom group, you must assign a group number to intercom buttons.
5. In the **DC** field, type **2**.

   This is the dial code for extension 2011, the destination extension.
6. Press **Enter** to save your changes.
7. Repeat steps 1 to 6 for extension 2011.

   Assign a dial code of **1** to 2011's automatic intercom button.

To give a member of a group the ability to make intercom calls to all the other members, administer a **Dial Intercom** button on the member's telephone. Type the number of the intercom group in the **Grp** field beside the **Dial Intercom** button.

You can also give one user instant, one-way access to another. For example, to give user A instant, one-way access to user B, administer an **Automatic Intercom** button on A's telephone only. You don't have to administer any intercom button on B's telephone. If B has a **Dial Intercom** button, he can make an intercom call to A the same way as he would to any other group member.

When users are in the same call pickup group, or if Directed Call Pickup is enabled on your server running Communication Manager, one user can answer an intercom call to another user. To allow users to pick up intercom calls to other users, you must enter **y** in the **Call Pickup on Intercom Calls** field on the **Feature-Related System Parameters** screen.

Related topics

See "Abbreviated Dialing" in *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205, for information on another way for users to call each other without dialing complete extension numbers.

See "Intercom" in *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205 for detailed information on intercom functions.

Setting up Automatic Answer Intercom Calls

Automatic Answer Intercom Calls (Auto Answer ICOM) allows a user to answer an intercom call within the intercom group without pressing the intercom button. Auto Answer ICOM works with digital, BRI, and hybrid phones with built-in speaker, headphones, or adjunct speakerphone.

SECURITY ALERT:

Press the **Do Not Disturb** button or the **Send All Calls** button on your telephone when you don't want someone in your intercom group to listen in on a call. Auto Answer ICOM does not work when the **Do Not Disturb** button or the **Send All Calls** button is pressed on the telephone.

Administration

This section contains an example, with step-by-step instructions, on how to set up Auto Answer ICOM.

In this example, you set up Auto Answer ICOM on station 12345. To do so, complete the following steps:

1. Type **change station 12345**.

   The **Station** screen for extension 12345 appears.

Figure 195: Station screen

```
change station 75001                                          Page 2 of X
                                    STATION

FEATURE OPTIONS
           LWC Reception? spe                 Auto Select Any Idle Appearance? n
          LWC Activation? y                             Coverage Msg Retrieval? y
  LWC Log External Calls? n                                   Auto Answer: none
             CDR Privacy? n                               Data Restriction? n
   Redirect Notification? y                       Idle Appearance Preference? n
 Per Button Ring Control? n
   Bridged Call Alerting? n                         Restrict Last Appearance? y
  Active Station Ringing: single

        H.320 Conversion? n             Per Station CPN - Send Calling Number? _
       Service Link Mode: as-needed
         Multimedia Mode: basic
    MWI Served User Type: ______                    Display Client Redirection? n
         Automatic Moves:
              AUDIX Name:                          Select Last Used Appearance? n
                         _                           Coverage After Forwarding? _
     Recall Rotary Digit? n                            Multimedia Early Answer? n

Remote Softphone Emergency Calls: as-on-local   Direct IP-IP Audio Connections? n
Emergency Location Ext: 75001                            IP Audio Hairpinning? n
```

2. Move to the **Auto Answer** field and enter **icom**.
3. Press **Enter** to save your changes.

Observing Calls

Use this procedure to allow designated users, normally supervisors, to listen to other users' calls. This capability is often used to monitor service quality in call centers and other environments where employees serve customers over the telephone. On Avaya Communication Manager, this is called "service observing" and the user observing calls is the "observer."

This section describes service observing in environments without Automatic Call Distribution (ACD) or call vectoring. To use service observing in those environments, see *Avaya Communication Manager Contact Center Guide to ACD Contact Centers*.

Before you start

On the **System Parameter Customer-Options** screen, verify that the:

- **Service Observing (Basic)** field is **y**.

If you want to enable remote service observing by allowing remote users to dial a feature access code, verify the:

- **Service Observing (Remote/By FAC)** field is **y**.

If the appropriate field is not enabled, contact your Avaya representative.

Instructions

SECURITY ALERT:
Listening to someone else's calls may be subject to federal, state, or local laws, rules, or regulations. It may require the consent of one or both of the parties on the call. Familiarize yourself with all applicable laws, rules, and regulations and comply with them when you use this feature.

In this example, we'll set up service observing for a manager. The manager's class of restriction is 5. We'll assign a feature button to the manager's telephone and allow her to monitor calls on local extensions that have a class of restriction of 10. Everyone on an observed call will hear a repetitive warning tone.

To set up service observing:

1. Set the observer's class of restriction to permit service observing:
   a. In the **Class of Restriction** screen for COR 5, type **y** in the **Can Be A Service Observer?** field.

b. Move to the page of the **Class of Restriction** screen that shows service observing permissions.

c. Type **y** in the field for class of restriction 10.

2. In the **Class of Restriction** screen for COR 10, type **y** in the **Can Be Service Observed?** field.

   Anyone with class of restriction 5 now has permission to observe extensions with class of restriction 10. To further restrict who can observe calls or be observed, you may want to create special classes of restriction for both groups and use these classes only for the appropriate extensions.

3. In the **Station** screen, assign a **Service Observing** button to the observer's telephone.

   A service observing button permits users to switch between listen-only and listen-and-talk modes simply by pressing the button.

4. To activate the warning tone, type **y** in the **Service Observing — Warning Tone** field on the **Feature-Related System Parameters** screen.

   A unique 2-second, 440-Hz warning tone plays before an observer connects to the call. While the call is observed, a shorter version of this tone repeats every 12 seconds.

In order for users to activate service observing by feature access codes, use the Feature Access Code screen to administer codes in one or both of the following fields:

- **Service Observing Listen Only Access Code**
- **Service Observing Listen/Talk Access Code**

When using feature access codes, observers must choose a mode at the start of the session. They cannot switch to the other mode without ending the session and beginning another.

Note:

Feature access codes are required for remote observing.

Related topics

See "Service Observing" in *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205, for detailed information on service observing.

Chapter 15: Managing Data Calls

Types of Data Connections

You can use Avaya Communication Manager to allow the following types of data elements/ devices to communicate to the world:

- Data Terminals
- Personal computers
- Host Computers (for example, CentreVu CMS or INTUITY AUDIX)
- Digital Phones (Digital Communications Protocol (DCP) and Integrated Services Digital Network-Basic Rate Interface (ISDN-BRI))
- Audio/Video Equipment
- Printers
- Local area networks (LAN)

You enable these connections using a large variety of data communications equipment, such as:

- Modems
- Data Modules
- Asynchronous Data Units (ADU)
- Modem Pools
- Data/modem pooling circuit packs

Once you have connected these data devices to Communication Manager, you can use networking and routing capabilities to allow them to communicate with other devices over your private network or the public network.

This section describes the system features available to enable data communications.

Data Call Setup

Data Call Setup provides multiple methods to set up a data call:

- Data-terminal (keyboard) dialing
- Telephone dialing
- Hayes AT command dialing
- Administered connections
- Hotline dialing

Administering Data Call Setup

For data-terminal dialing:

1. Choose one of the following data modules and administer all fields:
   - Processor/Trunk Data Module
   - Data Line Data Module
   - 7500 Data Module
2. On the **Modem Pool Group** screen, administer the **Circuit Pack Assignments** field.

   See Modem Pool Group on page 1287 for more information.

For telephone dialing:

1. Choose one of the following:
   - On the **Feature Access Code** screen, administer the **Data Origination Access Code** field. See Feature Access Code on page 956 for more information.
   - On the **Station** screen, assign one button as data-ext (Ext:). See Station on page 1409 for more information.
2. Choose one of the following data modules and administer all fields:
   - Processor/Trunk Data Module
   - Data Line Data Module
3. On the **Modem Pool Group** screen, administer the **Circuit Pack Assignments** field. See Modem Pool Group on page 1287 for more information.

Depending on the hardware used, assign ports to the following:

- Data modules
- 7400D-series or CALLMASTER digital telephones
- 7500D-series telephones with asynchronous data module (ADM)
- Analog modems (port is assigned using 2500 telephone screen)

Characters used in Data Call Setup

Basic-digit dialing is provided through an ADM or 7500B data module. The user can enter digits from 0 to 9, *, and # from a 7500 or 8500 series telephone keypad or an EIA-terminal interface. In addition, the user can dial the following special characters.

Table 5: Special characters

Character	Use
SPACE, -, (, and)	improves legibility. Communication Manager ignores these characters during dialing.
+ character (wait)	interrupts or suspends dialing until the user receives dial tone
, (pause)	inserts a 1.5-second pause
% (mark)	indicates digits for end-to-end signaling (touch-tone). This is required when the trunk is rotary. It is not required when the trunk is touch-tone.
UNDERLINE or **BACKSPACE**	corrects previously typed characters on the same line
@	deletes the entire line and starts over with a new DIAL: prompt

Each line of dialing information can contain up to 42 characters (the + and % characters count as two each).

Examples of dialing are:

- DIAL: 3478
- DIAL: 9+(201) 555-1212
- DIAL: 8, 555-2368
- DIAL: 9+555-2368+%9999+123 (remote access)

The following call-progress messages and their meanings are provided for DCP and ISDN-BRI modules.

Table 6: Call-progress messages *1 of 2*

Message	Application	Meaning
DIAL:	DCP	Equivalent to dial tone. Enter the desired number or FAC followed by Enter.
CMD	BRI	Equivalent to dial tone. Enter the desired number or FAC followed by Enter.
RINGING	DCP, BRI	Equivalent to ringing tone. Called terminal is ringing.
BUSY	DCP, BRI	Equivalent to busy tone. Called number is busy or out of service.
ANSWERED	DCP, BRI	Call is answered.
ANSWERED - NOT DATA	DCP	Call is answered and a modem answer tone is not detected.
TRY AGAIN	DCP, BRI	Equivalent to reorder tone. System facilities are currently not available.
DENIED	DCP, BRI	Equivalent to intercept tone. Call cannot be placed as dialed.
ABANDONED	DCP, BRI	Calling user has abandoned the call.
NO TONE	DCP, BRI	Tone is not detected.
CHECK OPTIONS	DCP, BRI	Data-module options are incompatible.
XX IN QUEUE	DCP, BRI	Current position in queue.
PROCESSING	DCP, BRI	Out of queue. Facility is available.
TIMEOUT	DCP, BRI	Time is exceeded. Call terminates.
FORWARDED	DCP, BRI	Equivalent to redirection-notification signal. Called terminal activates Call Forwarding and receives a call, and call is forwarded.
INCOMING CALL	DCP, BRI	Equivalent to ringing.

1 of 2

Table 6: Call-progress messages *2 of 2*

Message	Application	Meaning
INVALID ADDRESS	DCP	Entered name is not in alphanumeric-dialing table.
WRONG ADDRESS	BRI	Entered name is not in alphanumeric-dialing table.
PLEASE ANS-	DCP, BRI	Originating telephone user transferred call to data module using One-Button Transfer to Data.
TRANSFER	DCP	Data Call Return-to-Voice is occurring.
CONFIRMED	DCP, BRI	Equivalent to confirmation tone. Feature request is accepted, or call has gone to a local coverage point.
OTHER END	DCP, BRI	Endpoint has terminated call.
DISCONNECTED	DCP, BRI	Call is disconnected.
WAIT	DCP, BRI	Normal processing continues.
WAIT, XX IN QUEUE	DCP	Call is in a local hunt-group queue.
		2 of 2

DCP data modules

Data-terminal dialing

DCP data-terminal dialing allows a user to set up and disconnect data calls directly from a data terminal as follows.

1. At the **DIAL**: prompt, the user types the data number. The message **RINGING** displays.
2. If the call is queued, the message **WAIT, XX IN QUEUE** displays. The queue position XX updates as the call moves up in queue.
3. To originate and disconnect a call, the user presses **BREAK**. If the terminal does not generate a 2-second continuous break signal, the user can press originate/disconnect on the data module.
4. The user can enter digits at the **DIAL**: prompt.

Telephone dialing

DCP telephone dialing allows telephone users to originate and control data calls from a telephone.

Users can set up a call using any unrestricted telephone and then transfer the call to a data endpoint.

The primary way to make data calls is with multiappearance telephone data-extension buttons. Assign any administrable feature button as a data-extension button. The data-extension button provides one-touch access to a data module. The number of assigned data-extension buttons per telephone is not limited.

The following options, either alone or combined, permit flexibility in making data calls from a telephone.

- One-Button Transfer to Data

  A user can transfer a call to the associated data module by pressing the data-extension button after the endpoint answers.

- Return-to-Voice

  A user can change the connection from data to voice. The user presses the data-extension button associated with the busy data module. If the user hangs up, the call disconnects. Return of a data call to the telephone implies that the same data call is continued in the voice mode, or transferred to point.

  The Return-to-Voice feature is denied for analog adjuncts.

- Data Call Preindication

  A user, before dialing a data endpoint, can reserve the associated data module by pressing the data-extension button. This ensures that a conversion resource, if needed, and the data module are reserved for the call. Avaya recommends the use of Data Call Preindication before 1-button transfer to data for data calls that use toll-network facilities. Data Call Preindication is in effect until the associated data-extension button is pressed again for a 1-button transfer; there is no time-out.

ISDN-BRI data modules

Data-terminal dialing

A user can set up and disconnect data calls directly from a data terminal without using a telephone as follows:

1. The user presses **Enter** a few times. If the CMD: prompt does not appear, the user presses **Break A + T** at the same time, and then presses **Enter**.
2. At the **CMD**: prompt, the user types and presses **Enter**.
3. To disconnect, the user enters +++. At the CMD: prompt, the user types **end** and presses **Enter**.

Telephone dialing

To make a data call, an ISDN-BRI telephone user presses the data button on the terminal, enters the number on the dial pad, and then presses the data button again.

The following data functions are not available on ISDN-BRI phones:

- One-Button Transfer to Data
- Return-to-Voice
- Data Call Preindication
- Voice-Call Transfer to Data and Data-Call Transfer to Voice

The system handles all presently defined BRI bearer data-call requests. Some capabilities that are not supported by Avaya terminals are provided by non-Avaya terminals. If Communication Manager does not support a capability, a proper cause value returns to the terminal.

BRI terminals receive a cause or reason code that identifies why a call is being cleared. The BRI data module converts certain cause values to text messages for display.

In a passive-bus multipoint configuration, the system supports two BRI endpoints per port, thus doubling the capacity of the BRI circuit pack. When you change the configuration of a BRI from point-to-point to multipoint, the original endpoint does not need to reinitialize. Only endpoints that support service profile identifier (SPID) initialization can be administered in a multipoint configuration.

Analog modems

When a telephone user places a data call with a modem, the user dials the data-origination access code assigned in the system before dialing the endpoint.

Considerations

- A BRI telephone cannot call a data terminal, and a data terminal cannot call a BRI telephone.

Interactions

- Abbreviated Dialing

  Only 22 of the 24 (maximum) digits in an abbreviated-dialing number are available for keyboard dialing. The remaining two digits must contain the wait indicator for tone detection.

- Call Coverage

  A hunt group made up of data endpoints cannot be assigned a coverage path.

- Call Detail Recording

  CDR records the use of modem pools on trunk calls.

- Call Forwarding All Calls

  Calls received by a data module can be forwarded. Activate Call Forwarding All Calls with data-terminal (keyboard) dialing. If the forwarded-to endpoint is an analog endpoint and the caller is a digital endpoint, modem pooling is activated automatically.

- Pooled Modems with Hunt Groups

  UCD can provide a group of data modules or analog modems for answering calls to connected facilities (for example, computer ports).

- World-Class Tone Detection

  Multiple-line data-terminal dialing is supported if the administered level of tone detection is precise. You can administer tone-detection options. The message that Data Call Setup sends to users varies according to the option.

  If the option is not set to precise, and a data call is set up over an analog trunk, messages describing the status of the called endpoint (for example, RINGING, BUSY, TRY AGAIN) change according to which tone-detection option is selected.

Alphanumeric Dialing

Alphanumeric Dialing enhances data-terminal dialing by allowing users to place data calls by entering an alphanumeric name rather than a long string of numbers.

For example, a user could type 9+1-800-telefon instead of 9+1-800-835-3366 to make a call. Users need to remember only the alpha-name of the far-end terminating point.

Alphanumeric Dialing allows you to change a mapped string (digit-dialing address) without having to inform all users of a changed dial address. Users dial the alpha name.

When a user enters an alphanumeric name, the system converts the name to a sequence of digits according to an alphanumeric-dialing table. If the entered name is not found in the table, the system denies the call attempt and the user receives either an `Invalid Address` message (DCP) or a `Wrong Address` message (ISDN-BRI).

Because data terminals access Communication Manager via DCP or ISDN-BRI data modules, dialing procedures vary:

- For DCP, at the `DIAL`: prompt users type the alphanumeric name and press **Enter**.
- For ISDN-BRI, at the **CMD :** prompt users type **d**, a space, and the alphanumeric name, and press **Enter**.

More than one alphanumeric name can see the same digit string.

Administering Alphanumeric Dialing

To set up alphanumeric dialing:

1. On the **Alphanumeric Dialing Table** screen, administer the **Alpha-name** and **Mapped String** fields. See Alphanumeric Dialing Table on page 748 for more information.

Considerations

Note:

Alphanumeric dialing does not apply to endpoints with Hayes modems.

Data Hotline

Data Hotline provides for automatic-nondial placement of a data call preassigned to an endpoint when the originating media server goes off-hook. Use for security purposes.

Administering Data Hotline

To administer a data hotline:

1. You can use an abbreviated dialing list for your default ID. See "Abbreviated Dialing" in *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205 for more information.
2. On the **Station** screen, administer the following fields.
   - Abbreviated Dialing List
   - Special Dialing Option
   - Hot Line Destination
3. On the **Data modules** screen, administer the **Abbreviated Dialing List1** field.

The system automatically places Data Hotline calls to preassigned extensions or off-premises numbers. Calling terminals are connected to the system by a data module. Users should store the destination number in the abbreviated dialing list for future reference.

Interactions

- Call Forwarding — All Calls

  A Data Hotline caller cannot activate both Call Forwarding and Data Hotline. Dialing the Call Forwarding feature access code (FAC) causes activation of the Data Hotline instead.

Data Privacy

Data Privacy protects analog data calls from being disturbed by any of the system's overriding or ringing features.

Administering Data Privacy

To administer data privacy:

1. Choose either of the following:
   - On the **Feature Access Code** screen, administer the **Data Privacy Access Code** field.
   - On the **Class of Service** screen, administer the **Data Privacy** field.
2. On the **Station** screen, administer the **Class of Service** field.

To activate this feature, the user dials the activation code at the beginning of the call.

Considerations

- Data Privacy applies to both voice and data calls. You can activate Data Privacy on Remote Access calls, but not on other incoming trunk calls. Data Privacy is canceled if a user transfers a call, is added to a conference call, is bridged onto a call, or disconnects from a call. You can activate Data Privacy on calls originated from attendant consoles.
- For virtual extensions, assign the Data Privacy Class of Service to the mapped-to physical extension.

Interactions

- Attendant Call Waiting and Call Waiting Termination

  If Data Privacy is active, Call Waiting is denied.
- Bridged Call Appearance — Single-Line Telephone

  If you activate Data Privacy or assign Data Restriction to a station involved in a bridged call and the primary terminal or bridging user attempts to bridge onto the call, this action overrides Data Privacy and Data Restriction.
- Busy Verification

  Busy Verification cannot be active when Data Privacy is active.

- Intercom — Automatic and Dial

  An extension with Data Privacy or Data Restriction active cannot originate an intercom call. The user receives an intercept tone.

- Music-on-Hold Access

  If a user places a call with Data Privacy on hold, the user must withhold Music-on-Hold to prevent the transmission of tones that a connected data service might falsely interpret as a data transmission.

- Priority Calls

  If a user activates Data Privacy, Priority Calls are denied on analog telephones. However, Priority Calls appear on the next available line appearance on multiappearance telephones.

Default Dialing

Default Dialing provides data-terminal users who dial a specific number the majority of the time a very simple method of dialing that number. Normal data terminal dialing and alphanumeric dialing are unaffected.

Default Dialing enhances data terminal (keyboard) dialing by allowing a data-terminal user to place a data call to a preadministered destination by either pressing a **Enter** at the DIAL: prompt (for data terminals using DCP data modules) or typing **d** and pressing **Enter** at the CMD: prompt (for data terminals using ISDN-BRI data modules). The data-terminal user with a DCP data module can place calls to other destinations by entering the complete address after the DIAL: prompt (normal data terminal dialing or alphanumeric dialing). The data-terminal user with an ISDN-BRI data module can place calls to other destinations by typing **d**, a space, the complete address, and press **Enter** after the CMD: prompt.

Note:

DU-type hunt groups connecting the system to a terminal server on a host computer have hunt-group extensions set to **no** keyboard dialing.

For the AT command interface supported by the 7400A/7400B/8400B data module, to dial the default destination, enter the ATD command (rather than press return).

Administering Default Dialing

To administer default dialing:

1. You can use an abbreviated dialing list for your default ID. See "Abbreviated Dialing" in *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205, for more information.
2. On the **Data modules** screen, administer the following fields:
   - **Special Dialing Option** as default.
   - **Abbreviated Dialing List**, enter the list to use.
   - **AD Dial Code**.

Data Restriction

Data Restriction protects analog-data calls from being disturbed by any of the system's overriding or ringing features or system-generated tones.

Data Restriction applies to both voice and data calls.

Once you administer Data Restriction for an analog or multiappearance telephone or trunk group, the feature is active on all calls to or from the terminal or trunk group.

Note:

Do not assign Data Restriction to attendant consoles.

Administering Data Restriction

To administer data restriction:

1. On the **Station** screen, set the **Data Restriction** field to **y**.
2. Choose one of the following trunk groups and set the **Data Restriction** field to **y**. See Trunk Group on page 1534 for more information.
   - Access
   - Advanced Private-Line Termination (APLT)
   - Circuit Pack (CP)
   - Customer-Premises Equipment (CPE)
   - Direct Inward Dialing (DID)

- Foreign Exchange (FX)
- Integrated Services Digital Network-Primary Rate Interface (ISDN-PRI)
- Release-Link Trunk (RLT)
- Tandem
- Tie
- Wide Area Telecommunications Service (WATS)

Interactions - Data Restriction

- Attendant Call Waiting and Call Waiting Termination

  If Data Restriction is active, Call Waiting is denied.

- Busy Verification

  Busy Verification cannot be active when Data Restriction is active.

- Intercom — Automatic and Dial

  An extension with Data Privacy or Data Restriction activated cannot originate an intercom call. The user receives an Intercept tone.

- Music-on-Hold Access

  If a user places a call with Data Restriction on hold, The user must withhold Music-on-Hold to prevent the transmission of tones that a connected data service might falsely interpret as a data transmission.

- Priority Calls

  Priority Calls are allowed if the analog station is idle. Call Waiting (including Priority Call Waiting) is denied if the station is busy. However, Priority Calls appear on the next available line appearance on multiappearance telephones.

- Service Observing

  A data-restricted call cannot be service observed.

Data-Only Off-Premises Extensions

Data-Only Off-Premises Extensions allows users to make data calls involving data communications equipment (DCE) or digital terminal equipment (DTE) located remotely from the system site.

A Data-Only Off-Premises Extension uses an on-premises modular trunk data module (MTDM). The system communicates with remote data equipment through the private-line facility linking the on-premises MTDM and the remote data equipment.

Users can place data calls to this type of data endpoint using Telephone Dialing or Data Terminal (Keyboard) Dialing. Since there is no telephone at the remote site, originate data calls from the remote data terminal using Keyboard Dialing only.

Administering Data-Only Off-Premises Extensions

To administer data-only off-premises extensions:

1. On the **Processor/Trunk Data Module** screen, administer all fields.

   See Data modules on page 873 for more information.

Considerations

- The system does not support communications between two TDMs. Modem Pooling is similar to a TDM, it cannot be used on calls to or from a Data-Only Off-Premises Extension.

Interactions

- Telephone Dialing

  An on-premises multiappearance telephone may have a Data Extension button associated with the TDM used for a Data-Only Off-Premises Extension. The telephone user and the remote user share control of the data module. Actions of the user at the telephone may affect the remote user.

  - 1-Button Transfer to Data

    The telephone user can transfer a call to the Data-Only Off-Premises Extension. The Data Extension button lamp on the telephone lights and the Call in Progress lamp on the data module lights during a data call.

- Data Call Preindication

  The multiappearance telephone user presses the idle associated Data Extension button to reserve a data module. The data module is busy to all other users. When the user reserves a data module, the lamp associated with the Data Extension button winks and lights at any other associated telephones. A remote user receives the BUSY message when attempting to originate a call.

- Return-to-Voice

  To establish a data call, the telephone user presses the associated busy Data Extension button to transfer the call to the telephone. The data module associated with the Data Extension button is disconnected from the call. The Call in Progress lamp on the data module goes dark.

Data Modules — General

A Data Module is a connection device between a basic-rate interface (BRI) or DCP interface of the media server and DTE or DCE.

The following types of data modules can be used with the system:

- Announcement data module
- Data line data module
- Processor/trunk data module (P/TDM)
- Netcon data module (DEFINITY Server SI configurations only).
- Processor interface data module (DEFINITY Server SI configurations only).
- System port data module (DEFINITY Server R configurations only).
- X.25 data module (DEFINITY Server R configurations only).
- 7500 data module
- World Class BRI data module
- Ethernet data module. See for more information.
- Point-to-Point Protocol (PPP) data module.

For more information, see *Administration for Network Connectivity for Avaya Communication Manager*, 555-233-504.

Note:

The 51X series Business Communications Terminals (BCT) are not administered on the Data Module screen. The 510 BCT (equivalent to a 7405D with a display and built-in DTDM), 515 BCT (equivalent to a 7403D integrated with 7405D display module function, data terminal and built-in DTDM), and the 7505D, 7506D, and 7507D have a DCP interface but have built-in data module functionality. Both are administered by means of the **Station** screen in Communication Manager.

Detailed description of data modules

TTI allows data modules without hardware translation to merge with an appropriate data module connected to an unadministered port. The unadministered port is given TTI default translation sufficient to allow a terminal connected to the data module (connected to the port) to request a TTI merge with the extension of a data module administered without hardware translation.

Note:

TTI is not useful for Announcement and X.25 hardware.

Administration Without Hardware supports PDM, TDM, Data-Line, Announcement, and X.25 data modules.

Note:

The 513 BCT has an EIA interface rather than a DCP interface (no built in data module, attachable telephone, or telephone features). The 513 BCT is not administered; only the data module to which the 513 BCT is connected is administered.

7400A/7400B+/8400B+ Data Module

Use the 7400A data module instead of an MTDM when you support combined Modem Pooling. The 7400A data module supports asynchronous operation at speeds up to 19200-bps, and provides a DCP interface to the server and an EIA 232C interface to the associated modem. The 7400A operates in stand-alone mode as a data module.

7400B+ and 8400B+ data modules support asynchronous-data communications and operate in stand-alone mode for data-only service or in linked mode, which provides simultaneous voice and data service. The 7400B+ and 8400B+ provide voice and data communications to 7400D series phones and 602A1 CALLMASTER phones that have a connection to a data terminal or personal computer. The data modules integrate data and voice into the DCP protocol required to interface with the server via a port on a digital-line circuit pack. Use the 7400B+ or 8400B+ instead of an MPDM when you need asynchronous operation at speeds up to 19.2-kbps to provide a DCP interface to the server for data terminals and printers. The 7400B+ and 8400B+ do not support synchronous operation and keyboard dialing. Dialing is provided using the standard Hayes command set.

7400D

This data module supports synchronous operation with AUDIX, CMS, and DCS. It provides synchronous data transmissions at speeds of 19.2-Kbps full duplex.

7400C High Speed Link

The 7400C high-speed link (HSL) is a data-service unit that allows access to DCP data services. It provides synchronous data transmission at speeds of 56- and 64-Kbps and provides a link to high-speed data networks. Used for Group 4 fax applications that include electronic mail and messaging, and electronic storage of printed documents and graphics. Use the 7400C for video teleconferencing and LAN interconnect applications.

7500 Data Modules

The 7500 Data Module connects DTE or DCE to the ISDN network. The 7500 Data Module supports EIA 232C and V.35 interfaces and RS-366 automatic-calling unit interface (for the EIA 232C interface only).

The 7500 has no voice functions. Configure in the following ways:

- Asynchronous DCE

  300, 1200, 2400, 4800, 9600, 19200-bps

- Synchronous DCE

  1200, 2400, 4800, 9600, 19200, 56000, 64000-bps

- Asynchronous DTE (used for modem pooling)

  up to 19200-bps

The 7500 Data Module is stand-alone or in a multiple-mount housing.

Asynchronous Data Module

Note:

The **alias station** command cannot be used to alias data modules.

Use the Asynchronous Data Module (ADM) with asynchronous DTEs as a data stand for the 7500 and 8500 Series of ISDN-BRI phones, thus providing connection to the ISDN network. The ADM provides integrated voice and data on the same telephone and supports data rates of 300, 1200, 2400, 4800, 9600, and 19200-bps. This module also supports the Hayes command set, providing compatibility with PC communications packages.

Administered Connection

An Administered Connection (AC) is a connection between two access or data endpoints. Avaya Communication Manager automatically establishes and maintains the connection based on the attributes you administer. ACs provides the following capabilities.

- Support of both permanent and scheduled connections
- Auto Restoration (preserving the active session) for connections routed over Software Defined Data Network (SDDN) trunks
- Administrable retry interval (from 1 to 60 minutes) per AC
- Administrable alarm strategy per AC
- Establishment/retry/auto restoration order based on administered priority

Detailed description

Establish an AC between the following:

- Two endpoints on the same Avaya DEFINITY server or media server
- Two endpoints in the same private network, but on different servers
- One endpoint on the controlling server and another endpoint off the private network

In all configurations, administer the AC on the server having the originating endpoint. For an AC in a private network, if the two endpoints are on two different servers, normally the connection routes via Automatic Alternate Routing (AAR) through tie trunks (ISDN, DS1, or analog tie trunks) and intermediate servers. If required, route the connection via Automatic Route Selection (ARS) and Generalized Route Selection (GRS) through the public network. The call routes over associated ISDN trunks. When the far-end answers, a connection occurs between the far-end and the near-end extension in the `Originator` field on the AC screen.

Because the system makes an administered connection automatically, you do not use the following:

- Data Call Setup

  Do not assign a default dialing destination to a data module when it is used in an AC.

- Data Hotline

  Do not assign a hotline destination to a data module that is used in an AC.

- Terminal Dialing

  Turn off terminal dialing for data modules involved in an AC. This prevents display of call-processing messages (INCOMING CALL,...) on the terminal.

Access endpoints

Access endpoints are non-signaling trunk ports. They neither generate signaling to the far-end of the trunk nor respond to signaling from the far-end. Designate an access endpoint as the originating endpoint or destination endpoint in an AC.

Typical AC applications

The following are typical AC applications:

- A local data endpoint connection to a local or remote-access endpoint. Examples: an MPDM ACCUNET digital service connecting to SDDN via an ISDN trunk-group DS1 port; an MPDM ACCUNET digital service connecting to an ACCUNET Switched 56 Service via a DS1 port.
- A local-access endpoint connecting to a local or remote-access endpoint. Examples: a DSO cross-connect and a 4-wire leased-line modem to a 4-wire modem connection via an analog tie trunk.
- A local data endpoint connecting to a local or remote data endpoint such as a connection between two 3270 data modules.

Note:

The following guidelines do not include AAR and ARS, or GRS administration information for routing AC calls over trunk groups. See the respective feature elsewhere in this book for that information.

Establishing Administered Connections

The originating server or media server attempts to establish an AC only if one of the following conditions exist:

- AC is active.
- AC is due to be active (either a permanent AC or time-of-day requirements are satisfied if it is a scheduled AC).
- Originating endpoint is in in-service or idle state.

If the originating endpoint is not in service or is idle, no activity takes place for the AC until the endpoint transitions to the desired state. The originating server uses the destination address to route the call to the desired endpoint. When the server establishes two or more ACs at the same time, Communication Manager arranges the connections in order of priority.

AC attempts can fail for the following reasons:

- Resources are unavailable to route to the destination.
- A required conversion resource is not available.

- Access is denied by class of restriction (COR), facilities restriction level (FRL), or bearer capability class (BCC). Or, an attempt is made to route voice-band-data over SDDN trunks in the public switch network.
- Destination address is incorrect.
- Destination endpoint is busy.
- Other network or signaling failures occur.

In the event of a failure, an error is entered into the error log, which generates an alarm, if it is warranted by your alarming strategy. You can display AC failures via the **status-administered connection** command.

As long as an AC is due to be active, the originating server continues to establish an AC unless the attempt fails because of an administrative error (for example, a wrong number) or service-blocking condition (for example, outgoing calls barred).

- The frequency with which failed attempts are retried is determined by the administered retry interval (1 to 60 minutes) for each AC.
- Retries are made after the retry interval has elapsed regardless of the restorable attribute of the AC.
- ACs are retried in priority order.
- When you change the time of day on Communication Manager, an attempt is made to establish all ACs in the waiting-for-retry state.

Dropping Administered Connections

An AC remains active until one of the following occurs:

- The AC is changed, disabled, or removed.
- The time-of-day requirements of a scheduled AC are no longer satisfied.
- One of the endpoints drops the connection. This could be because of user action (in the case of a data endpoint), maintenance activity resulting from an endpoint failure, busying out of the endpoint, or handshake failure. If the endpoints are incompatible, the connection is successful until handshake failure occurs.

  Note:

  An AC between access endpoints remains connected even if the attached access equipment fails to handshake.

- An interruption (for example, facility failure) occurs between the endpoints.

If an AC drops because it was disabled/removed or is no longer due to be active, no action is taken. If an AC drops because of changed AC attributes, an immediate attempt is made to establish the connection with the changed attributes if it is still due to be active. Existing entries in the error/alarm log are resolved if they no longer apply. If handshake failure resulted in the dropping of the connection, in the case of an AC involving at least one data endpoint, no action is taken for that AC until the **change administered-connection** command is executed.

Administered Connections failure: Auto Restoration and Fast Retry

When an active AC drops prematurely, you must invoke either auto restoration or fast retry to determine whether auto restoration is attempted for an active AC.

If you option AC for auto restoration and the connection was routed over SDDN trunks, auto restoration is attempted. During restoration, connections are maintained between the server and both endpoints. In addition to allowing the active session to be maintained, AC also provides a high level of security by prohibiting other connections from intervening in active sessions. Auto restoration generally completes before the 60-second endpoint holdover interval. If auto restoration is successful, the call might be maintained (no guarantee). The restoration is transparent to the user with the exception of a temporary disruption of service while restoration is in progress. A successful restoration is reflected by the *restored* state on the status AC screen. Although the restoration was successful, the data session may not have been preserved.

If auto restoration is not active or if the AC is not routed over SDDN trunks, Communication Manager immediately attempts to reestablish the connection (fast retry). Communication Manager also attempts a retry if the originating endpoint initiated the drop. With fast retry, connections are not maintained on both ends. Fast Retry is not attempted for an AC that was last established via fast retry, unless the AC is active for at least two minutes.

If auto restoration or fast retry fails to restore or reestablish the connection, the call drops and the AC goes into retry mode. Retry attempts continue, at the administered retry interval, as long as the AC is due to be active.

Administering Administered Connections

To administer administered connections:

1. Choose one of the following data modules and administer all fields:
   - Data Line Data Module (use with Data Line circuit pack)
   - Processor/Trunk Data Module (use with one of the following:)

MPDMs, 700D, 7400B, 7400D, or 8400B MTDMs, 700B, 700C, 700E, or 7400A

- Processor Interface Data Module (for more information, see *Administration for Network Connectivity for Avaya Communication Manager*, 555-233-504)
- X.25 Data Module (for more information, see *Administration for Network Connectivity for Avaya Communication Manager*, 555-233-504)
- 7500 Data Module (use with ISDN Line 12-BRI-S-NT or ISDN Line 12-BRI-U-NT circuit pack)
- World Class Core BRI Data Module (use with wcbri)

2. On the **DS1 Circuit Pack** screen, administer all fields. Use with switch node carriers. See DS1 Circuit Pack on page 928 for more information.
3. On the **Access Endpoint** screen, administer all fields. See Access Endpoint on page 730 for more information.
4. Choose one of the following trunk groups and administer all fields. See Trunk Group on page 1534 for more information.
   - ISDN-BRI
   - ISDN-PRI
   - Tie
5. On the **Class of Restriction** screen, administer all fields. See Class of Restriction on page 813 for more information.
6. On the **Class of Service** screen, administer all fields. See Class of Service on page 830 for more information.
7. On the **Dial Plan Record** screen, administer the **Local Node Number** field with a number from 1-63 that matches the DCS switch node number and the CDR node number. See DCS to QSIG TSC Gateway screen on page 914 for more information.
8. On the **Administered Connection** screen, administer all fields. See Administered Connection on page 735 for more information.
9. On the **Station** screen, assign one button as ac-alarm. See Station on page 1409 for more information.
10. On the **Attendant Console** screen, assign one button as ac-alarm. See Attendant Console on page 759 for more information.

Interactions

- Abbreviated Dialing

  Use Abbreviated Dialing entries in the `Destination` field. Entries must comply with restrictions.

- Busy Verification of Stations and Trunks

  This feature does not apply to access endpoints because they are used only for data.

- Call Detail Recording

  For an AC that uses a trunk when CDR is active, the origination extension is the originator of the call.

- Class of Restriction

  Reserve a COR for AC endpoints and SDDN trunks. This restricts endpoints that are not involved in AC from connecting to SDDN trunks or endpoints involved in AC.

- Class of Service/Call Forwarding

  Assign to an AC endpoint a COS that blocks Call Forwarding activation at the endpoint.

- Digital Multiplexed Interface (DMI)

  Use DMI endpoints as the destination in an AC. DMI endpoints do not have associated extensions, so do not use them as the originator in an AC.

- Facility Test Calls

  The feature does not apply to access endpoints because an access endpoint acts as an endpoint rather than as a trunk.

- Modem Pooling

  If you require a modem in an AC, one is inserted automatically. If no modem is available, the connection is dropped.

- Non-Facility Associated Signaling (NFAS) and D-Channel Backup

  Auto restoration for an AC that is initially routed over an NFAS facility may fail if the only backup route is over the facility on which the backup D-channel is administered. The backup D-channel may not come into service in time to handle the restoration attempt.

- Set Time Command

  When you change the system time via the set time command, all scheduled ACs are examined. If the time change causes an active AC to be outside its scheduled period, the AC is dropped. If the time change causes an inactive AC to be within its scheduled period, Communication Manager attempts to establish the AC.

  If any AC (scheduled or continuous) is in retry mode and the system time changes, Communication Manager attempts to establish the AC.

- System Measurements

  Access endpoints are not measured. All other trunks in an AC are measured as usual.

Modem Pooling

Modem Pooling allows switched connections between digital-data endpoints (data modules) and analog-data endpoints via pods of acoustic-coupled modems. The analog-data endpoint is either a trunk or a line circuit.

Data transmission between a digital data endpoint and an analog endpoint requires conversion via a modem, because the DCP format used by the data module is not compatible with the modulated signals of an analog modem. A modem translates DCP format into modulated signals and vice versa.

Modem Pooling feature provides pools of integrated-conversion modems and combined-conversion modems.

Integrated-conversion modem pools have functionality integrated on the Pooled Modem circuit pack, providing two modems. Each one emulates a TDM cabled to a 212 modem. Integrated are modem pools not available in countries that use A-law companding.

Combined-conversion modem pools are TDMs cabled to any TDM-compatible modem. Combined-conversion modem pools can be used with all systems.

The system can detect the needs for a modem. Data calls from an analog-data endpoint require that the user indicate the need for a modem, because the system considers such calls to be voice calls. Users indicate this need by dialing the data-origination access code field on the **Feature Access Code** screen before dialing the digital-data endpoint.

The system provides a Hold Time parameter to specify the maximum time any modem can be held but not used (while a data call is in queue).

Administering Modem Pooling

For Integrated modem poolings:

1. On the **Modem Pool Group** screen, administer all fields. See Modem Pool Group on page 1287 for more information.
2. On the **Feature Access Code** screen, administer the **Data Origination Access Code** field. See Feature Access Code on page 956 for more information.
3. On the **Data Module** screen, administer all fields. See Data modules on page 873 for more information.

For Combined modem poolings:

1. On the **Modem Pool Group** screen, administer all fields. See Modem Pool Group on page 1287 for more information.
2. On the **Feature Access Code** screen, administer the **Data Origination Access Code** field. See Feature Access Code on page 956 for more information.

Considerations

- On data calls between a data module and an analog-data endpoint, Return-to-Voice releases the modem and returns it to the pool. The telephone user connects to the analog-data endpoint.
- For traffic purposes, the system accumulates data on modem-pooling calls separate from voice calls. Measurements on the pools also accumulate.
- Modem Pooling is not restricted. Queuing for modems is not provided, although calls queued on a hunt group retain reserved modems.
- Avoid mixing modems from different vendors within a combined pool because such modems may differ in transmission characteristics.
- Each data call that uses Modem Pooling uses four time slots (not just two). As a result, heavy usage of Modem Pooling could affect TDM bus-blocking characteristics.
- Tandem switches or servers do not insert a pooled modem. The originating and terminating servers or switches insert a pooled modem.

Interactions

- Call Detail Recording

  Data call CDR records the use of modem pools on trunk calls.

- Data Call Setup

  Data calls to or from a TDM cannot use Modem Pooling.

- Data Privacy and Data Restriction

  The insertion of a modem pool does not turn off Data Privacy or Data Restriction.

- Data-Only Off-Premises Extensions

  Calls to or from a Data-Only Off-Premises Extension cannot use Modem Pooling, when this type of digital-data endpoint uses a TDM.

- DMI Trunks

  If you place a data call from a local analog-data endpoint to a DMI trunk, you must dial the data-origination access code to obtain a modem. Data calls on DMI trunks to local analog-data endpoints automatically obtain modems.

- DS1 Tie Trunk Service

  Connect modems used for Modem Pooling to AVD DS1 tie trunks via Data Terminal Dialing or by dialing the feature-access code for data origination.

PC Interface

The personal computer (PC) Interface consists of the PC/PBX platforms and PC/ISDN Platform product family. These products are used with Avaya Communication Manager to provide users of IBM-compatible PCs fully-integrated voice and data workstation capabilities.

Two groups of different configurations are available for PC Interface: group 1 uses DCP and group 2 uses the ISDN-BRI (Basic Rate Interface) protocol.

The group 1 configurations consist of DCP configurations that use a DCP expansion card in the PC to link to the server or media server. Group 1 (shown in DCP PC interface configuration (Group 1) on page 535) uses the following connections:

- The PC Interface card plugs into an expansion slot on the PC. The card has 2 standard 8-pin modular jacks (line and telephone).
- The digital telephone plugs into the telephone jack on the PC Interface card.
- The line jack on the card provides a digital port connection to Avaya DEFINITY servers.
- The distance between the PC Interface card and the PBX should be no more than 1524m for 24-gauge wire or 1219m for 26-gauge wire.

Figure 196: DCP PC interface configuration (Group 1)

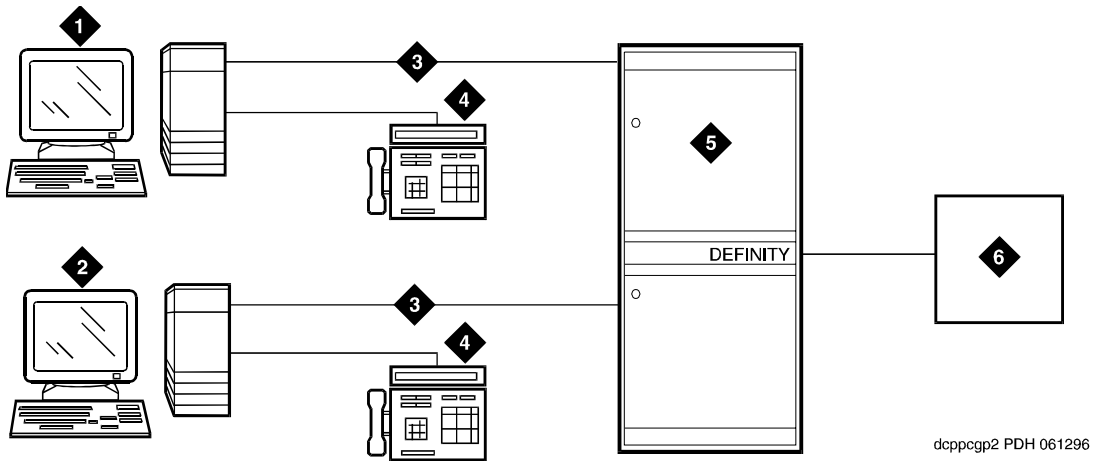


Figure notes:

1. **IBM-compatible PC with DCP Interface card**
2. **IBM-compatible PC with DCP Interface card**
3. **DCP**
4. **DCP telephone**
5. **Avaya (Digital Line, Digital Line (16-DCP-2-Wire), or Digital Line (24-DCP-2-wire) circuit pack)**
6. **Host**

The group 2 configurations link to the server using a PC/ISDN Interface card installed in the PC. This group can include a stand-alone PC terminal, or up to 4 telephones, handsets, or headsets. Group 2 (shown in Figure 197) uses PC/ISDN Interface cards (up to four cards) which plug into expansion slots on the PC. These cards each provide 2 standard 8-pin modular-jack connections for both line connections (to the server or media server) and telephone connections. A standard 4-pin modular jack is also available for use with a handset or headset.

Figure 197: ISDN—BRI PC interface configuration (Group 2)

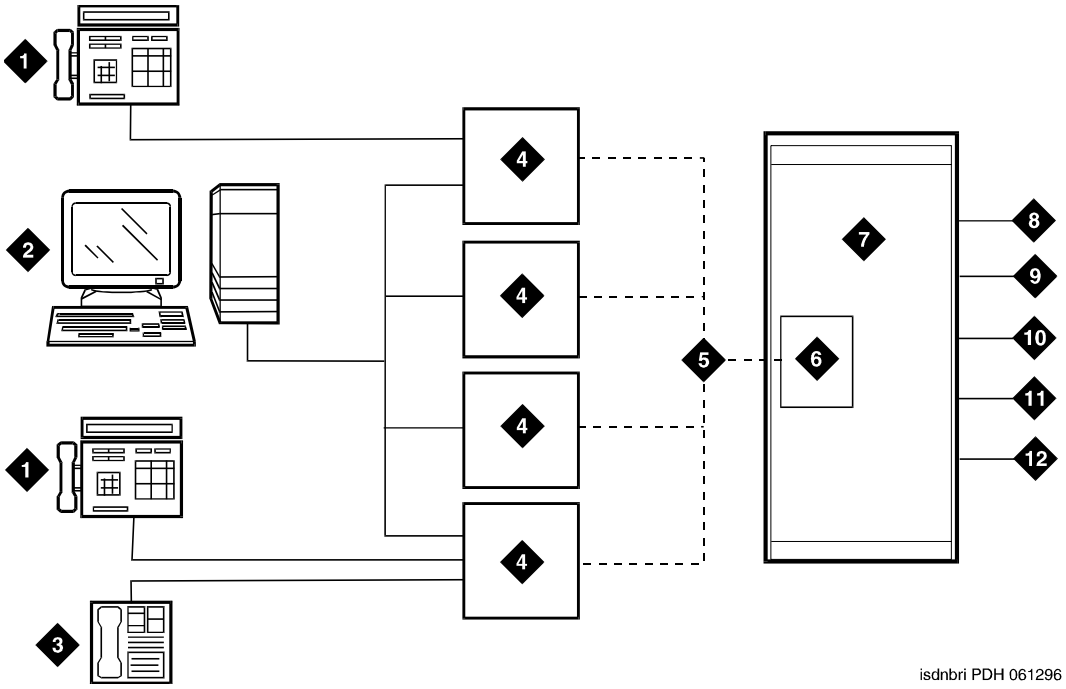


Figure notes:

1. **ISDN telephone**
2. **PC with application**
3. **Handset or Headset**
4. **BRI Interface card**
5. **2B + D**
6. **ISDN Line (12-BRI-S-NT) circuit pack)**
7. **Avaya Media Server**
8. **PRI trunks**
9. **BRI stations**
10. **Interworking**
11. **DMI**
12. **Switch features**

PC Interface users have multiple appearances (depending on the software application used) for their assigned extension. Designate one or more of these appearances for use with data calls. With the ISDN-BRI version, you can use up to 4 separate PC/ISDN Interface cards on the same PC. Assign each card a separate extension, and assign each extension one or more appearances. The availability of specific features depends on the COS of the extension and the COS for Communication Manager. Modem Pooling is provided to ensure general availability of off-net data-calling services.

Security

There are two areas where unauthorized use may occur with this feature: unauthorized local use and remote access.

SECURITY ALERT:
Unauthorized local use involves unauthorized users who attempt to make calls from a PC. The PC software has a security setting so users can place the PC in Security Mode when it is unattended. You also can assign Automatic Security so that the administration program on the PC is always active and runs in Security Mode. This mode is password-protected.

SECURITY ALERT:
Remote access involves remote access to the PC over a data extension. Remote users can delete or copy PC files with this feature. You can password-protect this feature. See the *Avaya Toll Fraud and Security Handbook* for additional steps to secure your system and to find out about obtaining information regularly about security developments.

Administering a PC interface

To administer a PC interface:

1. On the **Station** screen, set the **Type** field to **pc**.

Considerations

- Use the Function Key Module of the 7405D with PC Interface.
- BRI terminals normally are initializing terminals and require you to assign an SPID. The PC/ISDN Platform (Group 2), in a stand-alone configuration, is a non-initializing BRI terminal and does not require you to assign a SPID.
  - Set a locally-defined terminal type with General Terminal Administration
  - Define the terminal type as a non-initializing terminal that does not support Management Information Messages (MIM).
  - Assign the PC/ISDN Platform with an associated (initializing) ISDN-BRI telephone (such as an ISDN 7505) using a SPID.
  - Assign the station (using a locally-defined terminal type) to take full advantage of the capabilities of the PC Interface. This terminal type is also non-initializing with no support of MIMs.

- Do not use telephones with data modules with the PC Interface. (You can still use 3270 Data Modules if you also use 3270 emulation). If you attach a DCP data module or ISDN data module to a telephone that is connected to a PC Interface card, the data module is bypassed (not used). All the interface functions are performed by the interface card even if a data module is present.
- The 7404D telephone with messaging cartridge cannot be used with PC Interface. However, the 7404D with PC cartridge can be used, but only with Group 1 configurations.

Wideband Switching

Wideband Switching provides the ability to dedicate 2 or more ISDN-PRI B-channels or DS0 endpoints for applications that require large bandwidth. It provides high-speed end-to-end communication between endpoints where dedicated facilities are not economic or appropriate. ISDN-BRI trunks do not support wideband switching.

Wideband Switching supports:

- High-speed video conferencing
- WAN disaster recovery
- Scheduled batch processing (for example, nightly file transfers)
- LAN interconnections and imaging
- Other applications involving high-speed data transmission, video transmission, or high bandwidth

Detailed description

ISDN-PRI divides a T1 or E1 trunk into 24 (32 for E1) channels, where one channel is used for signaling, and all others for standard narrowband communication. Certain applications, like video conferencing, require greater bandwidth. You can combine several narrowband channels into one wideband channel to accommodate the extra bandwidth requirement. Avaya Communication Manager serves as a gateway to many types of high-bandwidth traffic. In addition, DS1 Converter circuit packs are used for wideband switching at DS1 remote EPN locations. They are compatible with both a 24-channel T1 and 32-channel E1 facility (transmission equipment). They support circuit-switched wideband connections (NxDS0) and a 192 Kbps packet channel.

The following table provides information on Wideband Switching channel types.

Channel Type	Number of Channels (DSOs)	Data Rate
H0 (T1 or E1)	6 (grouped 4 (T1) or 5 (E1) quadrants of 6 B-channels each)	384 Kbps
H11 (T1 or E1)	24 (on T1 - all 24 B-channels, with the D-channel not used; on E1 - B-channels 1 to 15, and 17 to 25, and B-channels 26 to 31 unused)	1536 Kbps
H12 (E1 only)	30 (B-channels 1 to 15 and 17 to 31)	1920 Kbps
NxDS0 (T1)	2-24	128 to 1536 Kbps
NxDS0 (E1)	2-31	128 to 1984 Kbps

Channel allocation

For standard narrowband communication, ISDN-PRI divides a T1 or E1 trunk as follows:

- T1 trunks are divided into 23 information channels are 1 signaling channel
- E1 trunks are divided into 30 information channels, 1 signaling channel, and 1 framing channel

Certain applications, like video conferencing, require greater bandwidth. You can combine several narrowband channels into one wideband channel to accommodate the extra bandwidth requirement. Avaya Communication Manager serves as a gateway to many types of high-bandwidth traffic. In addition, DS1 converters are used for wideband switching at remote locations.

Performed using one of the three allocation algorithms: fixed, flexible, or floating.

- Fixed allocation — Provides contiguous-channel aggregation. The starting channel is constrained to a predetermined starting point. (Used only for H0, H11, and H12 calls.)
- Flexible allocation — Allows a wideband call to occupy non-contiguous positions within a single T1 or E1 facility (NxDS0).
- Floating allocation — Enforces contiguous-channel aggregation. The starting channel is not constrained to a predetermined starting point (NxDS0).

Typical uses

A typical video application uses an ISDN-PRI interface to DS0 1 through 6 of the line-side facility. See Figure 198.

Figure 198: Wideband Switching Video Application

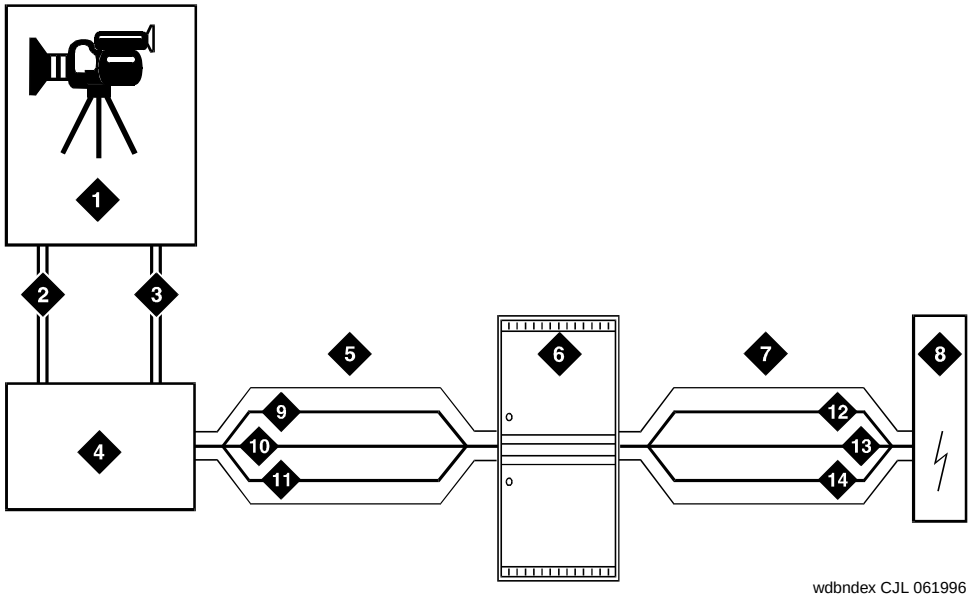


Figure notes:

1. **Video application**
2. **Port 1**
3. **Port 2**
4. **ISDN terminal adaptor**
5. **Line-side ISDN-PRI**
6. **Avaya Media Server**
7. **ISDN trunk**
8. **Network**
9. **DS0 24 D-channel**
10. **DS0 23 unused**
11. **DS0 1 to 6 wideband**
12. **DS0 24 D-channel**
13. **DS0 7 to 23 narrow bands**
14. **DS0 1 to 6 wideband**

Endpoint applications

An endpoint application is the origination or destination of a wideband call. Endpoint application can be any number of data applications based on the customer's particular needs.

ISDN-PRI terminal adapters

For wideband switching with non-ISDN-PRI equipment, you can use an ISDN-PRI terminal adapter. ISDN-PRI terminal adapters translate standard ISDN signaling into a form that can be used by the endpoint application and vice versa. The terminal adapter also must adhere to the PRI-endpoint boundaries as administered on Avaya Communication Manager when handling both incoming (to the endpoint) applications and outgoing calls.

Line-side (T1 or E1) ISDN-PRI facility

A line-side ISDN-PRI (T1 or E1) facility is comprised of a group of DS0s (24 for a T1 facility and 32 for an E1 facility). In this context, these DS0s are also called channels. T1 facilities have 23 B-channels and a single D-channel. E1 facilities have 30 B-channels, 1 D-channel, and a framing channel. Data flows bi-directionally across the facility between the server and the ISDN-PRI terminal adapter.

PRI-endpoints

A PRI-endpoint (PE) is a combination of DS0 B-channels on a line-side ISDN-PRI facility that has been assigned an extension.

A PRI-endpoint can support calls of lower bandwidth. In other words, a PE having a width 6 (six DS0s) can handle a call of one channel (64 Kbps) up to an including 6 channels. For example, an endpoint application connected to a PE defined as using B-channels 1 through 6 of an ISDN-PRI facility could originate a call using B-channels 1, 3, and 5 successfully. If the PE has been administered to use flexible channel allocation, the algorithm for offering a call to the PE starts from the first DS0 administered to the PE. Since only one active call is permitted on a PE, contiguous B-channels always are selected unless one or more B-channels are not in service.

One facility can support multiple separate and distinct PRI-endpoints (several extensions) within a single facility. Non-overlapping contiguous sets of DS0s (B-channels) are associated with each PE.

Universal digital signaling level 1 circuit pack

The UDS1 circuit pack is the interface for line-side and network facilities carrying wideband calls.

Non-signaling configuration

Wideband also can support configurations using non-signaling (non-ISDN-PRI) line-side T1 or E1 connections. The endpoints are the same as those defined for configurations with signaling.

Data service unit/channel service unit

This unit simply passes the call to the endpoint application. Unlike terminal adapters, the DSU/CSU does not have signaling capability.

Note:

No DSU/CSU is needed if the endpoint application has a fractional T1 interface.

Line-side (T1 or E1) facility

This facility, like the ISDN-PRI facility, is composed of a group of DS0s (24 for a T1 facility and 32 for an E1 facility; both T1 and E1 use 2 channels for signaling purposes). Line-side facilities are controlled solely from the server or media server. Through the access-endpoint command, a specific DS0 or group of DS0s is assigned an extension. This individual DS0 or group, along with the extension, is known as a Wideband Access Endpoint (WAE).

Wideband access endpoint

WAEs have no signaling interface to the server or media server. These endpoints simply transmit and receive wideband data when the connection is active.

Note:

Communication Manager can determine if the connection is active, but this doesn't necessarily mean that data is actually coming across the connection.

A WAE is treated as a single endpoint and can support only one call. If all DS0s comprising a wideband access endpoint are in service, then the wideband access endpoint is considered in service. Otherwise, the wideband access endpoint is considered out of service. If an in-service wideband access endpoint has no active calls on its DS0s, it is considered idle. Otherwise, the wideband access endpoint is considered busy.

Multiple WAEs are separate and distinct within the facility and endpoint applications must be administered to send and receive the correct data rate over the correct DS0s. An incoming call at the incorrect data rate is blocked.

Guidelines and examples

This section examines wideband and its components in relation to the following specific customer usage scenarios:

- Data backup connection
- Scheduled batch processing
- Primary data connectivity
- Networking

Data backup connection

Using wideband for data transmission backup provides customers with alternate transmission paths for critical data in the event of primary transmission path failure.

Scheduled batch processing

Scheduled batch processing applications are used for periodic database updates (for example, retail inventory) or distributions (for example, airline fare schedules). These updates are primarily done after business hours and are often referred to as nightly file transfers. Wideband meets the high bandwidth requirements at low cost for scheduled batch processing. In addition, wideband allows the dedicated-access bandwidth for busy-hour switching traffic to be used for these applications after business hours; no additional bandwidth costs are incurred.

The non-ISDN backup data connection is also appropriate for scheduled batch processing applications. Administered Connections are used to schedule daily or weekly sessions originating from this application.

Primary data connectivity

Permanent data connections (those always active during business hours), such as interconnections between LANs, are well suited for Avaya Communication Manager when ISDN-PRI endpoints are used. The ISDN end-to-end monitoring and the endpoint's ability to react to failures provide for critical data availability needs. With ISDN, endpoints can detect network failures and initiate backup connections through the server running Communication Manager; ISDN endpoints can also establish additional calls when extra bandwidth is needed.

Any failures not automatically restored by Avaya Communication Manager are signaled to the endpoint application, which can initiate backup data connections over the same PRI endpoint. Communication Manager routes the backup data connections over alternate facilities if necessary.

Networking

All of the wideband networking is over ISDN-PRI facilities (and the emulation of them by ATM-CES) but may connect to a variety of networks, other domestic interexchange carriers' services, private line, RBOC services, and services in other countries.

ISDN-PRI trunk groups and channel allocation

Only ISDN-PRI trunks (and the emulation of them by ATM-CES) support wideband calls to the network. Wideband's bandwidth requirements have necessitated modification of the algorithms by which trunks look for idle channels. The following section describes the search methods and their relationship to the available wideband data services.

Facility lists

The system always sends a wideband call over a single trunk group and a single DS1 facility (or other ISDN-PRI-capable facility). Since a trunk group may contain channels (trunk members) from several different DS1 facilities, the system maintains a facility list for each trunk group.

A facility list orders the trunk members based on signaling group. If the system is using non-facility associated signaling groups with multiple DS1 facilities, the system sorts trunk members in that signaling group according to the interface identifier assigned to the corresponding DS1 facility.

When searching for available channels for a wideband call placed over a given trunk group, the system starts with the channels in the lowest-numbered signaling group with the lowest interface identifier. If the system cannot find enough channels in a given signaling group with that interface identifier, it checks the next higher interface identifier. If no more interface identifiers are available in the current signaling group, the system moves its search to the channels in the next higher signaling group.

For example, if three facilities having signaling group/interface identifier combinations of 1/1, 1/2, and 2/1 were associated with a trunk group, then a call offered to that trunk group would search those facilities in the order as they were just listed. Also note that since trunks within a given facility can span several trunk groups, a single facility can be associated with several different trunk groups.

Given this facility list concept, the algorithms have the ability to search for trunks, by facility, in an attempt to satisfy the bandwidth requirements of a given wideband call. If one facility does not have enough available bandwidth to support a given call, or it is not used for a given call due to the constraints presented in the following section, then the algorithm searches the next facility in the trunk group for the required bandwidth (if there is more than one facility in the trunk group).

In addition to searching for channels based on facilities and required bandwidth, Port Network (PN) preferential trunk routing is also employed. This PN routing applies within each algorithm at a higher priority than the constraints put on the algorithm by the parameters listed later in this section. In short, all facilities that reside on the same PN as the originating endpoint are searched in an attempt to satisfy the bandwidth of a given call, prior to searching any facilities on another PN.

Direction of trunk/hunting within facilities

You can tell the system to search for available channels in either ascending or descending order. These options help you reduce glare on the channels because the system can search for channels in the opposite direction to that used by the network. If an ISDN trunk group is not optioned for wideband, then a cyclical trunk hunt based on the administration of trunks within the trunk group is still available.

H11

When a trunk group is administered to support H11, the algorithm to satisfy a call requiring 1,536 Kbps of bandwidth uses a fixed allocation scheme. That is, the algorithm searches for an available facility using the following facility-specific channel definitions:

- T1: H11 can only be carried on a facility without a D-channel being signaled in an NFAS arrangement (B-channels 1-24 are used).
- E1: Although the 1,536 Kbps bandwidth could be satisfied using a number of fixed starting points (for example, 1, 2, 3, and so forth), the only fixed starting point being supported is 1. Hence, B-channels 1-15 and 177-25 always are used to carry an H11 call on an E1 facility.

If the algorithm cannot find an available facility within the trunk that meets these constraints, then the call is blocked from using this trunk group. In this case, the call may be routed to a different trunk group preference via Generalized Route Selection (GRS), at which time, based on the wideband options administered on that trunk group, the call would be subject to another hunt algorithm (that is, either the same H11 algorithm or perhaps an N x DS0 algorithm described in a later paragraph).

Note that on a T1 facility, a D-channel is not considered a busy trunk and results in a facility with a D-channel always being partially contaminated. On an E1 facility, however, a D-channel is not considered a busy trunk because H11 and H12 calls may still be placed on that facility; an E1 facility with a D-channel and idle B-channels is considered an idle facility.

H12

Since H12 is 1,920 Kbps which is comprised of 30 B-channels, a 1,920-kbps call can only be carried on an E1 facility. As with H11, the hunt algorithm uses a fixed allocation scheme with channel 1 being the fixed starting point. Hence, an H12 call always is carried on B-channels 1 to 15 and 17 to 31 on an E1 facility (as shown in the following table). When offered any other call (other than a 1,536-kbps call), the algorithm behaves as it does when H11 is optioned.

		DS0s Comprising Each Channel	
Facility	**ISDN Interface**	**H11**	**H12**
T1 T1	23B + D 24B (NFAS)	- 1-24	- -
E1 E1	30B + D 31B (NFAS)	1-15, 17-25 1-15, 17-25	1-15, 17-31 1-15, 17-31

H0

When a trunk group is administered to support H0, the algorithm to satisfy a call requiring 384 Kbps of bandwidth also uses a fixed allocation scheme. Unlike the H11 fixed scheme which only supports a single fixed starting point, the H0 fixed scheme supports 4 (T1) or 5 (E1) starting points. The H0 algorithm searches for an available quadrant within a facility based on the direction of trunk or hunt administered. If the algorithm cannot find an available quadrant within any facility allocated to this trunk group, then the call is blocked from using this trunk group. Again, based on GRS administration, the call may route to a different trunk group preference and be subject to another algorithm based on the wideband options administered.

Note that a D-channel is considered a busy trunk and results in the top most quadrant of a T1, B-channels 19 to 24, always being partially contaminated. This is *not true* for NFAS.

If this H0 optioned trunk group is also administered to support H11, H12, or N x DS0, then the system also attempts to preserve idle facilities. In other words, when offered a narrowband, H0, or N x DS0 call, the system searches partially-contaminated facilities before it searches to idle facilities.

N x DS0

For the N x DS0 multi-rate service, a trunk group parameter determines whether a floating or a flexible trunk allocation scheme is to be used. The algorithm to satisfy an N x DS0 call is either floating or flexible.

- Floating (Contiguous) — In the floating scheme, an N x DS0 call is placed on a contiguous group of B-channels large enough to satisfy the requested bandwidth without any constraint being put on the starting channel (that is, no fixed starting point trunk).

- Flexible — In the flexible scheme, an N x DS0 call is placed on any set of B-channels as long as the requested bandwidth is satisfied. There is absolutely no constraint such as contiguity of B-channels or fixed starting points. Of course, as with all wideband calls, all the B-channels comprising the wideband call must reside on the same ISDN facility.

  Regardless of the allocation scheme employed, the N x DS0 algorithm, like the H11 and H12 algorithms, attempts to preserve idle facilities when offered B, H0, and N x DS0 calls. This is important so that N x DS0 calls, for large values of N, have a better chance of being satisfied by a given trunk group. However, if one of these calls cannot be satisfied by a partially-contaminated facility and an idle facility exists, a trunk on that idle facility is selected, thus contaminating that facility.

  There are additional factors to note regarding specific values of N and the N x DS0 service:

  - N = 1 — this is considered a narrowband call and is treated as any other voice or narrowband-data (B-channel) call.
  - N = 6 — if a trunk group is optioned for both H0 and N x DS0 service, a 384-kbps call offered to that trunk group is treated as an H0 call and the H0 constraints apply. If the H0 constraints cannot be met, then the call is blocked.
  - N = 24 — if a trunk group is optioned for both H11 and N x DS0 service, a 1,536-kbps call offered to that trunk group is treated as an H11 call and the H11 trunk allocation constraints apply.
  - N = 30 — if a trunk group is optioned for both H12 and N x DS0 service, a 1,920-kbps call offered to that trunk group is treated as an H12 call and the H12 trunk allocation constraints apply.

Glare and blocking

Glare prevention

Glare occurs when both sides of an ISDN interface select the same B-channel for call initiation. For example, a user side of an interface selects the B-channel for an outgoing call and, before Communication Manager receives and processes the SETUP message, the server also selects the same B-channel for call origination. Since any single wideband call uses more channels, the chances of glare are greater. With proper and careful administration, glare conditions can be reduced.

To reduce glare probability, the network needs to be administered so both sides of the interface select channels from opposite ends of facilities. This is called linear hunting, ascending or descending. For example, on a 23B+D trunk group, the user side could be administered to select B-channels starting at channel 23 while the network side would be administered to start selecting at channel 1. Using the same example, if channel 22 is active but channel 23 is idle, the user side should select channel 23 for re-use.

Blocking prevention

Blocking occurs when insufficient B-channels required to make a call are available. Narrowband calls require only one channel so blocking is less likely than with wideband calls which require multiple B-channels. Blocking also occurs for wideband calls when bandwidth is not available in the appropriate format (that is, fixed, floating, or flexible).

To reduce blocking, Communication Manager selects trunks for both wideband and narrowband calls to maximize availability of idle fixed channels for H0, H11, and H12 calls and idle floating channels for N x DS0 calls that require a contiguous bandwidth. The strategy for preserving idle channels to minimize blocking depends on the channel type. The chances for blocking are reduced if you use a flexible algorithm, assuming it is supported on the other end.

Channel Type	Blocking Minimization Strategy
H0	Preserve idle quadrants
H11	Preserve idle facilities
H12	Preserve idle facilities
Flexible NxDS0	Preserve idle facilities
Floating NxDS0	Preserve idle facilities as first priority

Administering Wideband Switching

Before you start

You need a DS1 Converter circuit pack. See the *Avaya System Description Pocket Reference* for more information on the circuit pack.

Instructions

To administer wideband switching:

1. On the **Access Endpoint** screen, administer all fields.

   See Access Endpoint on page 730 for more information.

2. On the **PRI Endpoint** screen, administer all fields.

   See on page 1347 for more information.

3. On the **ISDN Trunk Group** screen, administer all fields.

   See Trunk Group on page 1534 for more information.

4. On the **Route Pattern** screen, administer all fields.

   See Route Pattern on page 1367 for more information.

5. (Optional) On the **Fiber Link Administration** screen, administer all fields.

   See Upgrades and Additions for Avaya DEFINITY Server R for more information.

Considerations

- For wideband switching with non-ISDN-PRI equipment, you can use an ISDN-PRI terminal adapter.

Interactions

- Administered Connections

  Provides call initiation for WAEs. All Administered Connections that originate from WAEs use the entire bandwidth administered for WAE. The destination of an Administered Connection can be a PRI endpoint.

- Automatic Circuit Assurance

  Treats wideband calls as logical single-trunk calls so that a single ACA-referral call is made if an ACA-referral call is required. The call is referred to the lowest B-channel associated with the wideband call.

- Call Coverage

  A wideband endpoint extension cannot be administered as a coverage point in a call-coverage path.

- Call Detail Recording

  When CDR is active for the trunk group, all wideband calls generate CDR records. The feature flag indicates a data call and CDR records contain bandwidth and BCC.

- Call Forwarding

  You must block Call Forwarding through Class of Service.

- Call Management System and Basic Call Management System

  Wideband calls can be carried over trunks that are measured by CMS and BCMS. Wideband endpoints are not measured by CMS and BCMS.

- Call Vectoring

  PRI endpoints can use a vector-directory number (VDN) when dialing. For example, PRI endpoint 1001 dials VDN 500. VDN 500 points to Vector 1. Vector 1 can point to other PRI endpoints such as route-to 1002, or route-to 1003, or busy.

  Call Vectoring is used by certain applications. When an incoming wideband call hunts for an available wideband endpoint, the call can route to a VDN, that sends the call to the first available PRI endpoint.

- Class of Restriction

  COR identifies caller and called-party privileges for PRI endpoints. Administer the COR so that account codes are not required. Forced entry of account codes is turned off for wideband endpoints.

- Facility Busy Indication

  You can administer a busy-indicator button for a wideband-endpoint extension, but the button does not accurately track endpoint status.

- Facility Test Calls

  You can use Facility Test Calls to perform loop-back testing of the wideband call facility.

- Generalized Route Selection

  GRS supports wideband BCC to identify wideband calls. GRS searches a route pattern for a preference that has wideband BCC. Route preferences that support wideband BCC also can support other BCCs to allow different call types to share the same trunk group.

- CO Trunk (TTC - Japan) Circuit Pack

  This circuit pack cannot perform wideband switching. No member of the circuit pack should be a member of a wideband group.

CallVisor Adjunct-Switch Application Interface

CallVisor Adjunct-Switch Applications Interface (ASAI) links Avaya Communication Manager and adjunct applications. The interface allows adjunct applications to access switching features and supply routing information to Communication Manager. CallVisor ASAI improves Automatic Call Distribution (ACD) agents' call handling efficiency by allowing an adjunct to monitor, initiate, control, and terminate calls on the media server. The CallVisor ASAI interface may be used for Inbound Call Management (ICM), Outbound Call Management (OCM), and office automation/ messaging applications.

CallVisor ASAI is supported by two transport types. These are:

1. Integrated Services Digital Network (ISDN) Basic Rate Interface (BRI) transport (CallVisor ASAI-BRI)
2. LAN Gateway Transmission Control Protocol/Internet Protocol transport (Avaya LAN Gateway).

CallVisor ASAI messages and procedures are based on the ITU-T Q.932 international standard for supplementary services. The Q.932 Facility Information Element (FIE) carries the CallVisor ASAI requests and responses across the interface. An application program can access CallVisor ASAI services by supporting the ASAI protocol or by using a third-party vendor application programming interface (API).

For a simple ASAI configuration example, see Figure 199.

Figure 199: ASAI Switch Interface Link — BRI Transport

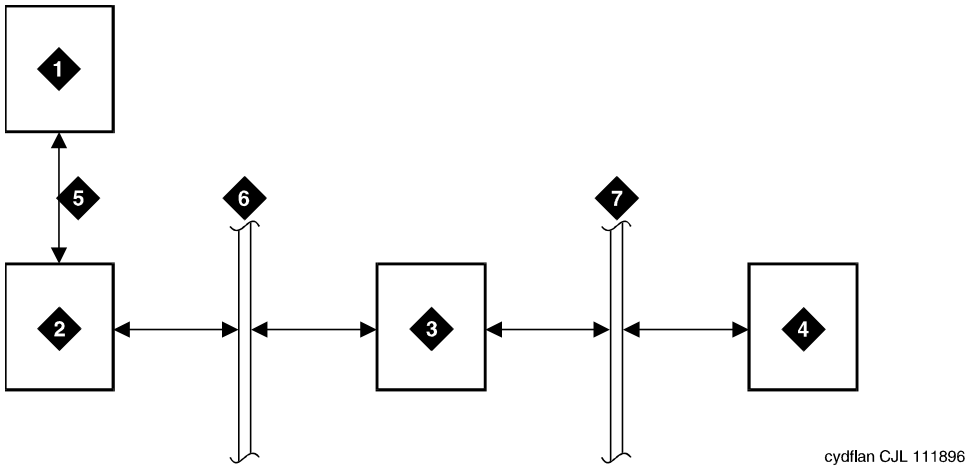


Figure notes:

1. **ASAI adjunct**
2. **ISDN Line circuit pack**
3. **Packet Controller circuit pack**
4. **Switch processing element (SPE)**
5. **ISDN-BRI**
6. **Packet bus**
7. **Memory bus**

ASAI Capabilities

For information concerning the types of associations over which various event reports can be sent, see *Avaya CallVisor ASAI Technical Reference*.

Considerations

- If your system has an expansion cabinet (with or without duplication), ASAI resources should reside on the system's Processor Cabinet.

Interactions

See *Avaya CallVisor ASAI Technical Reference*.

Setting up CallVisor ASAI

CallVisor Adjunct-Switch Applications Interface (ASAI) can be used in the telemarketing and help-desk environments. It is used to allow adjunct applications to monitor and control resources in Avaya Communication Manager.

Before you start

- On the **System-Parameters Customer-Options** screen, verify the:
  - **ASAI Interface** field is **y**. If not, contact your Avaya representative.
  - **ASAI Proprietary Adjunct Links** field is **y** if the adjunct is running the CentreVu Computer Telephony.

Instructions

To set up CallVisor ASAI:

1. Type **`add station`** ***nnnn***, where ***nnnn*** is the extension you want to assign to the ASAI adjunct. Press **Enter**.

   The **Station** screen appears.

```
change station 1014                                          Page 1 of X
                              STATION

Extension:1014               Lock Messages? n                    BCC: 0
     Type:asai               Security Code:                       TN: 1
     Port:                  Coverage Path 1:                      COR: 1
     Name:                  Coverage Path 2:                      COS: 1
                            Hunt-to Station:

STATION OPTIONS
              Loss Group: 2               Personalized Ringing Pattern: 3
            Data Module? n                         Message Lamp Ext: 1014
           Speakerphone: 2-way                  Mute button enabled? y
       Display Language? English                   Expansion Module?
                  Model:
                                                    Media Complex Ext:
                                                         IP Softphone? y
                                                  Remote Office Phone? y
```

2. In the **Type** field, type
   - **asai** if this adjunct platform is other than CentreVu Computer Telephony, for example, IBM CallPath.
   - **adjlk** (Computer Telephony adjunct link) if this is for the CentreVu Computer Telephony using the Telephony Services Application Programming Interface (TSAPI).
3. In the **Port** field, use the port address assigned to the LAN Gateway Interface circuit pack.
4. Press **Enter** to save your changes.

Setting up DEFINITY LAN Gateway

Before you start

This section describes pre-administrative tasks associated with the Co-Resident DLG.

For S8300 Media Server

Although the S8300 Media Server does not use a standard DEFINITY server port board, the pre-administration is the same.

1. Type **`display system-parameters customer-options`** and press **Enter**.
2. Go to page 2 and make sure the **ASAI Link Core Capabilities** and/or **Computer Telephony Adjunct Links** fields are **y**, and the **Co-Res DEFINITY LAN Gateway** field is **y**.
3. Go to page 4 and make sure that the **Processor Ethernet** field is **y**.
4. Type **`display ip-interfaces`** and press **Enter**.

   Make sure the PROCR is administered and its Ethernet port is enabled. If the PROCR is not listed (PROCR should appear in the **Service Type** field and the previously administered node name should appear in the **Node Name** field), add a PROCR.

Administering CTI Links - Summary

Use this information to familiarize yourself with the basic administrative tasks for administering CTI links using the Co-Resident DLG.

- Use the **`display system-parameters customer-options`** command make sure the following fields are set to **y**.
  - **ASAI Link Core Capabilities** and/or **Computer Telephony Adjunct Links**
  - **Co-Res DEFINITY LAN Gateway**
- Use the **`add cti-link`** command to administer a CTI link.
- Use the **`change ip-services`** command and in the **Service Type** field, enter **DLG**.

  When the **Service Type** field is **DLG**, the system adds a DLG Administration page as the last page of the screen.
  - Complete the DLG Administration page to add your client information.

  Note:

  For additional information on DLG administration for the S8300 Media Server, see *Avaya CallVisor ASAI Technical Reference*.

Chapter 16: Administering Media Servers

This chapter describes how to administer an Avaya S8700 or S8710 Media Server and an Avaya G700 Media Gateway and the Avaya S8300 Media Server. It is targeted for system administrators after the product is installed and tested. In a converged network where voice and data are both sent over a corporate local area network (LAN), this configuration can provide primary or standby telephony and communications-processing capabilities.

Users should have broad data networking experience with data products and technology to best understand this product. An in-depth knowledge of the call-processing engine of Avaya Communication Manager and of the Avaya P330 switches is also recommended.

Overview

To set up and maintain your S8300 Media Server with a G700 Media Gateway, you need to administer:

- the G700 Media Gateway and its internal processors, typically using the P330 Device Manager web-based tool or a command-line interface (CLI)
- the S8300 Media Server using the S8300 Media Server Web Interface
- call-processing features using Avaya Communication Manager

For details on administration access methods, tools, and alternatives, see *Welcome to the Avaya S8300 Media Server and Avaya G700 Media Gateway*. Documents mentioned in this chapter are available on the Avaya website: http://www.avaya.com/support.

Administering the G700 Media Gateway

The hardware elements of the G700 Media Gateway are summarized in this section. For details on any of these hardware components or on different configurations, see the *Hardware Guide for Avaya Communication Manager*, 555-245-207.

G700 Media Gateway physical design

(For S8300, S8700, or S8710 Media Servers) The G700 Media Gateway is a19-inch, rack-mount design similar to other Avaya P330 hardware. The media gateway can be a member of a P330 stack of Layer 2 or 3 devices, reside in a stack of other media gateways, or operate as a standalone unit. Figure 200 shows the front of the media gateway.

Figure 200: The G700 Media Gateway (front view)

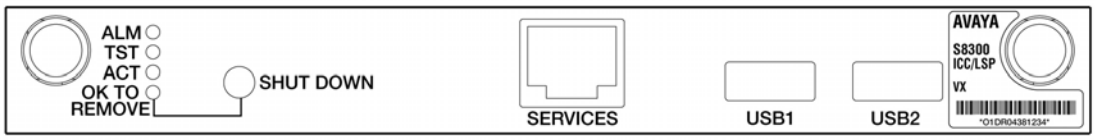
(For S8300, S8700, or S8710 Media Servers) Key components of the G700 Media Gateway include:

1. LED Status Panel (top left) for the media gateway
2. S8300 Media Server slot (V1). The G700 Media Gateway does not have an S8300 Media Server in its basic configuration.
3. P330 Expansion slot (bottom left) for adding modules such as a 16-port Ethernet expansion module, fiber interfaces, ATM access or WAN access
4. Dual 10/100Base-T Ethernet Ports for connection to the corporate LAN (bottom center of chassis)
5. Up to three additional media modules in slots V2 to V4 on the right side of the chassis (slots are numbered from top to bottom)
6. 8-Pin RS-232 Serial Port (Console) for direct-connect P330 stack processor administration (typically done by Services personnel)

S8300 Media Server faceplate

(For S8300 Media Servers) Figure 201 shows the faceplate of the S8300 Media Server.

Figure 201: S8300 Media Server faceplate



The faceplate of the S8300 Media Server has the following components:

- The LED array on the left indicates when the S8300 is active and when it is safe to power it down.
- A Shut Down button can be used to shut down the server (the S8300 can also be shut down and restarted using software commands).
- A Services Ethernet interface provides a direct connection to a laptop computer connected with a crossover cable. Normally only Services technicians use this interface; most ongoing S8300 administration is done using the Ethernet connection to the corporate LAN on the media gateway.
- One of the two USB connections may be used to attach an external modem, primarily used to report alarms to a Services support agency.
- The label on the right provides identification information for the S8300 Media Server.

G700 Media module description

(For S8300 Media Servers) The four slots on the G700 Media Gateway are populated with media modules according to the needs at your site. Up to four slots may be filled as follows:

- The S8300 Media Server is a special type of media module that hosts the call-processing (telephony) software. If present, the S8300 is installed in slot one (V1); only one can be installed per media gateway. The S8300 can act as a primary call controller for the system, or one or more S8300s can be configured to provide standby service for a primary call controller if needed. See Configuring the Local Survivable Processor on page 558 for details.
- (For S8300, S8700, or S8710 Media Servers) The other media module slots are filled as needed to meet the call-processing needs at your site. Media modules can be mixed and matched. Table 1 lists the administrative name and port capacities for the media modules that may be installed in a G700 Media Gateway.

Table 7: Media Module names and port capacities

Media module type	Administration name	Number of ports
Analog Line/Trunk	analog	8
DCP Telephone	dcp	8
T1/E1	ds1	24/32 (T1=24 in robbed-bit signalling mode and E1=32 for 1 control, 1 signal and 30 voice)
VoIP	voip	N/A
S8300 Media Server	icc	N/A
MM270	bri	8

Configuring the Local Survivable Processor

(For S8300, S8700, or S8710 Media Servers) An S8300 Media Server can be configured either as the primary call-processing controller, or as a Local Survivable Processor (LSP). An LSP can take over call processing if the primary call-processing system (such as another S8300, or an S8700 or S8710 Media Server) is unavailable for any reason (such as a network failure or server problem). The S8300 Media Servers can be either the primary or LSP media servers; they are set up to operate as a primary or standby LSP server during the configuration process using the S8300 Media Server Web Interface. The license file determines the mode that the S8300 Media Server runs in and the Configure Server web page provides supplementary instruction.

(For S8300, S8700, or S8710 Media Servers) If the S8300 Media Server loses contact with its G700 Media Gateway, it will take the media gateway out of service after a time-out period. However, if the G700 loses contact with the S8300 Media Server, the media gateway first tries to reestablish contact. If it fails after a time-out period, then the G700 tries to connect with another media server in its controller list. When a functional media server is located, the G700 reboots.

(For S8300, S8700, or S8710 Media Servers) If the primary call-processing server goes offline and an LSP is available as a standby unit, it will assume call processing as follows:

- IP phones and media gateways that were previously using the primary media server will try to register with the standby media server for call processing, provided that they have been administered to do so in the controller list (use the **`set mgc list`** command).

- The standby media server (LSP) will go into license error mode, then start to provide call processing. It cannot preserve any calls set up by the primary media server. IP telephone connections can stay up until the call is completed if they are shuffled, but no features are supported on the call.

  Note:

  The license error mode will run for up to 30 days and if the problem is not resolved, then it goes into no license mode and call processing halts.

- When the primary media server is available again, it will begin handling call processing. However, those endpoints that are now using the standby media server will continue to do so until the standby unit is rebooted.
- If the standby S8300 Media Server is rebooted, all devices will return to using the primary media server for call-processing service. Any calls in progress on the LSP will be dropped when the reboot occurs (the change back to the primary media server is not call preserving).
- With LSP functionality, there is full functionality and feature support.

Using Device Manager to administer G700 Media Gateway components

(For S8300, S8700, or S8710 Media Servers) Device Manager, also known as the P330 Embedded Web Manager, provides a browser-based graphical user interface (GUI) to assist you with ongoing media gateway administration. For complete information on using Device Manager, see the *Avaya User's Guide P330 Manager*.

(For S8300, S8700, or S8710 Media Servers) Device Manager allows you to:

- View the status of different devices on the network.
- Configure or modify devices including Virtual LAN (VLAN) groupings, port mirroring, and Simple Network Management Protocol (SNMP) traps.

Accessing Device Manager from a web browser

(For S8300, S8700, or S8710 Media Servers) To access the P330 Device Manager from a web browser:

1. Open a compatible web browser on your computer. Microsoft Internet Explorer 5.0 (or higher) and Netscape Navigator 4.7 and 6.2 are supported. The Java Plug-in 1.3.1_02 is required.
2. In the **Address (or Location)** field of your browser, type the IP address of the P330 stack.

3. Log in as prompted.
   - The default user name is: **root**
   - The default password for read-write access is: **root**

   Note:

   You should change the default passwords after you log in, if they have not already been changed. The passwords apply to all logins for these devices, whether accessed through Device Manager or a CLI.
4. The **Welcome** screen is displayed. Proceed with P330 and media gateway device administration.

Accessing Device Manager through MultiService Network Manager

(For S8300, S8700, or S8710 Media Servers) You can alternatively launch the P330 Device Manager from the Avaya MultiService (formerly P330View) Network Manager. This optional product is a complete Network Management System (NMS). Services include:

- viewing all network devices by type, subnet, or customized groupings
- logging and viewing SNMP traps and events
- launching and managing other applications including Avaya Site Administration

Contact your Avaya representative to obtain a copy of this program if desired.

Administering the command line interface

(For S8300, S8700, or S8710 Media Servers) Instead of using Device Manager, you could telnet into the Media Gateway Processor (MGP), or use the serial interface on the front of the chassis to establish a direct connection to the P330 stack processor.

- Command line interface (CLI) access procedures are covered in *Welcome to the Avaya G700 Media Gateway controlled by an Avaya S8300 Media Server or an Avaya S8700 Media Server*, 555-234-200.
- For a list of CLI commands, see the *Maintenance for the Avaya G700 Media Gateway controlled by an Avaya S8300 Media Server or an Avaya S8700 Media Server*, 555-234-101.

(For S8300, S8700, or S8710 Media Servers) SNMP alarms are different from server hardware- or software-generated Operations Support System (OSS) alarms that are recorded in the server logs, and may be reported to Avaya's Initialization and Administration System (INADS) or another services support agency over the server's optional modem interface or through SNMP notifications. Either method, both, or no alarm-reporting method may be used at a given site.

Administering the Media Server

(For S8300, S8700, or S8710 Media Servers) The S8300, S8700, or S8710 Media Server contains the call-processing software of Avaya Communication Manager, and controls its operation over the corporate network. Some of the primary functions controlled by the S8300 and S8700 or S8710 Media Server are:

- Backing up and restoring call processing, server, and security data.
- Checking server and process status.
- Administering network features for the S8300, S8700, or S8710 Media Server such as SNMP service, enabling or disabling the modem (if used), enabling FTP services, and installing license and authentication files.
- Installing new software and reconfiguring the server as needed.
- Performing routine diagnostics and troubleshooting such as viewing alarms and system logs, and running tests if needed.

Media Server Web Interface and online help

(For S8300, S8700, or S8710 Media Servers) S8300, S8700, or S8710 Media Server tasks are performed using the Media Server Web Interface. This browser-based server administration tool is an easy-to-use graphical user interface for performing the server administration tasks listed above. It contains an extensive help system that describes all web screens and media server procedures. This section covers highlights of media server administration.

Accessing the Media Server Web Interface

(For S8300, S8700, or S8710 Media Servers) The Web Interface can be accessed through the corporate LAN connection to the G700 Media Gateway, or through the Services Ethernet interface on the front of the S8300, S8700, or S8710 Media Server connected to a laptop PC using a crossover cable. Details on how to configure a laptop for a direct connection are in the online help and in *Welcome to the Avaya S8300 Media Server and Avaya G700 Media Gateway*.

(For S8300, S8700, or S8710 Media Servers) To access the media server, you must log in as follows:

1. Open a compatible Internet browser on your computer. Currently only Internet Explorer 5.x (Avaya recommends 5.5 with Service Pack 2) and Netscape 4.7x are supported.
2. In the **Address (or Location)** field of your browser, type the IP address or name of the Avaya media server and press **Enter**.
   - LAN access by IP address. If you are logging into the administrative interface over the corporate local area network, you can type the media server's unique IP address in standard dotted-decimal notation, such as http://192.152.254.201.
   - LAN access by server name. If the server has already been configured and if the corporate LAN includes a domain name service (DNS) server that has been administered with the servers' names, you can type the server's name into the address field instead of the IP address. Server names vary depending on local administration (such as http://media-server1.mycompany.com).
   - Laptop access by IP address. If you are logging in to the services Ethernet interface from a directly connected laptop, the IP address on the server is always **192.11.13.6**. The subnet mask is always **255.255.255.252**. New servers that have not yet been configured can only be accessed in this way. Server-name login is *not* available through the services interface because this connection is a closed (private) network with no DNS.

   Note:

   The name and IP address of the S8300 Media Server are specified during initial media server configuration.
3. If your browser does not have a valid security certificate, you will see a warning screen and instructions to load the security certificate.
   - If you are certain your connection is secure, accept the server security certificate to access the **Login** screen.
   - If you plan to use this computer and browser to access this or other Avaya media servers again, click the main menu link to Install Avaya Root Certificate after you log in.
4. The **Login** screen appears.
   - In the **Username** field, type your user name (login ID), such as **cust**.
   - Click the **Login** button or press **Enter**.

   Note:

   User names and passwords are case sensitive. Enter the login ID and confirmation information in upper- or lowercase as required.
5. Enter your login confirmation information as prompted:
   - Password prompt. Type your password in the **Password** field, and click **Login** or press **Enter** again.
   - ASG challenge. If your login is Access Security Gateway (ASG) protected, you will see a challenge screen. Enter the correct response and click **Login** or press **Enter**.

6. The server will match your login information against its authentication tables. If the information you entered is recognized as a valid login, the screen displays a two-frame window, with the main menu in the left panel and a usage-agreement notice in the right window.
   - If the main menu in the left panel has only about 10 links, you have accessed the server from a login ID that has restricted permissions. You can adjust the login service levels using the **Login Administration** screen in Avaya Site Administration or SAT.
   - If the server cannot recognize your user name or password, you receive an authentication failure message. Return to step 5. If you fail to enter the user name and login confirmation correctly 4 times within a few minutes, the **Login** screen will block further attempts to log in for a while.
7. When you successfully log in to the server, check the top of the left panel. The Avaya media server you are logged into is identified by name and server number. The S8300 Media Server number is always 1. The S8700 or S8710 Media Server shows two servers.

Web interface main menu

(For S8300, S8700, or S8710 Media Servers) The tasks you can perform are shown by a list of links in the left panel of the main web interface window.

(For S8300, S8700, or S8710 Media Servers) For help with any of these tasks, click the **Help** button on the main menu. Alternatively, if you have already clicked on a link, click the **About This Screen** button on the web interface page to go directly to the help for that specific screen.

Media server web interface tasks

(For S8300, S8700, or S8710 Media Servers) Key tasks that administrators typically perform on the S8300, S8700, or S8710 Media Server are summarized in this section. See the online help for complete information.

Backing up and restoring data

(For S8300, S8700, or S8710 Media Servers) One of the most important tasks is to set up a schedule to routinely back up system data to a safe location. Because the S8300 Media Server does not include a media storage device for backup purposes, the data must be emailed or transferred using file transfer protocol (FTP) to another server or computer for backup. The S8700 or S8710 Media Server has a local removable storage device.

(For S8300, S8700, or S8710 Media Servers) The web interface allows you to back up call-processing data (Avaya Communication Manager "translations"), server system data, and security files. Avaya recommends that you encrypt the backup files to keep this sensitive information secure.

Upgrading software and configuration

(For S8300, S8700, or S8710 Media Servers) Occasionally you may need to install new software on the media server. The new software (and a new license file if required) must be copied to the media server using one of the methods listed in Copying files to the server on page 564. The software installation process uses wizard-like screens on the web to guide you through the process.

(For S8300, S8700, or S8710 Media Servers) You may also need to update your server configuration from time to time, and reverify it after a software upgrade. IP addresses for the server and its gateway to the corporate LAN, or for the optional UPS unit, DNS servers, and modem, are specified using configuration wizard-like screens. The wizard-like screens also allow you to specify static network routes for sending information over the network, and to update or change the method by which the S8300 Media Server keeps time.

Copying files to the server

(For S8300, S8700, or S8710 Media Servers) Files must be copied to the media server from another computer or server in the network, or uploaded from a directly connected laptop computer. Files that may be copied to the server include license and authentication files, system announcements, and files for software upgrades. Files may be copied to the server using one of the following methods:

- Upload Files to Server (via browser) link to upload one or more files from your computer to the server's FTP directory using HTTP protocol.
- Download Files to Server (from web) link to copy files to the server from another server on the network; it works like the **Upload Files** screen.
- Transfer files from another computer or server accessible from the corporate network using FTP or Trivial FTP (TFTP). Files must be transferred in binary mode. Either a GUI or CLI FTP program can be used, depending on what is available on your computer.

Administering SNMP

(For S8300, S8700, or S8710 Media Servers) You can set up Simple Network Management Protocol (SNMP) services on the server to provide a means for a corporate NMS to monitor the server, and send alarm notifications to a services agency, to a corporate NMS, or both.

(For S8300, S8700, or S8710 Media Servers) To activate SNMP alarm notification for the S8300, S8700, or S8710 devices, use the **Configure Trap Destinations** screen to set up SNMP destinations in the corporate NMS. SNMP traps for other devices on the network can be administered using Device Manager; see Using Device Manager to administer G700 Media Gateway components on page 559.

Administering Call-processing

(For S8300, S8700, or S8710 Media Servers) The telephony features of the S8300 Media Server are administered using the same commands and procedures as an S8700 or S8710 Media Server or a legacy DEFINITY Enterprise Communications System.

Accessing Avaya Communication Manager

(For S8300, S8700, or S8710 Media Servers) Avaya Communication Manager resides on the S8300, S8700, or S8710 Media Server. It can be accessed through either Avaya Site Administration (ASA) or the System Access Terminal (SAT) program.

Avaya Site Administration

Avaya Site Administration features a graphical user interface (GUI) that provides access to SAT commands as well as wizard-like screens that provide simplified administration for frequently used features. You can perform most of your day-to-day administration tasks from this interface such as adding or removing users and telephony devices. You can also schedule tasks to run at a non-peak usage time. ASA is available in several languages.

The S8300, S8700, or S8710 Media Server can be used to download Avaya Site Administration. A downloadable version of this package may be accessed through the S8300, S8700, or S8710 Media Server Web Interface. This software must be installed on a computer running a compatible Microsoft Windows operating system such as Windows 95, 98, NT 4.0, Millennium Edition, and Windows 2000. Once installed, it can be launched from a desktop icon, from the P330 Device Manager, or through a link in the S8300 Media Server Web Interface.

System Access Terminal

The System Access Terminal (SAT) program uses a Command Line Interface (CLI) interface for telephony administration. SAT is available through the Avaya Site Administration package.

Security Considerations

(For S8300, S8700, or S8710 Media Servers) Levels of security for administration of the G700 Media Gateway are the same as traditionally for Avaya Communication Manager. This means that administration login passwords are passed in plain text with no encryption. Exceptions to this no-encryption policy include:

- The ASG program that is installed on all Avaya media servers.
- An encrypted web interface to the S8300 Media Server (see the security certificate information in the media server online help)

- Optional encryption for data backups (see Backing up and restoring data on page 563).
- Support for RADIUS authentication for media gateways and P330 stack elements using the P330 Device Manager. See Using Device Manager to administer G700 Media Gateway components on page 559.

Command syntax changes for media modules

(For S8300, S8700, or S8710 Media Servers) The syntax for using the SAT commands for a G700 Media Gateway or S8300 Media Server has changed. In a traditional DEFINITY system, ports are identified by the cabinet number, carrier, slot, and port. For example: 02A0704

(For S8300, S8700, or S8710 Media Servers) Because this numbering convention does not make sense for media modules, a new convention was developed. The numbering convention for the media modules uses the same seven-character field as does a traditional system, but the fields represent the media gateway number, media module slot (V1 to V4), and port number (00 to 99 are supported; the actual number of ports that can be specified depends on the type of media module).

Example: 001V205

In this example, the 001 represents the media gateway number, the V2 represents the slot number (possibly V1 through V4), and 05 is the port number.

Accessing the Communication Manager SAT CLI

You can access the command line interface (CLI) of the Avaya Communication Manager SAT using any of the following methods:

- Using Telnet over the Customer LAN on page 567
- Using Avaya Terminal Emulator for LAN Connection to Communication Manager on page 573
- Using Windows for PPP Modem Connection (Windows 2000 or XP) on page 569

  This connection requires you to have a modem on your PC. It also requires you to do the following first:

  - Setting up Windows for Modem Connection to the Media Server (Windows 2000 or XP) on page 568
  - Configuring the remote PC for PPP modem connection (Windows 2000 or XP, Terminal Emulator, or ASA) on page 569

- Using Avaya Terminal Emulator for Modem Connection to Communication Manager on page 574

  This connection requires you to have a modem on your PC. It also requires you to do the following first:

  - Setting up Windows for Modem Connection to the Media Server (Windows 2000 or XP) on page 568
  - Configuring the remote PC for PPP modem connection (Windows 2000 or XP, Terminal Emulator, or ASA) on page 569

- Configuring Avaya Site Administration on page 575

Using Secure Shell for remote login

You can log in remotely to the following platforms using Secure Shell (SSH) as a secure protocol:

- G350 Media Gateway
- S8300, S8500, S8700, or S8710 Media Server Linux command line
- Communication Manager System Administration Terminal (SAT) interface on a Media Server using port 5022.

The SSH capability provides a highly secure method for remote access. The capability also allows a system administrator to disable Telnet when it is not needed, making for a more secure system. For details on disabling Telnet, see Turning off Telnet for increased security on page 568.

Note:

The client device for remote login must also be enabled and configured for SSH.

Refer to your client P.C. documentation for instructions on the proper commands for SSH.

Using Telnet over the Customer LAN

Note:

For ease of administration, it is recommended that you use the Avaya Terminal Emulator, instead of Telnet, whenever possible.

To use Telnet over the customer LAN:

1. Make sure you have an active Ethernet (LAN) connection from your computer to the Media Server.
2. Access the telnet program; for example:
   - On a Windows system, go to the **Start** menu and select **Run**.
   - Type **`telnet <server_IP_address> 5023`**. You might also type the server name if your company's DNS server has been administered with the media server name.

3. When the **login** prompt appears, type the appropriate user name (such as **cust** or **craft**).
4. When prompted, enter the appropriate password or ASG challenge.
5. If you log in as **craft**, you are prompted to suppress alarm origination. Generally you should accept the default value (yes).
6. Enter your preferred terminal type.

   The system displays the SAT command line.

Turning off Telnet for increased security

If you are using SSH to log in to the server or gateway, you may wish to turn off Telnet for an extra level of security. To turn off Telnet:

1. On the Maintenance Web page main menu, click on **Firewall**.

   The system displays the **Firewall** page.
2. Click the **Telnet Input to Server** boxes so that the check mark disappears.
3. Click the **Telnet Output from Server** box so that the check mark disappears.
4. Click the **Submit** button.

Setting up Windows for Modem Connection to the Media Server (Windows 2000 or XP)

Note:

The remote dial-up PC must be configured for PPP access. Also, Avaya Terminal Emulator does *not* support Windows XP.

To set up Windows for Modem Connection:

1. Right-click **My Network Places** and click **Properties**.
2. Click **Make New Connection** and follow the Network Connection Wizard:
3. Select **Dial-up to private network** on the **Nework Connection Type** screen.
4. In the **Phone number** field, enter the appropriate telephone number inserting special digits such as 9 and 1 or *70, if necessary.
5. On the **Connection Availability** screen, click **For all users** or **Only for myself**, as appropriate.
6. On the **Completing the Network Connection Wizard** screen, type the name you want to use for this connection. This name will appear in the **Network and Dial-up Connections** list.
7. Check **Add a shortcut to my desktop**, if desired, and click **Finish**.
8. If a **Connect** screen appears, click **Cancel**.

Configuring the remote PC for PPP modem connection (Windows 2000 or XP, Terminal Emulator, or ASA)

To configure the remote PC for PPP modem connection:

1. On your PC's desktop, right-click **My Network Places** and click **Properties**.

   The system diplays the **Network and Dial-up Connections** screen.

2. Double-click the connection name you made in the previous task, Setting up Windows for Modem Connection to the Media Server (Windows 2000 or XP) on page 568.

   Note:

   Depending on your system, the **Connect** screen may appear, from which you must select **Properties**.

3. Click the **Security** tab.
4. Select the **Advanced (custom settings)** radio button.
5. Check the **Show terminal window** checkbox.
6. Click the **Networking** tab.
7. In the **Components** box, verify that Internet Protocol (TCP/IP) and Client for Microsoft Netwrks are both checked.
8. Select **Internet Protocol (TCP/IP)** and click **Properties**.
9. Click the **Advanced** button.
10. Uncheck (clear) the **Use default gateway on remote network** box.
11. Click **OK** three times to exit and save the changes.

Using Windows for PPP Modem Connection (Windows 2000 or XP)

This connection requires you to have a modem on your PC. It also requires you to do the following first:

- Setting up Windows for Modem Connection to the Media Server (Windows 2000 or XP) on page 568
- Configuring the remote PC for PPP modem connection (Windows 2000 or XP, Terminal Emulator, or ASA) on page 569.

To use Windows for PPP modem connection:

1. Return to the **Network and Dial-up Connections** screen and right-click the connection you just created.
2. Select **Connect**.
3. Leave the **User Name**, **Password**, and **Domain** fields blank. If the **Dial** field is blank, enter the appropriate telephone number.

4. Click the **Dial** button. When the media server's modem answers, the system displays the **After Dial Terminal** screen.
5. Log on to the LAN.
   a. Enter your remote access login name and password.
   b. When the **Start PPP Now!** message appears, click **Done**.
   c. The system displays a small double-computer icon in the lower right portion of your screen.
6. Double-click the double-computer icon.
7. The system displays the connection's **Dialup Status** box.
8. Click on the **Details** tab.
9. Note the **Server** IP address.
10. Open a telnet session to the Media Server:

    Type **`telnet <ip-address>`**, where ***`<ip-address>`*** is the Server IP address as noted in the **Dialup Status** box from Step 9.
11. Access SAT or use the CLI commands as needed.

International Modem Settings

You can change modem settings for a specific country using the Linux command line. To enter AT commands from a Linux command prompt:

1. On the command line, type (without quotes): **cu -l /dev/ttys1**, where -l is a lowercase L. Replace "ttyS1" with the tty port for the USB (e.g., /dev/usb/ttyACM0 for USB port 1). Press **Enter**.

   The system displays a "Connected" message.
2. Type **`ATI`**. Press **Enter**.

   The system displays an **OK** message.
3. Type the appropriate AT command to change the modem setting for the desired country. For example, for Japan, type **AT%T19,0,10&W0**. Press **Enter**.

   The system displays an **OK** message.
4. To return to the command prompt, type "**~**" (without quotes).

The following table shows AT commands and result codes for country-specific modem configuration

:

Country	AT Command (Hexadecimal)	Result Code (Decimal)
Argentina	AT%T19,0,34	52
Australia	AT%T19,0,01	1
Austria	AT%T19,0,34	52
Belgium	AT%T19,0,34	52
Brazil	AT%T19,0,34	52
Canada	AT%T19,0,34	52
China	AT%T19,0,34	52
Cyprus	AT%T19,0,34	52
Czech Republic	AT%T19,0,25	37
Denmark	AT%T19,0,34	52
Finland	AT%T19,0,34	52
France	AT%T19,0,34	52
Germany	AT%T19,0,34	52
Greece	AT%T19,0,34	52
Hong Kong	AT%T19,0,30	48
Hungary	AT%T19,0,30	48
Iceland	AT%T19,0,34	52
India	AT%T19,0,30	48
Indonesia	AT%T19,0,30	48
Ireland	AT%T19,0,34	52
Italy	AT%T19,0,34	52
Japan	AT%T19,0,10	16
Korea	AT%T19,0,30	48
Liechtenstein	AT%T19,0,34	52

1 of 2

Country	AT Command (Hexadecimal)	Result Code (Decimal)
Luxembourg	AT%T19,0,34	52
Mexico	AT%T19,0,34	52
Netherlands	AT%T19,0,34	52
New Zealand	AT%T19,0,09	9
Norway	AT%T19,0,34	52
Philippines	AT%T19,0,30	48
Poland	AT%T19,0,30	48
Portugal	AT%T19,0,34	52
Russia	AT%T19,0,34	52
Singapore	AT%T19,0,30	48
South Africa	AT%T19,0,35	53
Slovak Republic	AT%T19,0,34	52
Slovenia	AT%T19,0,30	48
Spain	AT%T19,0,34	52
Sweden	AT%T19,0,34	52
Switzerland	AT%T19,0,34	52
Turkey	AT%T19,0,34	52
United Kingdom	AT%T19,0,34	52
United States	AT%T19,0,34	52
		2 of 2

Enabling transmission over IP networks for modem, TTY, and fax calls

Prerequisites

The ability to transmit fax, modem, and TTY calls over IP trunks or LANs and WANs assumes that the endpoints sending and receiving the calls are connected to a private network that uses H.323 trunking or LAN connections between gateways and/or port networks. This type of transmission also assumes that calls can either be passed over the public network using ISDN-PRI trunks or passed over an H.323 private network to Communication Manager switches that are similarly enabled.

As a result, it is assumed that you have assigned, or will assign, to the network gateways the IP codec you define in this procedure. For our example, the network region 1 will be assigned codec set 1, which you are enabling to handle fax, modem, and TTY calls.

To enable transmission over IP networks for modem, TTY, and fax calls:

1. Type **`change ip-codec-set 1`** and press **Enter**.

   The **IP Codec Set** screen appears.

Complete the fields as required for each media type you want to enable, and press **Enter**.

Using Avaya Terminal Emulator for LAN Connection to Communication Manager

You can download the Avaya Terminal Emulator from the main menu for the VisAbility Management Suite. Simply click **Download** next to the Administration menu item and follow the instructions.

Once the Terminal Emulator is installed on your PC, use the following steps to establish a LAN connection to your Media Server.

1. Double-click the Terminal Emulator icon on your desktop. Alternatively, go to the Start menu, select Programs, then Avaya, then Terminal Emulator.

   The system displays the Terminal Emulator.

2. From the menu bar across the top of the screen, select **Phones**, then select **Connection List**.

   The system displays the **Connections** screen.

3. From the menu bar across the top, select **Connection**, then select **New Connection**.

   The system displays the **Connection Settings** screen.

4. Put in a name for the connection. Usually, this will be the name of your Media Server.
5. On the **Host** screen, click **Telnet**.
6. Click the **Emulation** tab at the top.

   The system displays the **Emulation** tab.

7. From the Emulator dragdown box, select the emulator you desire, usually 513BCT (default), AT&T 4410, AT&T or DECVT100.
8. In the **Keyboard** window, select **pbx**.
9. Click the **Network** tab.

   The system displays the **Network** tab.

10. In the **IP address** field, type the IP address of the Media Server.
11. In the **TCP/IP port number** field, type **5023** to log in directly to the Communication Manager SAT command line.

12. Click **OK**.

    The **Connection Settings** screen disappears.

13. On the **Connections** screen, double-click the name of the connection you just set up.

    The Login prompt for the Communication Manager software appears.

14. Log in to Communication Manager to access the SAT command screen.

Using Avaya Terminal Emulator for Modem Connection to Communication Manager

This connection requires you to have a modem on your PC. It also requires you to do the following first:

- Setting up Windows for Modem Connection to the Media Server (Windows 2000 or XP) on page 568
- Configuring the remote PC for PPP modem connection (Windows 2000 or XP, Terminal Emulator, or ASA) on page 569

Once the Terminal Emulator is installed on your PC and you have a modem attached and configured to both your PC and the Media Server, use the following steps to establish a modem connection to your Media Server.

1. Double-click the Terminal Emulator icon off of your desktop. Alternatively, go to the Start menu, select **Programs**, then select **Avaya**, and finally select **Terminal Emulator**.

   The system displays the Terminal Emulator.

2. From the menu bar across the top of the screen, select **Phones**, then select **Connection List**.

   The system displays the **Connections** screen.

3. From the menu bar across the top, select **Connection**, then select **New Connection**.

   The system displays the **Connection Settings** screen.

4. Put in a name for the connection. Usually, this will be the name of your Media Server.
5. In the Host window, click **Telnet**.
6. Click the **Emulation** tab at the top.

   The system displays the **Emulation** tab.

7. From the Emulator dragdown box, select the emulator you desire, usually 513BCT (default), AT&T 4410, AT&T or DECVT100.
8. In the Keyboard window, select **pbx**.
9. Click the **Modem** tab.

   The system displays the **Modem** tab.

10. In the **IP address** field, type the IP address of the connection **Dialup Status** box as noted in Step 9.
11. In the **TCP/IP port number** field, type **5023** to log in directly to the Communication Manager SAT command line.
12. In the **Modem** field, use the dragdown box to select the type of modem that your PC uses.
13. In the **Serial port** field, select the COM port you are using for your modem connection.
14. In the **Baud rate** field, select 9600 from the dragdown box.
15. Click the **Dial Numbers** tab.

    The system displays the **Display Numbers** tab.
16. Type the telephone number of the Media Server as appropriate. Enter **1** in the **Country Code** field for long-distance.
17. Click **OK**.
18. On the **Connections** screen, double-click. the name of the connection you just set up.
19. The PC dials up the Media Server, and when connected, the login prompt for the Communication Manager software appears.
20. Log in to Communication Manager to access the SAT command prompt screen.

Configuring Avaya Site Administration

When Avaya Site Administration is initially installed on a client machine, it needs to be configured to communicate with Communication Manager on the Media Server.

When it runs initially, after downloading, you need to create a new entry for the switch connection. To create new entries for the switch, follow the procedure Adding a Switch Administration Item on page 575.

Adding a Switch Administration Item

1. Click **File > New > Voice System**.

   The system displays the **Add Voice System** screen .
2. Enter a name in the Voice System Name: field.
3. Click **Next**. The connection type dialog box displays.
4. Click the **Network connection** radio button.
5. Click **Next**. The Network Connection dialog box displays.
6. Enter the IP address used to connect to the Media Server.
7. Click **Next**. The Network Connection/Port Number dialog box displays.
8. TCP/IP Port Number: For the port number, enter port **5023**.

9. Click **Next**. The Network Connection/Timeout Parameters dialog box displays. Leave the default values for the timeout parameters.
10. Click **Next**. The login type dialog box displays.
11. Click the **"I want to login manually each time"** radio button.
12. Click **Next**. The switch summary dialog box displays.
13. Check the information, use the **Next** button to make corrections if necessary, and click the **Next** button to test the connection.
14. When the connection is successfully tested, click **Next** and then **Finish**.

Logging in to the Media Server with ASA

To start Avaya Site Administration, click **Start > Programs > Avaya > Site Administration**. Avaya Site Administration supports a terminal emulation mode, which is directly equivalent to SAT command interface. Avaya Site Administration also supports a whole range of other features, including the GEDI and Data Import. For more information refer to the Online Help, Guided Tour, and Show Me accessed from the Avaya Site Administration Help menu.

To use Avaya Site Administration, open the application and select the switch (media server) you want to access. When prompted, log in.

When you are logged in, click **Start GEDI**.

Screen and command change summary

(For S8300, S8700, or S8710 Media Servers) The following screens have been added or changed to accommodate G700 Media Gateways, S8300 Media Servers, and other media modules. Changes relevant to an S8300 or G700 configuration are summarized in this section.

Media Gateway changes

(For S8300, S8700, or S8710 Media Servers) The commands and screens for media gateways include:

- The **Media-Gateway** administration screen is used to administer G700 Media Gateways and their media modules. Information is similar to the **list media-gateway** screen (next item), but also includes MAC address, network region, location and site data.
- The **`list media-gateway ['print' or 'schedule']`** command shows the list of currently administered gateways. Information includes the media gateway number, name, serial number, IP address, and whether or not this media gateway is currently registered with the call controller. The IP address field is blank until the media gateway registers once, then remains populated.
- The **`list configuration media-gateway x`** command allows you to list all the assigned ports on the media modules for the G700 Media Gateway specified by its number (***x***).

System-Parameters Customer-Options screen

(For S8300, S8700, or S8710 Media Servers) This screen has been updated to indicate the presence of an S8300 Media Server as a primary or standby call processor as listed below. See System-Parameters Customer-Options in the Screen Reference chapter of this document for a complete description of this screen.

- The OPTIONAL FEATURES section contains a **Local Survivable Processor (LSP)** field. If it displays a **y** (yes), this S8300 Media Server is configured to provide standby call processing in case the primary media server is unavailable. See Configuring the Local Survivable Processor on page 558 for details. This display-only field can be set only by the license file.
- Two additional fields in this section indicate if the primary call-processing controller is an S8300 Media Server. If traditional port networking is disabled and Processor Ethernet is enabled, an S8300 Media Server is controlling telecommunications.
  - **Port Network Support**: set to **n** indicates that traditional port networking is disabled. An S8300 Media Server is the primary call controller.
  - **Processor Ethernet**: set to **y** indicates the presence of an S8300 Media Server.

Quality of Service Monitoring

(For S8300, S8700, or S8710 Media Servers) Several screen changes allow you to monitor Quality of Service (QoS) on an S8300 Media Server with a G700 configuration. The media gateway can send data to a real-time control protocol (RTCP) server, which in turn monitors the network region's performance. Screens include:

- An RTCP MONITOR SERVER section on the **IP-Options System Parameters** screen allows you to enter a single default IP address, server port, and RTCP report period that can be utilized by all administered network regions. This means you do not have to re-enter the IP address each time you access the **IP Network Region** screen.
- The **IP Network Region** screen also must be administered for QoS monitoring (for details, see *Administration for Network Connectivity for Avaya Communication Manager*, 555-233-504). If the **RTCP Enabled** field is left at default (**y**), then be sure to set a valid IP address in the **IP-Options System Parameters** screen. For situations that require customization, this screen is administered on a per IP network regional basis. Items to customize include:
  - Enabling or disabling of RTCP monitoring
  - Modifications to the report flow rate
  - Changes to the server IP address and server port
- The **`list ip-network-region qos`** and **`list ip-network-region monitor`** commands list quality of service and monitor server parameters from the **IP Network Region** screen as follows:

- **qos** displays VoIP media and call control (and their 802.1p priority values), BBE DiffServ PHB values, RSVP profile and refresh rate.
- **monitor** displays RTCP monitor server IP address, port number, report flowrate, codec set, and UDP port range parameters.

Media Gateway serviceability commands

(For S8300, S8700, or S8710 Media Servers) Additional commands related to media gateways appear in *Maintenance for the Avaya G700 Media Gateway controlled by an Avaya S8300 Media Server or an Avaya S8700 Media Server*. These include:

- The **`status media-gateways`** command provides an alarm summary, busyout summary, and link summary of all configured media gateways.
- Several commands have been modified to support the media gateway port identification format described in Command syntax changes for media modules on page 566. These include:
  - Message Sequence Trace (mst)
  - display errors
  - display alarms

SNMP Agents

The S8300, S8500, S8700, or S8710 Media Servers contain SNMP modules. A simplified SNMP manager exists to collect traps from network elements such as the Uninterruptible Power Supply (UPS), Ethernet switches, and media gateways. An SNMP alarm sub-agent exists to send SNMP traps to customer provided network managers. An SNMP alarm sub-agent exists to send alarms as traps to the Initialization and Administration System (INADS). An SNMP master agent, MIB-II sub-agent, and Communication Manager SNMP agent (CMSA) are provided.

The co-resident CMSA is used to manage the co-resident call-processing software.

Note:

The following procedures are for the S8300, S8500, S8700, or S8710 Media Servers only.

Administering log-in permissions for SNMP

To set up a Communication Manager SNMP agent (CMSA), you first need the ability to display administration commands and get system measurements.

Instructions

1. On the SAT, type **change permissions acpsnmp** and press **Enter**.

   The **Command Permissions Categories** screen displays.

Figure 202: Command Permission Categories screen

```
change permissions acpsnmp                                      Page 1 of x

                          COMMAND PERMISSION CATEGORIES

                            Login Name: acpsnmp

COMMON COMMANDS

              Display Admin. and Maint. Data? y
                         System Measurements? y
              System Mgmt Data Transfer Only? n

ADMINISTRATION COMMANDS

          Administer Stations? n                  Administer Features? n
            Administer Trunks? n                Administer Permissions? n

     Additional Restrictions? n

MAINTENANCE COMMANDS
            Maintain Stations? n        Maintain Switch Circuit Packs? n
              Maintain Trunks? n       Maintain Process Circuit Packs? n
              Maintain System? n                Maintain Enhanced DS1: n
```

2. In the **Display Admin. and Maint. Data** field, type **y**.
3. In the **System Measurements** field, type **y**.

   Note:

   To be able to do **busy out** and **release** commands, do steps 4-9, if applicable (based on your customer options):

4. In the **Maintain Stations** field, type **y**.
5. In the **Maintain Trunks** field, type **y**.
6. In the **Maintain System** field, type **y**.

7. In the **Maintain Switch Circuit Packs** field, type **y**.
8. In the **Maintain Process Circuit Packs** field, type **y**.
9. In the **Maintain Enhanced DS1** field, type **y**.
10. Press **Enter** to save your changes.

Adding trap destinations

A trap destination is an IP address that needs to receive SNMP alarm traps from the media server.

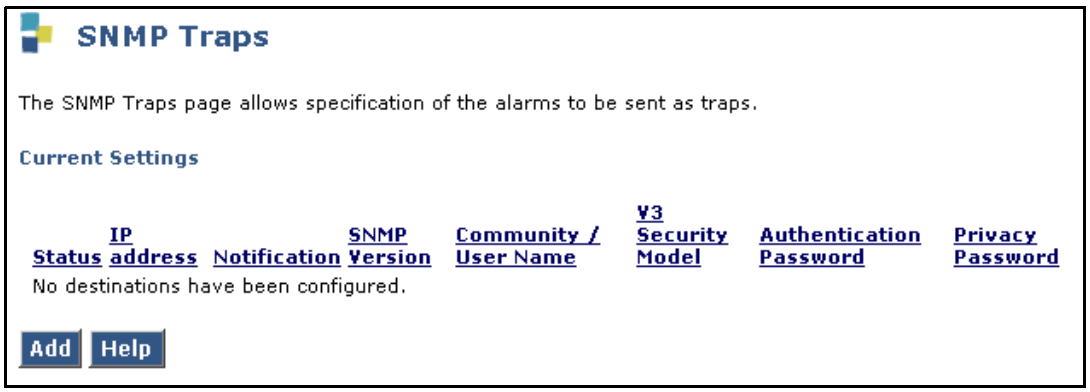
Instructions

To add a trap destination:

1. On the main page, under **Alarms**, click on **SNMP Traps**.

   The SNMP Trap page displays.

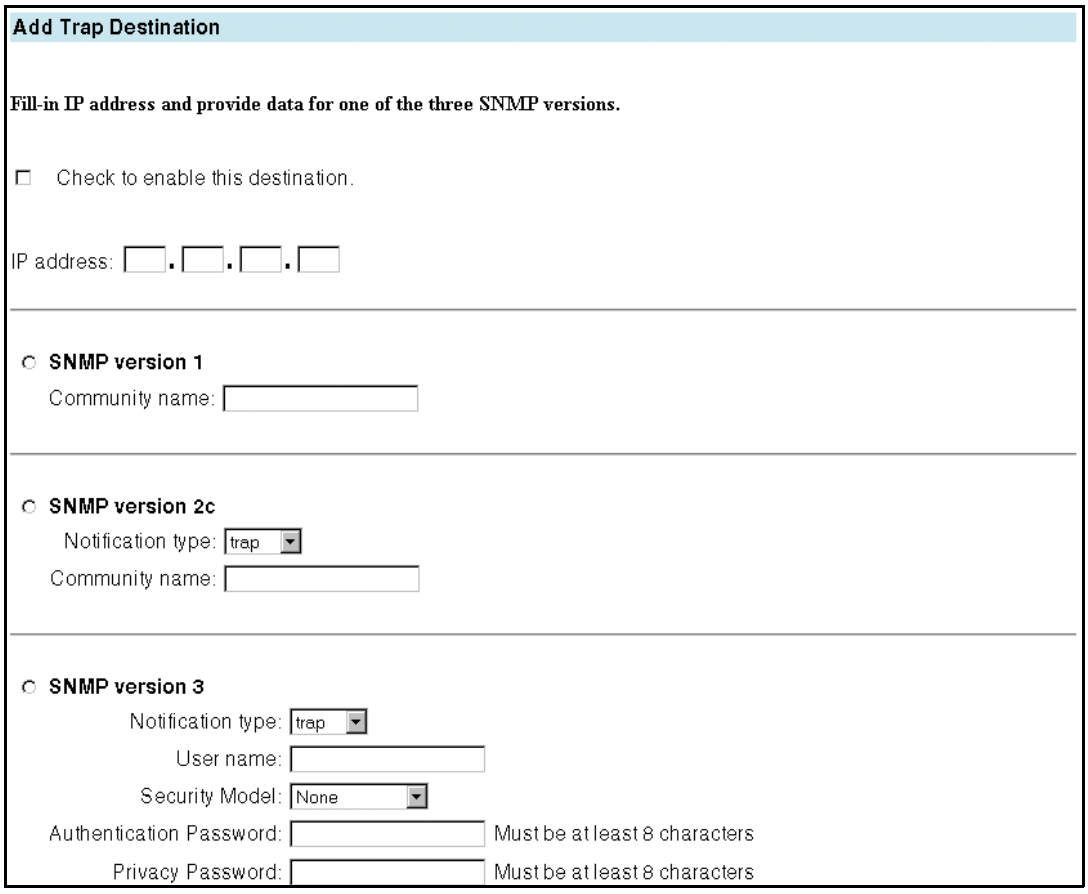
Figure 203: SNMP Trap screen



2. Click **Add**.

   The **Add Trap Destination** screen displays.

Figure 204: Add Trap Destination screen



3. Check the **Check to enable this destination** box.
4. In the **IP address** field, enter the IP address for destination.
5. Select the SNMP version for the destination of the traps. You may select one or more than one version.
6. Complete the fields associated with the version(s) you selected:
   - **SNMP version 1**

     In the `Community name` field, enter the SNMP community name.
   - **SNMP version 2c**

     a. Select the **Notification type**: **trap** or **inform**.

     b. In the `Community name` field, enter the SNMP community name.
   - **SNMP version 3**

     a. Select the **Notification type**: **trap** or **inform**.

     b. In the **User name** field, enter an SNMP user name that the destination will recognize.

c. In the **Security Model** field, type:

none — Traps are sent in plain text without a digital signature.

authentication — An authentication password must be given. SNMP v3 uses this pass phrase to digitally "sign" v3 traps using MD5 protocol (associate them with the user).

privacy — Both an authentication password and a privacy password must be given in order to provide user-specific authentication and encryption. Traps are signed as above and also encrypted using Data Encryption Standard (DES) protocol.

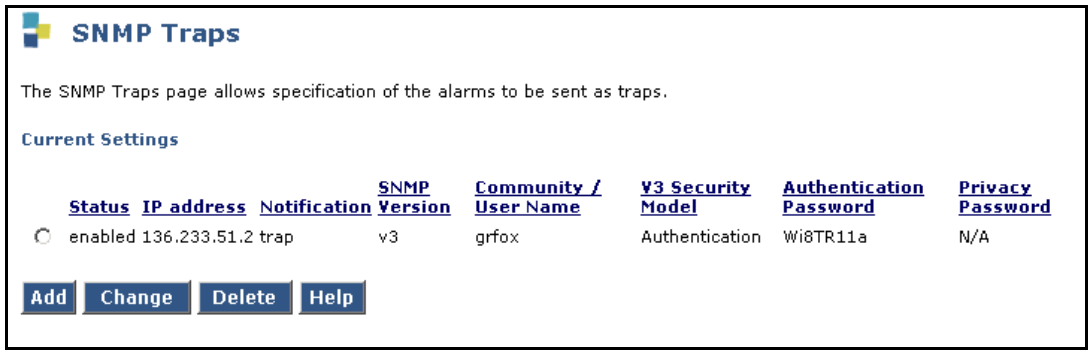
d. If you selected **authentication** as your security model, then in the Authentication Password field, type a text string at least 8 characters long to provide user-specific authentication by means of a digital signature. The pass phrase can contain any characters except: ` \ & , ' " (for example: hon3yb38r).

If you selected **privacy** as your security model, then complete the Authentication Password field as described in the previous paragraph. Also, in the Privacy Password field, type a text string at least 8 characters long to provide user-specific authentication and trap encryption. The pass phrase can contain any characters except: ` \ & , ' " (for example: k33p58f3).

7. Press **Add** to save your changes.
8. If you go back to check what the information looks like, under SNMP, click on **SNMP Traps**.

   The updated **SNMP Traps** screen displays.

Figure 205: Updated SNMP Traps screen



9. Repeat the steps in this procedure to add as many trap destinations as are required.

Administering SNMP agents

This link provides support for administration of the co-resident SNMP master agent and Communication Manager SNMP agent (CMSA).

- Enable and disable access to the CMSA
- Enable and disable each protocol (V1, V2, V3) individually. Any combination may be enabled.
- Administration of the password for the ACPSNMP login
- Enable SNMP busyout/release of Communication Manager objects and setting QOS parameters

You can use this page to allow or restrict SNMP services at the application level. (You can use the Firewall page — Advance Settings to inhibit the reception of SNMP messages at the network level (see Turning on access for the SNMP ports at the network level on page 587). Both must be enabled for SNMP to function.)

This screen is divided into three sections:

1. Communication Manager SNMP Agent
   - Shows if the CMSA is enabled (the checkbox is checked).
   - Shows if the CMSA can be used to busyout and release boards, ports, trunk groups, and trunk members or to set QOS parameters (checkbox is checked).
2. SNMP Users / Communities
   - Shows the status of SNMP version 1 read-only and read-write communities. If a community has been administered, it displays.
   - Shows the status of SNMP version 2c read-only and read-write communities. If a community has been administered, it displays.
   - Shows the status of SNMP version 3 read-only and read-write users. If a user has been administered, its information displays. Passwords do not display; stars display to show a password has been administered. If a privacy password is used, an authentication password also must be given.
3. IP Addresses for SNMP Access
   - A list of IP addresses from which SNMP access is allowed, or
   - All IP addresses.

Each SNMP version can be enabled and disabled independently of the other versions.

Administering SNMP versions 1, 2c, and 3

CAUTION:

On S8700 or S8710 Media Servers, you must administer an SNMP agent exactly the same way in two places — on both the active and duplicated media servers.

If you decide to use SNMP Version 3, there are three levels of security provided.

1. The user name with no password provides the same level of security as SNMP Version 1 and Version 2c.
2. The Authentication Password field provides an enhanced level of security.
3. The Privacy Password provides an additional level of security.

Instructions

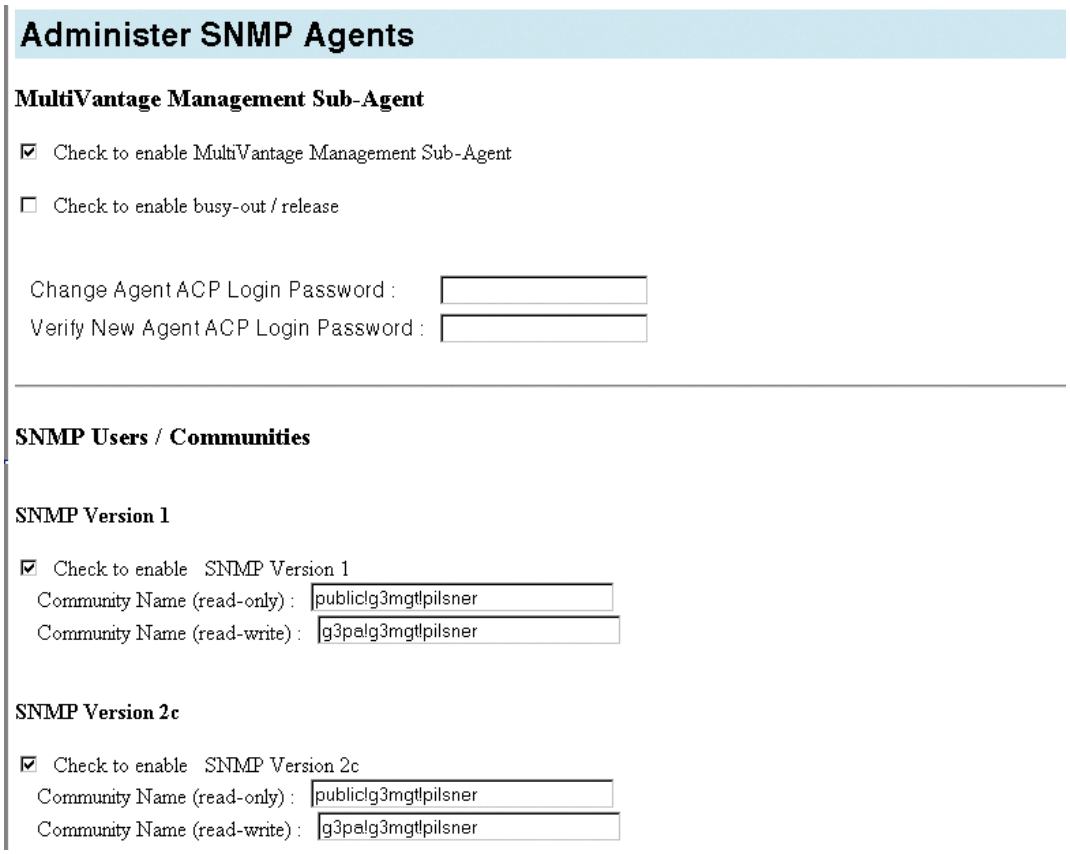
To set up SNMP, you can use Versions 1, 2c, 3 or any combination of the three versions for the IP addresses.

To set up versions 1 and 2c:

1. On the main page, under **Alarms**, click on **SNMP Agents**.

   The system displays the **SNMP Agents** screen.

Figure 206: SNMP Agents



2. Check the **Master Agent status** field for status of the SNMP Master Agent. If the agent is down, the Start/Stop button displays **Start**. In this case, CMSAs cannot operate. To restart the SNMP master agent, click **Start**.

   If the SNMP master agent is running, the Start/Stop button displays **Stop**. To stop the SNMP master agent, click **Stop**.

3. In the Communication Manager SNMP Agent section, check the **Enable Agent** box.

   If this box is checked, an SNMP manager can interact with Communication Manager using this CMSA and MIB-II sub-agent within the restrictions administered on this screen.

   If this box is not checked, SNMP access to Communication Manager is denied. Disabling this CMSA does not inhibit access to the SNMP master agent or MIB-II sub-agent running on this server.

   If you check the **Enable busy-out / release and QOS parameter setting** box, this CMSA can be used to busyout and release boards, ports, trunk groups, and trunk members. The CMSA can also be used to set QOS parameter settings (using SNMP commands).

4. In the **Password** field under **Agent ACP Login**, enter a new password to change the password this CMSA uses to log in to the call-processing software.

5. In the **Re-enter Password** field, repeat the new password.

   Note:

   If these password fields are not administered correctly, all SNMP requests to the CMSA will fail..

6. In the SNMP Version 1 section, check the Enable SNMP Version 1 box if SNMP Version 1 is needed.

   This is the SNMP version of the protocol to be used in communicating with the CMSAs on the server.

   Note:

   The **Community Name** fields are the authentication mechanism for SNMP v1 and v2c. The name can contain up to 50 characters. Any character can be used except ". There must be no spaces and the text is case sensitive.

7. In the **Community Name(read-only)** field, type the name.
8. **In the Community Name(read-write)** field, type the name.

   The user can query agents for information (SNMP GETSs) and send commands to an agent (SNMP SETs).

9. In the SNMP version 2c section, check the **Enable SNMP Version 2c** box if SNMP Version 2 is needed.
10. In the **Community Name** `(read-only)` field, type the name.
11. In the **Community Name** `(read-write)` field, type the name.
12. Scroll down to the SNMP Version 3 section and check the **Enable SNMP Version 3** box if SNMP Version 3 is needed.

    This is the SNMP version of the protocol to be used in communicating with the SNMP agents on the server.

13. In the **User Name** field, type the user name.

    The user can query only for information.

    Your entry must be in any alphanumeric format and up to 50 characters in length.

14. (Optional) In the **Authentication Password** field, type the security password.

    Your entry must be in any alphanumeric format and up to 50 characters in length.

    Note:

    If you did not do Step 14, you cannot do Step 15.

15. (Optional) In the **Privacy Password** field, type the security password.
16. In the IP Addresses for SNMP Access section, check either:
    - **Any IP Address**, and go to Step 19.
    - **Following IP addresses**, and continue with Step 17.

17. If you selected **Following IP addresses**, in the **IP address 1** field, type the IP address.

    The SNMP agents on this server can be programmed to accept incoming SNMP messages from a restricted list of SNMP managers. If an SNMP message is received from an IP address not listed, it is dropped.

18. Repeat Step 17until all IP addresses are listed.
19. Press **Enter** to save your changes.

Displaying the MIB Reference Document

The Management Information Base (MIB) reference document lists the details of Communication Manager data for SNMP management. The MIB can be used to interpret SNMP data that is supported by the Communication Manager SNMP Agent.

To display the MIB reference document:

1. On the main page, under **Alarms**, click on **SNMP Agents.**

   The system displays the **SNMP Agents** page.

2. Click on **Download G3-Avaya-MIB**.

   The system displays a text file of the MIB.

Turning on access for the SNMP ports at the network level

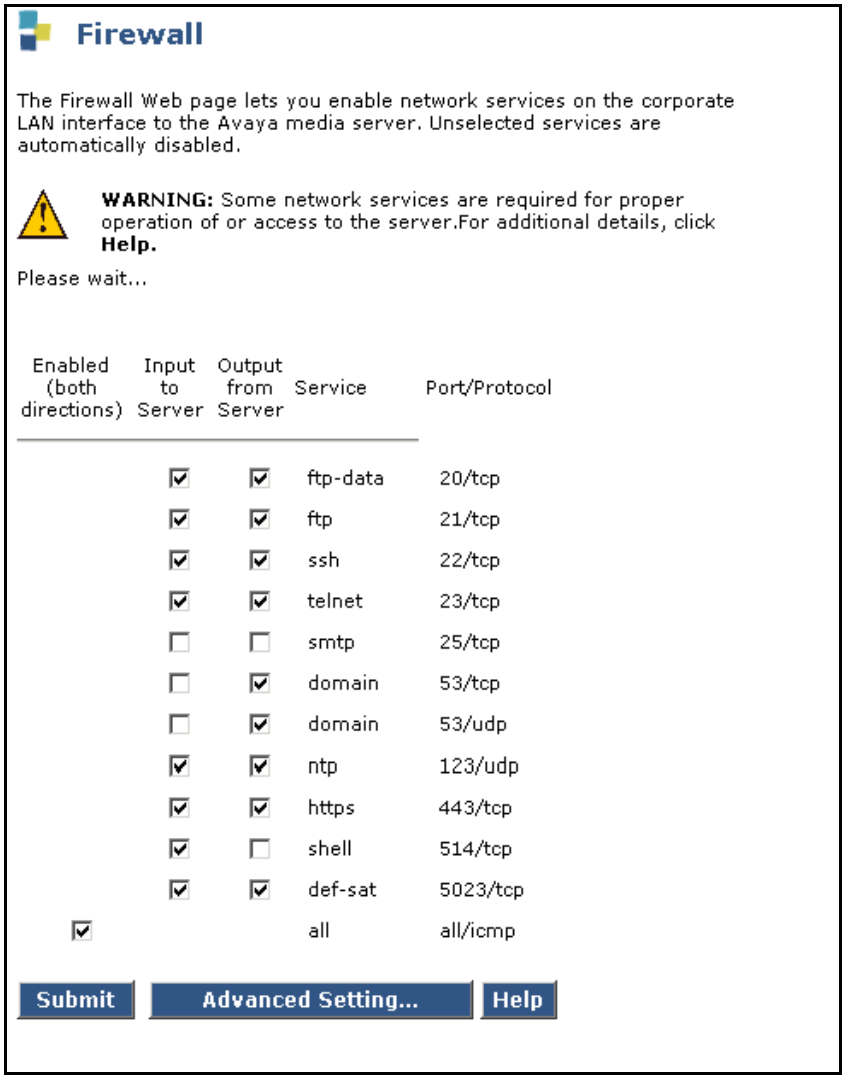
On an S8300, S8500, S8300, S8700, or S8710 Media Server, you must turn on network access for the SNMP ports to allow SNMP access to Communication Manager. To turn on (or turn off) network access for SNMP ports, complete the following steps.

To display the MIB reference document:

1. On the main page, under **Security**, click **Firewall**.

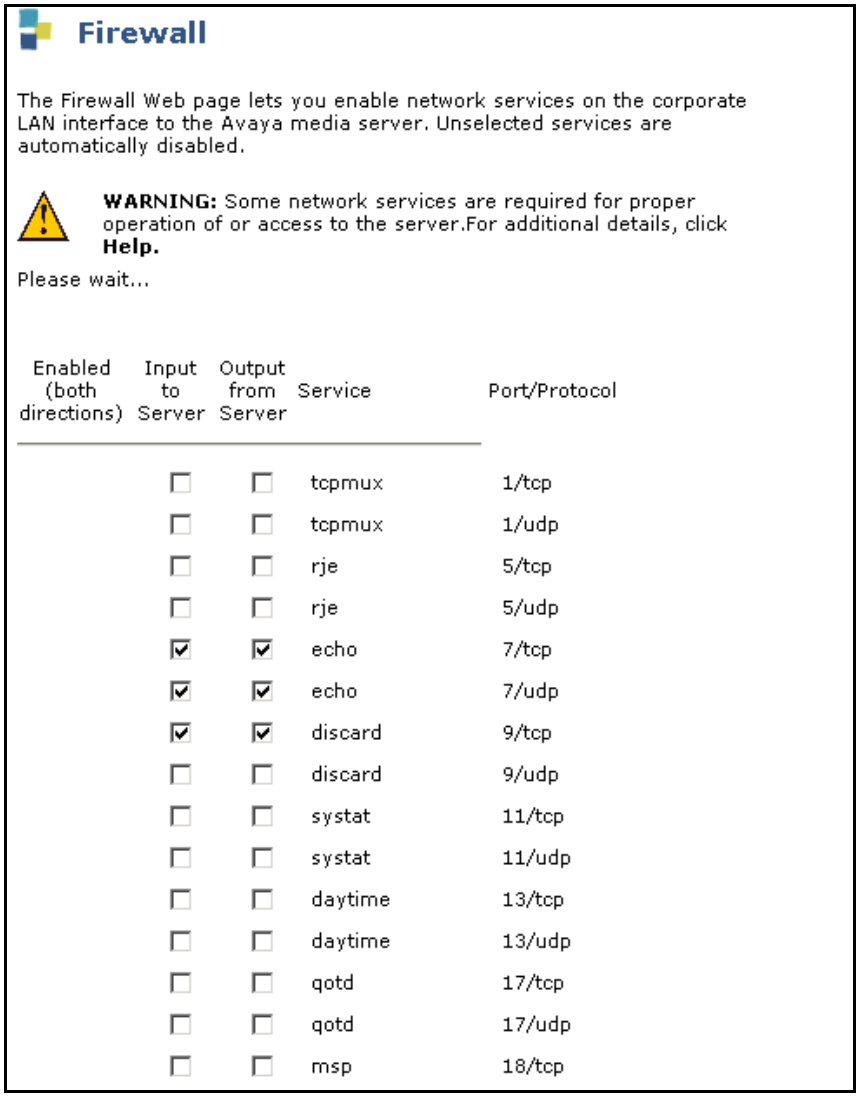
   The system displays the **Firewall** screen.

Figure 207: Firewall screen



2. Click the **Advanced Setting** button.

Figure 208: Firewall Advanced Setting screen



3. Scroll down to the fields for SNMP. These are:
   - snmp 161/tcp
   - snmp 161/udp
   - snmptrap 1
4. Click on the **Input to Server** and **Output to Server** boxes to either turn the ports on or turn them off. When a check appears in the box, the ports are on.

CAUTION:

All input and output boxes for SNMP ports must be turned on (checked) for SNMP access to Communication Manager to work.

Chapter 17: Collecting Billing Information

Collecting Information About Calls

Call Detail Recording (CDR) collects detailed information about all incoming and outgoing calls on specified trunk groups. If you use intraswitch CDR, you can also collect information about calls between designated extensions on Avaya Communication Manager. Communication Manager sends this information to a printer or to some other CDR output device that collects call records and that may also provide reports.

You can have a call accounting system directly connected to your server or media server running Communication Manager. If you are recording call details from several servers, Communication Manager may send the records to a collection device for storage. A system called a poller may then take these records and send them to the call accounting system. The call accounting system sorts them, and produces reports that you can use to compute call costs, allocate charges, analyze calling patterns, detect unauthorized calls, and keep track of unnecessary calls.

Before you start

The call accounting system that you use may be sold by Avaya, or it may come from a different vendor. You need to know how your call accounting system is set up, what type of call accounting system or call detail recording unit you are using, and how it is connected to the server running Communication Manager. You also need to know the format of record that your call accounting system requires.

Instructions

In this example, we are going to establish call detail recording for all calls that come in on trunk group 1 (our CO trunk). We are going to set up CDR so that any call that is handled by an attendant produces a separate record for the attendant part of the call.

1. Type **`change trunk-group 1`** and press **Enter**.

   The **Trunk Group** screen appears.

Figure 209: Trunk Group screen

```
                                TRUNK GROUP

Group Number: 1                     Group Type: co           CDR Reports: y
  Group Name: Outside calls               COR: 85      TN: 1__      TAC: 105
   Direction: two-way        Outgoing Display? n
 Dial Access? n                Busy Threshold: 99     Night Service: 1234
Queue Length: 0           Country: 1           Incoming Destination: attd
   Comm Type: voice                 Auth Code? n  Digit Absorption List: _
    Prefix-1? y                   Trunk Flash? n        Toll Restricted? y

TRUNK PARAMETERS
            Trunk Type: ground start
    Outgoing Dial Type: tone                              Cut-Through? n
     Trunk Termination: rc                    Disconnect Timing(msec): 500_

            Auto Guard? n    Call Still Held? n     Sig Bit Inversion: none

                                 Trunk Gain: high

Disconnect Supervision - In? y  Out? n
Answer Supervision Timeout: 10             Receive Answer Supervision? n
```

2. In the **CDR Reports** field, type **y**.

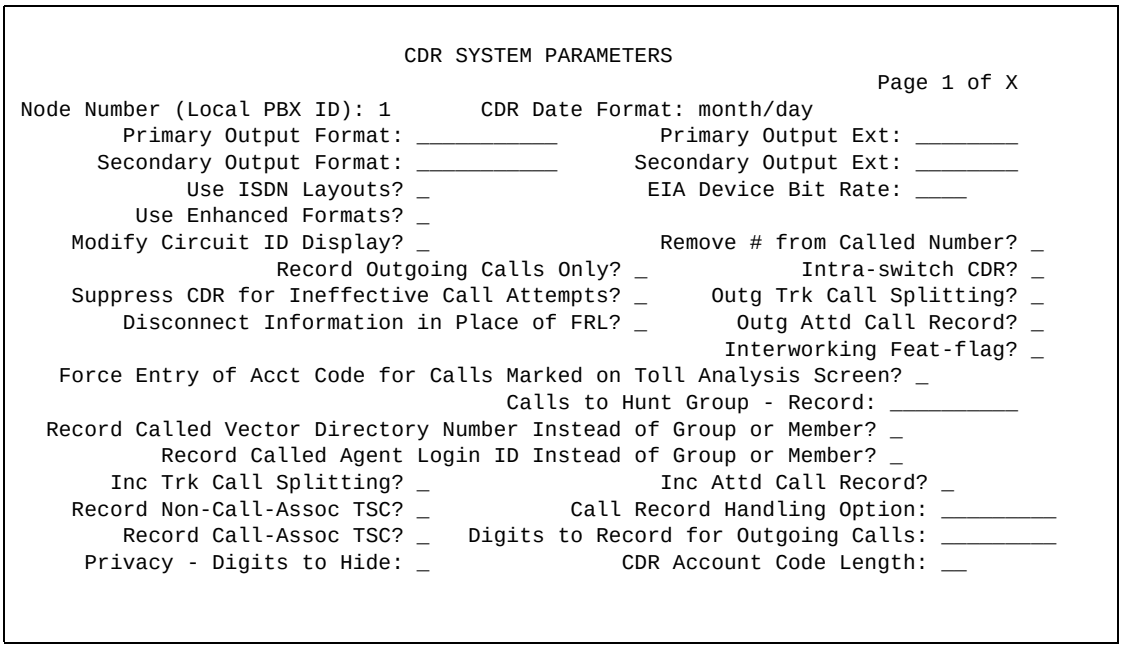
   This tells Communication Manager to create call records for calls made over this trunk group.

3. Press **Enter** to save your changes.

4. Type **change system-parameters cdr** and press **Enter**.

   The **CDR System Parameters** screen appears.

Figure 210: CDR System Parameters



```
                              CDR SYSTEM PARAMETERS
                                                                  Page 1 of X
Node Number (Local PBX ID): 1       CDR Date Format: month/day
        Primary Output Format: ____________        Primary Output Ext: _________
      Secondary Output Format: ____________      Secondary Output Ext: _________
             Use ISDN Layouts? _                  EIA Device Bit Rate: ____
         Use Enhanced Formats? _
    Modify Circuit ID Display? _                   Remove # from Called Number? _
                   Record Outgoing Calls Only? _            Intra-switch CDR? _
    Suppress CDR for Ineffective Call Attempts? _     Outg Trk Call Splitting? _
        Disconnect Information in Place of FRL? _       Outg Attd Call Record? _
                                                     Interworking Feat-flag? _
   Force Entry of Acct Code for Calls Marked on Toll Analysis Screen? _
                                    Calls to Hunt Group - Record: ___________
  Record Called Vector Directory Number Instead of Group or Member? _
           Record Called Agent Login ID Instead of Group or Member? _
       Inc Trk Call Splitting? _                   Inc Attd Call Record? _
    Record Non-Call-Assoc TSC? _           Call Record Handling Option: __________
        Record Call-Assoc TSC? _   Digits to Record for Outgoing Calls: __________
     Privacy - Digits to Hide: _               CDR Account Code Length: __
```

5. In the **CDR Format** field, type **month/day**.

   This determines how the date will appear on the header record.

6. In the **Primary Output Format** field, type **Unformatted**.

   This is the record format that our call accounting system requires. Check with your call accounting vendor to determine the correct record format for your system.

7. In the **Primary Output Ext.** field, type **2055**.

   This is the extension of the data module that we use to connect to our call accounting system.

8. In the **Record Outgoing Calls Only** field, type **n**.

   This tells Communication Manager to create records for both incoming and outgoing calls over all trunk groups that use CDR.

9. In the **Outg Trk Call Splitting** and **Inc Trk Call Splitting** fields, type **y**.

   This tells the system to create a separate record for any portion of an incoming or outgoing call that is transferred or conferenced.

10. In the **Outg Att Call Record** and **Inc Att Call Record** fields, type **y**.

    This tells the system to create a separate record for the attendant portion of any incoming or outgoing call.

More information

You can also administer Communication Manager to produce separate records for calls that are conferenced or transferred. This is called Call Splitting. There are many other variations that you can administer for CDR, and these are described in the screens section of this book.

Related topics

For additional information on CDR, see *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205.

For more information about how to connect a CDR output device, see the *Installation for Adjuncts and Peripherals for Avaya Communication Manager*.

The *Call Detail Acquisition and Processing Reference Manual* also contains CDR information, but has not been updated in several years and is no longer entirely accurate.

Recording Calls Between Users on the Same Server

Call detail recording generally records only those calls either originating or terminating outside the server running Communication Manager. There may be times when you need to record calls between users on the local server. Intra-switch CDR lets you track calls made to and from local extensions.

Instructions

In this example, we administer Communication Manager to record all calls to and from extensions 5100, 5101, and 5102.

1. Type ***change system-parameters cdr*** and press **Enter**.

   The **CDR System Parameters** screen appears.

2. In the **Intraswitch CDR** field, type **y**.
3. Press **Enter** to save your changes.
4. Type **change intra-switch-cdr** and press **Enter**.

   The **Intra-Switch CDR** screen appears.

Figure 211: Intra-Switch CDR screen

```
 Page 1 of 1
                          INTRA-SWITCH CDR
Assigned Members: 2 of 100 administered
 1: 72447 _  19:         37:         55:         73:         91:
 2: 72448    20:         38:         56:         74:         92:
 3:        _ 21:         39:         57:         75:         93:
 4:        _ 22:         40:         58:         76:         94:
 5:          23:         41:         59:         77:         95:
 6:          24:         42:         60:         78:         96:
 7:          25:         43:         61:         79:         97:
 8:          26:         44:         62:         80:         98:
 9:          27:         45:         63:         81:         99:
10:          28:         46:         64:         82:        100.
11:          29:         47:         65:         83:        101:
12:          30:         48:         66:         84:        102:
13:          31:         49:         67:         85:        103:
14:          32:         50:         68:         86:        104:
15:          33:         51:         69:         87:        105:
16:          34:         52:         70:         88:        106:
17:          35:         53:         71:         89:        107:
18:          36:         54:         72:         90:        108:
```

5. In the first three available slots, type **5100**, **5101**, and **5102**.
6. Press **Enter** to save your changes.
7. Communication Manager will now produce call records for all calls to and from these extensions, including those that originated on the local server.

Related topics

See Intra-Switch CDR on page 1118 for more detailed information.

Tracking Calls by Account Code

You can have your users to enter account codes before they make calls. By doing this, you can have a record of how much time was spent on the telephone doing business with or for a particular client.

Instructions

In this example, we are going to set up the system to allow the user at extension 5004 to enter a 5-digit account code before making a call.

1. Type **change system-parameters cdr** and press **Enter**.

   The **CDR System Parameters** screen appears.

Figure 212: CDR System Parameters screen

```
                              CDR SYSTEM PARAMETERS
                                                                  Page 1 of X
   Node Number (Local PBX ID): 1                      CDR Date Format: _________
       Primary Output Format: ___________       Primary Output Ext: ________ Secondary
Output Format: ___________      Secondary Output Ext: ________
             Use ISDN Layouts? _                  EIA Device Bit Rate: ____
         Use Enhanced Formats? _
    Modify Circuit ID Display? _                   Remove # from Called Number? _
                  Record Outgoing Calls Only? _            Intra-switch CDR? _
    Suppress CDR for Ineffective Call Attempts? _     Outg Trk Call Splitting? _
        Disconnect Information in Place of FRL? _       Outg Attd Call Record? _
                                                     Interworking Feat-flag? _
   Force Entry of Acct Code for Calls Marked on Toll Analysis Screen? _
                                    Calls to Hunt Group - Record: __________
  Record Called Vector Directory Number Instead of Group or Member? _
           Record Called Agent Login ID Instead of Group or Member? _
       Inc Trk Call Splitting? _                  Inc Attd Call Record? _
    Record Non-Call-Assoc TSC? _           Call Record Handling Option: _________
        Record Call-Assoc TSC? _   Digits to Record for Outgoing Calls: _________
     Privacy - Digits to Hide: _              CDR Account Code Length: __
```

2. In the **CDR Account Code Length** field, type **5**.
3. Press **Enter** to save your changes.
4. Assign an account button on the **Station** screen for extension 5004. See Adding Feature Buttons on page 121 for more information.
5. Provide your users with a list of account codes to use.
6. You can also assign a Feature Access Code and give this to your users.

Forcing Users to Enter Account Codes

Forced Entry of Account Codes is another form of account code dialing. You can use it to allow certain types of calls only with an account code, to track fax machine usage by account, or just to make sure that you get account information on all relevant calls.

Before you start

Before you can administer Forced Entry of Account Codes, it must be enabled on the **System Parameters Customer-Options** screens. If it is not, please contact your Avaya representative.

Instructions

In this example, we administer the system to force users in our North American office to enter an account code before making international calls.

1. Type **`change system-parameters cdr`** and press **Enter**.

   The **CDR System Parameters** screen appears.

2. In the **Force Entry of Acct Code for Calls Marked on Toll Analysis Form** field, type **y**.
3. In the **CDR Account Code Length** field, type **5**.
4. Press **Enter** to save your changes.
5. Type **`change toll 0`** and press **Enter**.
6. The **Toll Analysis** screen appears.

Figure 213: Toll Analysis screen

```
change toll analysis                                          Page 1 of 1
                                TOLL ANALYSIS                 Percent Full:_
                               Location:
                      Total          Toll  CDR  <--Unrestricted Call List-->
    Dialed String     Min  Max  RCL  List  FEAC  1  2  3  4  5  6  7  8  9 10
0_____________________ 1__  1__  _    x     _    _  _  _  _  _  _  _  _  _  _
01____________________ 9__  17_  _    x     x    _  _  _  _  _  _  _  _  _  _
011___________________ 10_  18_  _    x     x    _  _  _  _  _  _  _  _  _  _
_____________________ ___  ___  _    _     _    _  _  _  _  _  _  _  _  _  _
_____________________ ___  ___  _    _     _    _  _  _  _  _  _  _  _  _  _
_____________________ ___  ___  _    _     _    _  _  _  _  _  _  _  _  _  _
_____________________ ___  ___  _    _     _    _  _  _  _  _  _  _  _  _  _
_____________________ ___  ___  _    _     _    _  _  _  _  _  _  _  _  _  _
_____________________ ___  ___  _    _     _    _  _  _  _  _  _  _  _  _  _
_____________________ ___  ___  _    _     _    _  _  _  _  _  _  _  _  _  _
_____________________ ___  ___  _    _     _    _  _  _  _  _  _  _  _  _  _
_____________________ ___  ___  _    _     _    _  _  _  _  _  _  _  _  _  _
```

7. In the first available **Dialed String** field, type **011**.

   This is the international access code for this office.

8. In the **Total Min** and **Max** columns, type **10** and **18**, respectively.

   This is the minimum and maximum number of digits the system will analyze when determining how to handle the call.

9. In the **Toll List** and **CDR FEAC** columns, type **x**.

10. Press **Enter** to save your changes.

More information

You can also establish a class of restriction with **Forced Entry of Account Codes** set to **y**, and assign this class of restriction (COR) to trunks or other facilities that you want to restrict. With this method, all users with this COR must enter account codes before making any outgoing trunk calls. See Class of Restriction on page 813 for more information.

Receiving Call-Charge Information

Avaya Communication Manager provides two ways to receive information from the public network about the cost of calls. Note that this service is not offered by the public network in some countries, including the US.

- Advice of Charge (AOC, for ISDN trunks) collects charge information from the public network for each outgoing call. Charge advice is a number representing the cost of a call; it may be recorded as either a charging or currency unit.
- Periodic Pulse Metering (PPM, for non-ISDN trunks) accumulates pulses transmitted from the public network at periodic intervals during an outgoing trunk call. At the end of the call, the number of pulses collected is the basis for determining charges.

Before you start

You need to request either AOC or PPM service from your network provider. In some areas, your choice may be limited. Your Avaya technical support representative can help you determine the type of service you need.

Note:

This service is not offered by the public network in some countries, including the U.S.

Collecting call charge information over ISDN

In this example, we administer the system to provide Advice of Charge over an existing ISDN trunk group, at the end of a call. This information will appear on CDR reports.

1. Type **`change trunk-group 2`**.

   The **Trunk Group** screen appears with existing administration for this ISDN trunk group.

Figure 214: Trunk Group screen

```
                                                              Page 1 of x
                              TRUNK GROUP

Group Number: 2                     Group Type: isdn          CDR Reports: y
  Group Name: OUTSIDE_CALL_____________ COR: 1_       TN: 1__      TAC: ____
   Direction: two-way_       Outgoing Display? n
 Dial Access? n                Busy Threshold: 99_         Night Service: ________
Queue Length: 0__
Service Type: public-ntwrk          Auth Code? n             TestCall ITC: rest
                         Far End Test Line No: _______________
TestCall BCC: 4
TRUNK PARAMETERS
         Codeset to Send Display: 6     Codeset to Send TCM,Lookahead: 6
        Max Message Size to Send: 260   Charge Advice: none_____________
  Supplementary Service Protocol: a     Digit Handling(in/out):enbloc/enbloc

            Trunk Hunt: cyclical             QSIG Value-Added Avaya? n
     Connected to Toll? n      STT Loss: normal       DTT to DCO Loss: normal
Calling Number - Delete: ___ Insert: _______________ Numbering Format: ________
              Bit Rate: 1200_        Synchronization: async    Duplex: full
 Disconnect Supervision - In? y  Out? n
 Answer Supervision Timeout: 0__
```

2. In the **CDR Reports** field, type **y**.

   This ensures that the AOC information appears on the CDR report.

3. Verify that **Service Type** is **public-ntwrk**.
4. In the **Supplementary Service Protocol** field, type **a**.
5. The **Charge Advice** field, type **end-on-request**.

   This ensures that Communication Manager will place one request for charge information. This reduces the amount of information passed to Communication Manager and consumes less processor time than other options.

6. Press **Enter** to save your changes.

Receiving call-charge information over non-ISDN trunks

In this example, we will administer an existing Direct Inward and Outward Dialing (DIOD) trunk to receive PPM from the public network.

1. Type **`change trunk-group 3`**.

   Page 1 of the **Trunk Group** screen appears with existing administration for this trunk group. Click the numbered page tabs or **Next Page** to find fields that appear on subsequent pages of the **Trunk Group** screen.

Figure 215: Trunk Group screen

```
                                                              Page 1 of x
                                TRUNK GROUP

 Group Number: 3                     Group Type: diod          CDR Reports: y
   Group Name: DIOD_PPM_____________________ COR: 1_       TN: 1__      TAC: 112_
    Direction: two-way_       Outgoing Display? n
 Dial Access? n                 Busy Threshold: 99_         Night Service: ________
 Queue Length: 0__
    Comm Type: voice                  Auth Code? n   Digit Absorption List:
     Prefix-1? y                    Trunk Flash? n         Toll Restricted? y
TRUNK PARAMETERS
            Trunk Type: auto/immed
    Outgoing Dial Type: tone                          Incoming Dial Type: tone
     Trunk Termination: rc
       Digit Treatment:                                        Digits:
       Expected Digits:                                 Sig Bit Inversion: none

                                 Trunk Gain: high       Drop Treatment: silence
  Disconnect Supervision - In? y  Out? n
  Answer Supervision Timeout: 0__           Recieve Answer Supervision: y
```

2. In the **CDR Reports** field, type **y**.

   This ensures that the PPM information appears on the CDR report.

3. In the **Direction** field, type **two-way**.
4. In the **PPM** field, type **y**.
5. In the **Frequency** field, type **50/12**.

   This is the signal frequency (in kHz). The frequency you will use depends on what the circuit pack you use is able to accept. See Tone Generation on page 1529 for more information.

6. In the **Administrable Timers** section, set the **Outgoing Glare Guard** timer to **5** seconds.
7. Press **Enter** to save your changes.
8. You also need to ensure that the values of the **Digital Metering Pulse Minimum**, **Maximum** and **Value** on the **DS1 Circuit Pack** screen are appropriate to the values offered by your service provider. See DS1 Circuit Pack on page 928 for more information.

Related topics

For more information about AOC and PPM, see "Call Charge Information" in *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205.

Viewing Call Charge Information

Avaya Communication Manager provides two ways for you to view call-charge information: on a telephone display or as part of the Call Detail Recording (CDR) report. From a display, users can see the cost of an outgoing call, both while the call is in progress and at the end of the call.

Instructions

In this example, we administer extension 5040 to be able to view the charge of a call in progress. The charges will appear in currency units (in this case, Lira) on the user's telephone display.

1. Type **change trunk-group 2** and press **Enter**.

   The **Trunk Group** screen appears.

Figure 216: Trunk Group screen

```
add trunk-group next                                          Page 2 of x
TRUNK FEATURES
         ACA Assignment? _     Measured: ____
                                              Maintenance Tests? _
                       Data Restriction? _

  Abandoned Call Search? _
  Suppress # Outpulsing? _

      Charge Conversion: _____
          Decimal Point: ______
        Currency Symbol: ___
            Charge Type: _______   Receive Analog Incoming Call ID: ________
                                        Per Call CPN Blocking Code: ___
                                     Per Call CPN Unblocking Code: ___
                                                  MF Tariff Free? _
             Outgoing ANI: _______________
```

2. In the **Charge Conversion** field, type **200**.

   This indicates that one charge unit sent from the service provider is equal to 200 units, in this case, Lira.

3. In the **Decimal Point** field, type **none**.
4. In the **Charge Type** field, type **Lira**.
5. Press **Enter** to save your changes.

6. Type **change system-parameters features** and press **Enter**.

   The **Feature-Related System Parameters** screen appears.

7. In the **Charge Display Update Frequency (seconds)** field, type **30**.

   Frequent display updates may have considerable performance impact.

8. Press **Enter** to save your changes.

9. Now assign extension 5040 a disp-chrg button to give this user the ability to control the charge display. See Adding Feature Buttons on page 121 for more information.

More information

If you want end users to control when they view this information, you can assign a display button that they can press to see the current call charges. If you want call charges to display automatically whenever a user places an outgoing call, you can set **Automatic Charge Display** to **y** on the user's **COR** screen.

Chapter 18: Command Reference

This chapter contains descriptions of Communication Manager commands that are used in performing administrative tasks. These are most often commands that are invoked by actions such as **add**, **change**, and **remove**. For maintenance-related commands that are invoked using commands such as **list**, **display**, and **status**, see *Maintenance Commands for Avaya Communication Manager, Media Gateways and Servers*, 03-300191. Also see *Reports for Avaya Communication Manager,* 555-233-505.

Avaya Communication Manager software utilizes the following command syntax:

action object qualifier

Use the commands in the tables below to access each administration screen.

Enter text in the **command font** exactly as shown, or an abbreviated form of the word may be entered. MAX is the maximum number available in your system configuration. Enter the proper variable for your system when ***variable font*** appears.

Brackets [ ] indicate the qualifier is optional. Some qualifiers are not available on all systems. For example, on an Avaya S8300 Media Server with Avaya G700 Media Gateway, the **print** qualifier is supported on Linux and Windows platforms only when the command has been entered on a 4410-emulation terminal type. Entering **print** using emulation of any other terminal type will result an error message: "**pr not supported by platform**."

For additional command information, see *Maintenance Commands Reference*, 555-245-101, or *Reports for Avaya Communication Manager,* 555-233-505.

Command Reference

AAR and ARS Digit Analysis Table

Action	Object	Qualifier
change	**aar analysis**	***digits 0 to 9*** **| * | x | X** [**min** ***n***]
change	**ars analysis**	***digits 0 to 9*** **| * | x | X** [**location** (***n*** **| all**)] [**min n**]

Action	Object	Qualifier
display	aar analysis	*digits 0 to 9* | * | x | X [min *n*] [print | schedule]
display	ars analysis	*digits 0 to 9* | * | x | X [location (*n* | all)] [min n] [print | schedule]
list	aar analysis	[start *string*] [count *n*] [route (*n* | r1-r32)] [node *n*] [to-node *n*] [print | schedule]
list	ars analysis	[start *string*] [route (*n* | r1-r32 | p1-p2000)] [location (*n* | all)] [node *n*] [to-node *n*] [count *n*] [schedule]
list	aar route-chosen	*dialed number* [location (*n* | all)] [partition *n*] [print | schedule]
list	ars route-chosen	*dialed number* [location (*n* | all)] [partition *n*] [schedule]

2 of 2

AAR and ARS Digit Conversion Table

Action	Object	Qualifier
change	aar digit-conversion	*digits 0 to 9* | * | x | X
change	ars digit-conversion	*digits 0 to 9* | * | x | X [location (*n* | all)]
display	aar digit-conversion	*digits 0 to 9* | * | x | X [print | schedule]
display	ars digit-conversion	*digits 0 to 9* | * | x | X [location (*n* | all)] [print | schedule]
list	aar digit-conversion	[start *matching pattern*] [count *n*] [print | schedule]
list	ars digit-conversion	[start *matching pattern*] [count *n*] [‘location (*n* | all)] [print | schedule]

Abbreviated Dialing Lists

Action	Object	Qualifier
add	**abbreviated-dialing enhanced**	***dial code first digit(s)*** (3 digits 0-9 | 4 digits 00-99) [**list 1 | list 2**]
change	**abbreviated-dialing enhanced**	***dial code first digit(s)*** (3 digits 0-9 | 4 digits 00-99) [**list 1 | list 2**]
display	**abbreviated-dialing enhanced**	***dial code first digit(s)*** (3 digits 0-9 | 4 digits 00-99) [**print** | **schedule**]
remove	**abbreviated-dialing enhanced**	***dial code first digit(s)*** (3 digits 0-9 | 4 digits 00-99) [**list 1 | list 2**]

Action	Object	Qualifier
add	**abbreviated-dialing group**	***n*** | **next**
change	**abbreviated-dialing group**	***n***
display	**abbreviated-dialing group**	***n*** [**print** | **schedule**]
list	**abbreviated-dialing group**	[**number** ***n***] [**to-number** ***n***] [**count** ***n***] [**print** | **schedule**]
remove	**abbreviated-dialing group**	***n***

Action	Object	Qualifier
change	**abbreviated-dialing personal**	***extension*** with personal list and **list** (1-3)
display	**abbreviated-dialing personal**	***extension*** with personal list and **list** (1-3) [**print** | **schedule**]
list	**abbreviated-dialing personal**	[**ext** ***n***][**to-ext** ***n***][**count** ***n***] [**print** | **schedule**]

Action	Object	Qualifier
add	**abbreviated-dialing system**	
change	**abbreviated-dialing system**	
display	**abbreviated-dialing system**	[**print** | **schedule**]
remove	**abbreviated-dialing system**	

Action	Object	Qualifier
add	**abbreviated-dialing 7103A-buttons**	
change	**abbreviated-dialing 7103A-buttons**	
display	**abbreviated-dialing 7103A-buttons**	[**print** | **schedule**]
remove	**abbreviated-dialing 7103A-buttons**	

Access Endpoint

Action	Object	Qualifier
add	**access-endpoint**	***extension*** | **next**
change	**access-endpoint**	***extension***
display	**access-endpoint**	***extension*** [**print** | **schedule**]
duplicate	**access-endpoint**	***extension***
remove	**access-endpoint**	***extension***
list	**access-endpoint**	***extension*** (starting) [**count** ***n***] [**print** | **schedule**]

Administered Connection

Action	Object	Qualifier
add	administered-connection	*n* | next
change	administered-connection	*n*
display	administered-connection	*n* | all [print | schedule]
duplicate	administered-connection	*administered-connection n* [count *n*]
remove	administered-connection	*n*
list	administered-connection	*starting number* [count *n*] [print | schedule]

Administration Change Notification

Action	Object	Qualifier
notify	history	

Alias Station

Action	Object	Qualifier
change	alias station	
display	alias station	[print | schedule]

Alphanumeric Dialing Table

Action	Object	Qualifier
change	alphanumeric-dial-table	[entry *name*] [next]
display	alphanumeric-dial-table	[entry *name*] [print | schedule]

Announcements/Audio Sources

Action	Object	Qualifier
change	announcements	*n*
change	integ-annc-brd-loc	
display	announcements	*n* [print | schedule]
display	integrated-annc-boards	[print | schedule]
disable	announcement-board	*n*
enable	announcement-board	*n*
erase	announcements	*board-location* | *gateway location*
list	integrated-annc-boards	*board-location* | *gateway location* [schedule]
copy	announcements	[SPE-A | SPE-B] [target device [disk | removable-media]]
restore	announcements	to cabinet *location* [disk | removable-media] [to cabinet *location*]
save	announcements	[SPE-A | SPE-B | active | standby | either | both] [disk | removable-media] [from *cabinet location*)] [spe-active | spe-standby] [from *cabinet location*]

ARS Toll Table

Action	Object	Qualifier
change	**ars toll**	***n*:*xxx*** (1 to MAX:office code) ***n:xyy:nn***; **:** ***x***(2-MAX); ***y*** (0-MAX); ***y*** (0-MAX)
display	**ars toll**	***n:xxx*** (1 to MAX:office code) ***n:xyy: nn***; **:** ***x*** (2-MAX); ***y*** (0-MAX); ***y*** (0-MAX) [**print** | **schedule**]

ATM Board

Action	Object	Qualifier
status	**atm board**	***board location*** [**print**]

Attendant Console

Action	Object	Qualifier
add	**attendant**	***n***
change	**attendant**	***n***
display	**attendant**	***n*** [**print** | **schedule**]
remove	**attendant**	***n***

Authorization Code COR Mapping

Action	Object	Qualifier
change	authorization-code	auth *n*
display	authorization-code	auth *n* [print | schedule]
list	authorization-code	[start *code*] [count *n*] [cor *x*] [to-cor *x*] [print | schedule]

Best Service Routing

Action	Object	Qualifier
add	best-service-routing	*n* | next
change	best-service-routing	*n*
display	best-service-routing	*n* [print | schedule]
list	best-service-routing	*starting number* [count *n*] [print | schedule]
remove	best-service-routing	*n*

Boot Image

Action	Object	Qualifier
get	boot-image	*board location* [print]
set	boot-image	*board location* [image 1 | image 2]

Bulletin Board

Action	Object	Qualifier
change	**bulletin-board**	
display	**bulletin-board**	[**print** | **schedule**]

Button Type

Action	Object	Qualifier
list usage	**button-type**	[**print** | **schedule**]
list usage	**button-type crss-alert**	[**schedule**]
list usage	**button-type hunt-ns**	[**hunt-grp** *n*] [**schedule**]
list usage	**button-type night-serv**	[**schedule**]
list usage	**button-type trunk-ns**	[**trunk-grp** *n*] [**schedule**]

Call Vector

Action	Object	Qualifier
change	**vector**	*n*
display	**vector**	*n* [**print** | **schedule**]

CAMA Numbering Format

Action	Object	Qualifier
change	**cama-numbering**	
display	**cama-numbering**	[**print** | **schedule**]

CDR System Parameters

Action	Object	Qualifier
change	**system-parameters cdr**	
display	**system-parameters cdr**	[**print** | **schedule**]

C-LAN

Action	Object	Qualifier
status	**clan-ip**	***c-lan board location***
status	**clan-port**	***c-lan board location***
status	**clan-usage**	

Class of Restriction

Action	Object	Qualifier
change	**cor**	0-MAX
display	**cor**	0-MAX [**print** | **schedule**]
list	**cor**	[**cor** ***n***] [**to-cor** ***n***] [**print** | **schedule**]

Class of Service

Action	Object	Qualifier
change	**cos**	
display	**cos**	[**print** | **schedule**]

Code Calling IDs

Action	Object	Qualifier
change	**paging code-calling-ids**	
display	**paging code-calling-ids**	[**print** | **schedule**]

Command Permission Categories

Action	Object	Qualifier
change	**permissions**	***login-id***
display	**permissions**	***login-id*** [**print** | **schedule**]

Console-Parameters

Action	Object	Qualifier
change	**console-parameters**	
display	**console-parameters**	[**print** | **schedule**]

Coverage Answer Group

Action	Object	Qualifier
add	**coverage answer-group**	***n*** | **next**
change	**coverage answer-group**	***n***
display	**coverage answer-group**	***n*** [**print** | **schedule**]
list	**coverage answer-group**	[**number** ***n***] [**to-number** ***n***] [**name** ***x***] [**count** ***n***] [**print** | **schedule**]
remove	**coverage answer-group**	***n***

Coverage Path

Action	Object	Qualifier
add	**coverage path**	*n* | **next**
change	**coverage path**	*n*
display	**coverage path**	*n* [**print** | **schedule**]
list	**coverage path**	[**path** *n*] [**to-path** *n*] [**count** *n*] [**print** | **schedule**]
remove	**coverage path**	*n*

Coverage Time of Day

Action	Object	Qualifier
add	**coverage time-of-day**	*n* | **next**
change	**coverage time-of-day**	*n*
display	**coverage time-of-day**	*n* [**print** | **schedule**]
remove	**coverage time-of-day**	*n*

Crm-Features

Action	Object	Qualifier
list	**crm-features**	

CTI-Link

Action	Object	Qualifier
add	**cti-link**	***link n*** | **next**
busyout	**cti-link**	***link n***
change	**cti-link**	***link n***
display	**cti-link**	***link n*** [**print** | **schedule**]
list	**cti-link**	[***cti-link number n***] [**count** ***n***] [**print** | **schedule**]
list	**usage cti-link**	[***cti-link number n***] [**print** | **schedule**]
remove	**cti-link**	***link n***
test	**cti-link**	***link n*** [**long** | **short**] [**repeat** (1-100) | **clear**]

Data Modules

Action	Object	Qualifier
add	**data-module**	***extension*** | **next**
change	**data-module**	***extension***
display	**data-module**	***extension*** [**print** | **schedule**]
list	**data-module**	[**ext** ***n***] [**to-ext** ***n***][**name** ***x***][**count** ***n***] [**print** | **schedule**]

Date and Time

Action	Object	Qualifier
set	**time**	
display	**time**	[**print** | **schedule**]

Daylight Savings Rules

Action	Object	Qualifier
change	daylight-savings-rules	
display	daylight-savings-rules	[print | schedule]

DCS to QSIG TSC Gateway

Action	Object	Qualifier
change	isdn dcs-qsig-tsc-gateway	
display	isdn dcs-qsig-tsc-gateway	[print | schedule]

Dial Plan Analysis Table

Action	Object	Qualifier
change	dialplan analysis	
display	dialplan analysis	[print | schedule]

Dial Plan Parameters

Action	Object	Qualifier
change	dialplan parameters	
display	dialplan parameters	[print | schedule]

Digit Absorption

Action	Object	Qualifier
change	**digit-absorption**	***0-4***
display	**digit-absorption**	***0-4*** [**print** | **schedule**]

Digit String

Action	Object	Qualifier
list usage	**digit-string**	***string*** (0-9, +?*#~prwWms) [**print** | **schedule**]

Display-Messages Button Labels

Action	Object	Qualifier
change	**display-message button-labels**	
display	**display-message button-labels**	[**print** | **schedule**]

DLG CTI Link

Action	Object	Qualifier
status	**dlg cti-link**	[**print**]

DLG Interface

Action	Object	Qualifier
status	**dlg interface**	[**print**]

DS1 Circuit Pack

Action	Object	Qualifier
add	**ds1**	***board location***
change	**ds1**	***board location*** [**print** | **schedule**]
display	**ds1**	***board location*** [**print** | **schedule**]
list	**measurements ds1 summary**	***board location*** [**local | carrier-local | remote**] [**print** | **schedule**]
list	**measurements ds1 log**	***board location*** [**local | carrier-local | remote**] [**print** | **schedule**]
remove	**ds1**	***board location***

Ethernet Option

Action	Object	Qualifier
get	**ethernet-option**	[cabinet ***n*** carrier (A-E) slot ***n***] [**print**]
set	**boot-image**	[cabinet ***n*** carrier (A-E) slot ***n***] **image 1** | **image 2**

Extended Pickup Group

Action	Object	Qualifier
change	**extended-pickup-group**	***n***
display	**extended-pickup-group**	***n*** [**print** | **schedule**]
list	**extended-pickup-group**	[**number** ***n***] [**to-number** ***n***] [**count** ***n***] [**print** | **schedule**]

Extension

Action	Object	Qualifier
list usage	**extension**	***extension*** [**vector**] [**print** | **schedule**]
change	**extension-station**	***extension***

Extensions Administered to have an MCT-Control Button

Action	Object	Qualifier
change	**mct-group-extensions**	

ExtensionType

Action	Object	Qualifier
list	**extension-type**	[***extension*** (starting ***n***)] [[**type** ***x***] [**cor** ***x***] [**to-cor** ***x***] [**cos** ***x***] [**to-cos** x] [**count** ***n***] [**print** | **schedule**]

Feature Access Code

Action	Object	Qualifier
change	**feature-access-codes**	
display	**feature-access-codes**	[**print** | **schedule**]

Feature-Related System Parameters

Action	Object	Qualifier
change	**system-parameters features**	
display	**system-parameters features**	[**print** | **schedule**]

Firmware Station Download

Action	Object	Qualifier
change	**firmware station-download**	
display	**firmware station-download**	
status	**firmware station-download**	[last]

Group Paging

Action	Object	Qualifier
add	**group-page**	***n*** | **next**
change	**group-page**	***n***
display	**group-page**	***n*** [**print** | **schedule**]

History

Action	Object	Qualifier
list	**history**	[**date** ***MM*** **|** **date** ***MM/DD***] [**time** ***HH*** | **time** ***HH:MM***] [**login** ***x***] [**action** x] [**object** ***x***] [**qualifier** ***x***] [**print** | **schedule**]

Holiday Table

Action	Object	Qualifier
change	**holiday-table**	***n***
display	**holiday-table**	***n*** [**print** | **schedule**]
list usage	**holiday-table**	***n*** [**print** | **schedule**]

Hospitality

Action	Object	Qualifier
change	**system-parameters hospitality**	
display	**system-parameters hospitality**	[**print** | **schedule**]

Hunt Group

Action	Object	Qualifier
add	**hunt-group**	***n*** | **next**
change	**hunt-group**	***n***
display	**hunt-group**	***n*** [**number** ***n***][**to-number** ***n***] [**count** ***n***] [**print** | **schedule**]
list	**hunt-group**	[**number** ***n***] [**to-number** ***n***] [**name** ***x***] [**type** ***x***] [**ext** ***n***] [**to-ext** ***n***] [**count** ***n***] [**print** | **schedule**]
remove	**hunt-group**	***n***
list usage	**hunt-group**	***group n*** [**vector**][**print** | **schedule**]
list	**members hunt-group**	***hunt group n*** [**ext** ***n***] [**to-ext** ***n***] [**name** ***x***] [**loginid** ***n***] [**to-loginid** ***n***] [**logname** ***x***] [**count** ***n***] [**print** | **schedule**]

Integrated Announcements

Action	Object	Qualifier
change	**integ-annc-brd-loc**	
display	**integrated-annc-boards**	[**print** | **schedule**]

Intercom Group

Action	Object	Qualifier
add	**intercom-group**	***n*** | **next**
change	**intercom-group**	***n***
display	**intercom-group**	***n*** [**print** | **schedule**]
list	**intercom-group**	[**number** ***n***] [**to-number** ***n***] [**count** ***n***] [**print** | **schedule**]
remove	**intercom-group**	***n***

Inter-Exchange Carrier (IXC) codes

Action	Object	Qualifier
change	**ixc-codes**	
display	**ixc-codes**	[**print** | **schedule**]

Intra-Switch CDR

Action	Object	Qualifier
add	**intra-switch-cdr**	
change	**intra-switch-cdr**	***extension***
display	**intra-switch-cdr**	***extension*** [**print** | **schedule**]
list	**intra-switch-cdr**	***extension*** [**count** ***n***] [**print** | **schedule**]

IP Address

Action	Object	Qualifier
list usage	**ip-address**	***IP address*** [**print** | **schedule**]

IP Codec Set

Action	Object	Qualifier
change	**ip-codec-set**	***n***
display	**ip-codec-set**	***n*** [**print** | **schedule**]

IP Interface

Action	Object	Qualifier
list	**ip-interface**	[**print** | **schedule**]
add	**ip-interface**	[cabinet ***n***]; carrier (***A-E***); slot ***n*** procr]
change	**ip-interface**	[cabinet ***n***]; carrier (***A-E***); slot ***n*** procr]
display	**ip-interface**	[cabinet ***n***]; carrier (***A-E***); slot ***n*** procr [**schedule**]]
remove	**ip-interface**	[cabinet ***n***]; carrier (***A-E***); slot ***n*** procr]

IP Network Map

Action	Object	Qualifier
change	**ip-network-map**	
display	**ip-network-map**	[**print** | **schedule**]

IP Network Regions

Action	Object	Qualifier
change	**ip-network- region**	region number ***n***
display	**ip-network- region**	***n*** [**print** | **schedule**]
status	**ip-network- region**	***n*** [**print** | **schedule**]

IP Node Name

Action	Object	Qualifier
change	**node-name ip**	[***node-name*** (up to 15 characters)]
list-usage	**node-name**	***node name*** [**print** | **schedule**]

IP Options System Parameters

Action	Object	Qualifier
display	**system-parameters ip-options**	***n*** [**print** | **schedule**]

IP Route

Action	Object	Qualifier
add	**ip-route**	***n*** | **next**
change	**ip-route**	***n***
display	**ip-route**	***n*** [**print** | **schedule**]

IP Services

Action	Object	Qualifier
change	**ip-services**	
display	**ip-services**	[**print** | **schedule**]

IP Station

Action	Object	Qualifier
reset	ip-station	[ip-phones | all-ip] [network-region *n* | all-regions]

ISDN Numbering Private

Action	Object	Qualifier
change	isdn private-numbering	
display	isdn private-numbering	[print | schedule]

ISDN Numbering Public/ Unknown

Action	Object	Qualifier
change	isdn public-unknown-numbering	
display	isdn public-unknown-numbering	[print | schedule]

ISDN Tandem Calling Party Number

Action	Object	Qualifier
change	tandem-calling-party-number	

ISDN-BRI Trunk Circuit Pack

Action	Object	Qualifier
add	**bri-trunk-board**	[***board location*** | ***gateway location***]
change	**bri-trunk-board**	[***board location*** | ***gateway location***]
display	**bri-trunk-board**	[cabinet ***n***; carrier (A-E); slot ***n*** [**print** | **schedule**]
remove	**bri-trunk-board**	[***board location*** | ***gateway location***]

Language Translations (Display Messages)

Action	Object	Qualifier
change	**display-messages auto-wakeup-dn-dst**	
display	**display-messages auto-wakeup-dn-dst**	[**print** | **schedule**]
change	**display-messages call-identifiers**	
display	**display-messages call-identifiers**	[**print** | **schedule**]
change	**display-messages date-time**	
display	**display-messages date-time**	[**print** | **schedule**]
change	**display-messages leave-word-calling**	
display	**display-messages leave-word-calling**	[**print** | **schedule**]
change	**display-messages malicious-call-trace**	
display	**display-messages malicious-call-trace**	[**print** | **schedule**]
change	**display-messages miscellaneous-features**	
display	**display-messages miscellaneous-features**	[**print** | **schedule**]
change	**display-messages property-management**	
display	**display-messages property-management**	[**print** | **schedule**]

Action	Object	Qualifier
change	display-messages softkey-labels	
display	display-messages softkey-labels	[print | schedule]
change	display-messages time-of-day-routing	
display	display-messages time-of-day-routing	[print | schedule]
change	display-messages transfer-conference	
display	display-messages transfer-conference	[print | schedule]
change	display-messages view-buttons	
display	display-messages view-buttons	[print | schedule]

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Listed Directory Numbers

Action	Object	Qualifier
change	listed-directory-numbers	
display	listed-directory-numbers	[print | schedule]

Locations

Action	Object	Qualifier
change	locations	
display	locations	[print | schedule]

Location Parameters

Action	Object	Qualifier
change	**location-parameters**	[***n***]
display	**location-parameters**	[***n***]

Login Administration

Action	Object	Qualifier
add	**login**	***login-id***
change	**login**	***login-id***
display	**login**	***login-id*** [**print** | **schedule**]

Local Survivable Processor

Action	Object	Qualifier
change	**lsp**	
display	**lsp**	

Loudspeaker Paging

Action	Object	Qualifier
change	**paging loudspeaker**	
display	**paging loudspeaker**	[**print** | **schedule**]

Measurements Announcements

Action	Object	Qualifier
list	**measurements announcements all**	[**yesterday-peak | today-peak | last-hour**] [**print** | **schedule**]
list	**measurements announcements integ-all**	[**yesterday-peak | today-peak | last-hour**] [**print** | **schedule**]
list	**measurements nnouncements board**	[**to cabinet** ***location*** **|** ***gateway location***] V9 [**yesterday-peak | today-peak | last-hour**] [**print** | **schedule**]
list	**measurements announcements-board-loc**	[**to cabinet** ***location*** | **disk |** ***gateway location***] V9 [**yesterday-peak | today-peak | last-hour**] [**schedule**]

Media Gateway

Action	Object	Qualifier
add	**media-gateway**	[***n*** | **next**]
list	**media-gateway**	[***n***] [**print** | **schedule**]
remove	**media-gateway**	[***n***]

Mode Code Related System Parameters

Action	Object	Qualifier
change	**system-parameters mode-code**	
display	**system-parameters mode-code**	[**print**]

Modem Pool Group

Action	Object	Qualifier
add	**modem-pool num**	[*n*]
change	**modem-pool num**	[*n*]
display	**modem-pool num**	[*n*] [**print** | **schedule**]
list	**modem-pool num**	[**print** | **schedule**]
remove	**modem-pool num**	[*n*]

Multifrequency-Signaling

Action	Object	Qualifier
change	**multifrequency-signaling**	[*n*]
display	**multifrequency-signaling**	[*n*]

Multiple Level Pecedence and Preemption (MLPP)

Action	Object	Qualifier
change	**system-parameters mlpp**	
display	**system-parameters mlpp**	

Music Sources

Action	Object	Qualifier
change	**music-sources**	[*n*]
display	**music-sources**	[*n*] [**print** | **schedule**]

Night Service

Action	Object	Qualifier
enable	**night-service**	[**attendant-group | hunt group | trunk-group**] ***group number***
disable	**night-service**	[**attendant-group | hunt group | trunk-group**] ***group number***

Packet Gateway Board

Action	Object	Qualifier
add	**pgate**	***board location***
change	**pgate**	***board location***
display	**pgate**	***board location***
remove	**pgate**	***board location***

Partition Route Table

Action	Object	Qualifier
change	**partition-route-table**	[**index** ***n***]
display	**partition-route-table**	[**index** ***n***] [**print** | **schedule**]

Pickup Group

Action	Object	Qualifier
add	**pickup-group**	[*n* | **next**]
change	**pickup-group**	[*n*]
display	**pickup-group**	[**number** *n*] [**to-number** *n*] [**count** *n*] [**print** | **schedule**]
remove	**pickup-group**	[*n*]

Port

Action	Object	Qualifier
display	**port**	***port n*** [**print** | **schedule**]

Precedence Routing Digit Analysis

Action	Object	Qualifier
change	**precedence-routing analysis**	[**'location' (1 to 250 or all)**] [**'min' (1 to 28)**]
display	**precedence-routing analysis**	**0 to 9**, '*****', '**x**', '**X**', [**'schedule'**]

Precedence Routing Digit Conversion

Action	Object	Qualifier
change	**precedence-routing digit-conversion**	**0 to 9**, '*****', '**x**', '**X**', [**'location' (1 to 250 or all)**]
display	**precedence-routing digit-conversion**	**0 to 9**, '*****', '**x**', '**X**', [**'min' (1 to 28)**], [**'schedule'**]

PRI Endpoint

Action	Object	Qualifier
add	**pri-endpoint**	***extension*** | **next**
change	**pri-endpoint**	***extension***
display	**pri-endpoint**	***extension*** [**print** | **schedule**]
list	**pri-endpoint**	***extension*** [**count** ***n***] [**print** | **schedule**]
remove	**pri-endpoint**	***extension***

QSIG to DCS TSC Gateway

Action	Object	Qualifier
change	**isdn qsig-dcs-tsc-gateway**	
display	**isdn qsig-dcs-tsc-gateway**	[**print** | **schedule**]

Remote Access

Action	Object	Qualifier
change	**remote-access**	**[n]**
display	**remote-access**	**[print | schedule]**
status	**remote-access**	

Remote Call Coverage Table

Action	Object	Qualifier
change	**coverage remote**	***n***
display	**coverage remote**	***n***

Remote Office

Action	Object	Qualifier
add	**remote-office**	[***n*** | **next**]
change	**remote-office**	[***n***]
display	**remote-office**	[***n***] [**print** | **schedule**]
list	**remote-office**	[**print** | **schedule**]
remove	**remote-office**	[***n***]
status	**remote-office**	[***n***] [**print**]

RHNPA Table

Action	Object	Qualifier
change	**rhnpa**	RHNPA and code n:xxx n***n*** x(0-MAX) x(0-MAX) x(0-MAX)
display	**rhnpa**	RHNPA and code n:xxx n***n*** x(0-MAX) x(0-MAX) x(0-9) [**print** | **schedule**]

Route Pattern

Action	Object	Qualifier
change	**route-pattern**	[*n*]
display	**route-pattern**	[*n*] [**print** | **schedule**]
list	**route-pattern**	[**trunk** *n*] [**service** /***feature name-string***] [**print** | **schedule**]

Security-Related System Parameters

Action	Object	Qualifier
change	**system-parameters security**	
display	**system-parameters security**	[**print** | **schedule**]

Signaling Group

Action	Object	Qualifier
add	**signaling-group**	[*n*] | **next**
change	**signaling-group**	[*n*]
display	**signaling-group**	[*n*] [**print** | **schedule**]
list	**signaling-group**	[*n*] [**print** | **schedule**]
remove	**signaling-group**	[*n*]

Site Data

Action	Object	Qualifier
change	site-data	
display	site-data	[print | schedule]
list	set-data	*extension*

Station

Action	Object	Qualifier
add	station	*extension* | next
change	station	*extension*
display	station	[ext *n*] [to-ext *n*] [count *n*] [print | schedule]
duplicate	station	*extension* [start *extension*] [board *location* | x | w | i] [count *n*] [one-ep]
list	station	[ext *n*] [to-ext *n*] [port *x*] [[type *x*] [movable *x*] [count *n*] [print | schedule]
status	station	*extension* [print]

System Parameters Call Coverage/Call Forwarding

Action	Object	Qualifier
change	system-parameters coverage-forwarding	
display	system-parameters coverage-forwarding	[print | schedule]

System Parameters Country-Options

Action	Object	Qualifier
display	system-parameters country-options	[print | schedule]

System Parameters Crisis Alert

Action	Object	Qualifier
change	system-parameters crisis-alert	
display	system-parameters crisis-alert	[print | schedule]

System Parameters Customer-Options

Action	Object	Qualifier
display	system-parameters customer-options	[print | schedule]

System Parameters IP Options

Action	Object	Qualifier
change	system-parameters ip-options	
display	system-parameters ip-options	[print | schedule]

Telecommuting Access

Action	Object	Qualifier
change	**telecommuting-access**	
display	**telecommuting-access**	[**print** | **schedule**]

Tenant

Action	Object	Qualifier
change	**tenant**	[*n*]
display	**tenant**	[*n*] [**print** | **schedule**]

Terminal Parameters

Action	Object	Qualifier
change	**terminal-parameters**	[*n*]
display	**terminal-parameters**	[*n*][**schedule**]

Terminating extension Group

Action	Object	Qualifier
add	**term-ext-group**	[*n*] | **next**
change	**term-ext-group**	[*n*]
display	**term-ext-group**	[*n*] [**print** | **schedule**]
list	**term-ext-group**	[**number** *n*] [**to-number** *n*] [**name** *x*] [**ext** *n*] [**to-ext** *n*] [**count** *n*] [**print** | **schedule**]
remove	**term-ext-group**	*n*

TFTP Server

Action	Object	Qualifier
change	**tftp-server**	[**print** | **schedule**]
display	**tftp-server**	[**print** | **schedule**]

Time of Day Routing Plan

Action	Object	Qualifier
add	**coverage time-of-day**	***n*** | **next**
change	**coverage time-of-day**	***n***
display	**coverage time-of-day**	***n*** [**print** | **schedule**]
remove	**coverage time-of-day**	***n***

Time of Day Routing Plan

Action	Object	Qualifier
change	**time-of-day**	***n*** (***routing plan number***)
display	**time-of-day**	***n*** (***routing plan number***) [**print** | **schedule**]

Toll Analysis

Action	Object	Qualifier
change	**toll**	***n 0-9*** | *** | x** | **X** [**location** (***n*** | **all)**] [**min** 1-28]
display	**toll**	***n 0-9*** | *** | x** | **X** [**location** (***n*** | **all)**] [**min** 1-28] [**print** | **schedule**]
list	**toll toll-list**	[**start** ***dialed-string***] [**count** ***n***] [**location** (***n*** | **all)**] [**print** | **schedule**]
list	**toll all**	[**start** ***dialed-string***] [**count** ***n***] [**location** (***n*** | **all)**] [**print** | **schedule**]
list	**toll restricted-call**	[**start** ***dialed-string***] [**count** ***n***] [**location** (***n*** | **all)**] [**print** | **schedule**]
list	**toll unrestricted-call**	[***n***] [**start** ***dialed-string***] [**count** ***n***] [**location** (***n*** | **all**] [**print** | **schedule**]

Tone Generation

Action	Object	Qualifier
change	**tone-generation**	[***n***]
display	**tone-generation**	[***n***]

Trace

Action	Object	Qualifier
list	**trace attendant**	***n*** [**print** | **schedule**]
list	**trace data-module**	[**ext** ***n***] [**to-ext** ***n***] [**name** ***x***][**count** ***n***] [**print** | **schedule**]
list	**trace ewt**	
list	**trace ras**	
list	**trace station**	***extension*** | **ext/a** (atm) | **q** (qsig) **ext** ***1-40*** (button ***n***)] [**print**]
list	**trace tac**	***Trunk access code n*** **| tac/a** (atm) | **ac/d** (digits) **q (**qsig) [**print**]
list	**trace vdn**	***extension*** [**print** | **schedule**]
list	**trace vector**	***n***

Trunk Groups

Action	Object	Qualifier
add	**trunk-group**	1 to MAX | **next**
change	**trunk-group**	1 to MAX | TAC X...
display	**trunk-group**	1 to MAX [**number** ***n***] [**to-number** ***n***] [**count** ***n***] [**tac** ***n***] [**print** | **schedule**]
list	**trunk-group**	[**number** ***n***] [**to-number** ***n***] [**name** ***x***] [**type** ***x***] [**count** ***n***] [**print** | **schedule**]
remove	**trunk-group**	1 to MAX | TAC X...

Uniform Dial Plan

Action	Object	Qualifier
change	**uniform-dialplan**	***number 0-9***
display	**uniform-dialplan**	***number 0-9*** [**print** | **schedule**]
list	**uniform-dialplan**	[**start** ***matching-pattern***] [**len** *(3-7)*] [**insert** *(0-MAX)*] [**net** ***x***] [**node** ***n***] [**to-node** ***n***] [**count** ***n***] [**print** | **schedule**]

Usage CTI-Link

Action	Object	Qualifier
list	**usage cti-link**	**link** ***number*** [**print** | **schedule**]

Usage extension

Action	Object	Qualifier
list	**usage** ***extension***	***extension*** [**vector**] [**print** | **schedule**]

Vector

Action	Object	Qualifier
change	**vector**	***n***
list usage	**vector**	***n***

Vector Directory Numbers

Action	Object	Qualifier
add	**vdn**	***extension*** | **next**
change	**vdn**	***extension***
display	**vdn**	***extension*** [**print** | **schedule**]

XMOBILE Configuration Set

Action	Object	Qualifier
change	**xmobile configuration-set**	digits 1 to 10
display	**xmobile configuration-set**	digits 1 to 10

XMOBILE station to cell telephone mapping

Action	Object	Qualifier
list	**xmobile mapping**	***cell telephone number*** **|** ***partial string**** **|** **all** [**print** | **schedule**]

Partial string of a cell telephone number followed by the wildcard "".

Chapter 19: Telephone Reference

This reference section describes many of the telephones and adjuncts that you can connect to servers running Avaya Communication Manager.

Use this section to:

- determine where to connect a telephone—is it analog, digital, hybrid, or IP? Is it designed for a 2-wire or 4-wire environment?
- determine whether a telephone will fit your users' needs—does it accommodate enough feature buttons? does it include a display?
- understand how to assign the user's feature buttons—how do the buttons on the **Station** screen map to buttons on the physical telephone.

Because you need to know how the physical buttons on the telephone relate to your button assignments on the **Station** screen, this section includes figures for those phones to which you can assign feature buttons.

This section includes descriptions of the following telephones:

- 500 telephones on page 648
- 2400 telephones on page 648
- 2500-series telephones on page 649
- 4600-series IP telephones on page 649
- 6200-series telephones on page 651
- 6400-series telephones on page 655
- 7100-series telephones on page 659
- 7300-series telephones on page 660
- 731x-series hybrid telephones on page 663
- 7400-series telephones on page 668
- ISDN telephones (7500s & 8500s) on page 684
- 8110 telephones on page 689
- 8400-series telephones on page 689
- CALLMASTER telephones on page 696
- Cordless telephone on page 699
- Internet Protocol (IP) Softphones on page 703

500 telephones

The 500 telephones are single appearance analog rotary-dial telephones which provides cost-effective service wherever it is located. It provides limited access to features because the rotary dial has no * or # positions.

2400 telephones

2402 Telephone

The 2402 Telephone is a low-cost, low function DCP telephone. The 2402 has a 2 x 24 display, and four groups of permanently-labeled feature buttons (Speaker, Mute, Volume, Hold, Conference, Transfer, Redial, Drop, Messages, a, b, Feature). The 2402 Telephone is a Class B device.

2410 telephone

The 2410 Telephone is a mid-range, two-wire, circuit-switched digital telephone. The 2410 has a display that is similar to the 4610, but is used in character mode (text only). A single set type will be able to handle both Eurofont and Katakana fonts. The 2410 is electrically similar to the 2420. The LCD can display a minimum of 29 characters, in five lines. There are six programmable call appearance/feature buttons (two pages of six buttons for a total of 12 programmable buttons). There are four applications buttons (Call Log, Speed Dial, Options, Label), a built-in speakerphone, fixed feature buttons, and downloadable firmware. The user interface of the 2410 is similar to the 2420. The telephone is a Class B device. The 2410 does not have an EU24 port or modular capability.

2420 telephone

The 2420 telephone is a digital telephone with an optional feature expansion module and downloadable call appearance/feature buttons information. The 2420 Digital Communications Protocol (DCP) telephone does not need paper labels. The button information appears on a screen on the telephone. The firmware for the 2420 can be upgraded over its DCP connection to the switch. A new firmware image first must be downloaded into switch memory from a local trivial file transfer protocol (TFTP) server.

Avaya recommends that you administer call appears only on the first 8 call appearance/feature buttons on the 2420 telephone.

2500-series telephones

The 2500-series telephones consist of single appearance analog telephones with conventional touch-tone dialing. You can allow 2500-series phones users to access features by giving them the appropriate feature access codes. For more information about providing feature access codes to your users, see Changing Feature Parameters on page 57.

4600-series IP telephones

The 4600-series telephones are DCP telephones that use Internet Protocol (IP) technology with Ethernet line interfaces. Firmware for these phones can be downloaded from the internet. The 4600 series includes a 2-button set, 6-button sets, a 12-button set, and a 24-button set.

Each of the 4600-series phones includes 6 standard feature buttons:

- SPEAKER button, which can access a 2-way speakerphone or allow group listen
- MUTE button, which mutes the handset or speakerphone microphone
- HOLD button
- REDIAL button
- TRANSFER/TEST button for transferring a call or testing the lights and display on the telephone
- CONF/RING button for setting up a conference call and for selecting a personalized ringing pattern.

The 4604, 4612, and 4624 phones do not have a standard Drop button, but you can assign a drop button to any feature button. The 4600-series display phones show the date and time in Normal mode, so you do not have to assign a Date/Time button to these phones.

4601 IP telephone

The 4601 Telephone is a low-cost VoIP telephone for Avaya. It is based on the 4602, but will not have an LCD display. The telephone supports H.323 Avaya proprietary call processing. It supports two call appearances, and has no speakerphone. The 4601 looks similar to the 4602, but has some changes to the buttons (no SPKR, no MUTE), LEDs, and bezel. The 4601 is available in both a gray and white version. The power options are the same as the 4602; Power over Ethernet 802.3af. The telephone stand assembly is identical to that of the 4602. The 4601 Telephone is a Class B device.

4602/4602SW IP telephone

The 4602 IP telephone has separate LEDs to indicate the on/off status of the speakerphone and mute buttons. The 4602 has a 2x24 display. The 4602 has no administrable feature buttons, 2 fixed call appearance buttons, one-way speakerphone or no speakerphone option, a fixed Drop button, a listen-only Speaker button, a redial button, and a fixed voice mail retrieval button that provides automatic access to voice mail.

4606 IP telephones

The 4606 IP telephone is a multi-appearance digital telephone with six call appearance/feature buttons: a red Hold button, a Redial button, a Transfer button that can also be used for the Test feature, and a Conference button that can also be used to select a personalized ringing pattern, a red Message light, and a Volume control button.

4610SW IP telephone

The 4610SW is intended to bridge the market gap between two existing members of the 46xx IP Telephone family: the low-end 4602, and the higher-end 4620. With a 168-by-80 pixel 4-grayscale display, 4 softkeys, 6 dynamically labeled call appearance/feature buttons, and 4 unique fixed feature buttons, the 4610SW provides advanced feature functionality with an intuitive and innovative user interface. Although the 4610SW display lacks the size of the 4620's display, the 4610SW can nonetheless provide telephony, Speed Dial, Call Log, and Options functionality analogous to the 4620.

the 4610SW uses an upper housing design as similar to the 4620 upper housing as possible. Similarly, for consistency of user experience, the 4610SW user interface as specified herein is as consistent as possible to that of the 4620, subject to the constraint of the smaller display.

4612 IP telephone

The 4612 IP telephone is a multi-appearance digital telephone with 12 call appearance/feature buttons. With the 4612 IP telephone the end-user can access 12 features with the softkeys and display control buttons.

4620/4620SW IP telephone

The 4620/4620SW IP telephone comes in two models: the 4620 IP, which uses G3 hardware, and the 4620SW IP, which uses G3.5 hardware.

The 4620 IP telephone has an optional feature expansion module and downloadable call appearance/feature button information. It also includes features such as speed dial, call log, and Web browsing using the Wireless Markup Language (WML). The 4620 IP telephone does not need paper labels. The button information appears on a screen on the telephone.

The 4620 uses icons to indicate the status of call appearances, bridge call appearances and features. The telephone maintains a call log with calling party and called party information. The 4620 has a local button for headset on/off. The button label information for the 4620 is downloaded automatically to the telephone when a link is established between the switch and the telephone. There are three speakerphone options on the 4620. Labels on the 4620 can be downloaded in English, French, Italian, Spanish, user-defined, or Unicode. As of the January 2005 2.2 Release of Communication Manager, Unicode support is available for

4624 IP telephone

The 4624 IP telephone is a multi-appearance digital telephone with 24 call appearance/feature buttons. With the 4624 telephone the end-user can access 12 features with the softkeys and display control buttons. These 12 features can be used in addition to the features you assign to the call appearance/feature buttons.

6200-series telephones

6210 telephone

The 6210 telephones are single-line analog models. They have fixed Flash, Redial, and Hold feature buttons and a message waiting light.

6218 telephone

The 6218 telephones are single-line analog models. They have 10 programmable dialing buttons. These phones also have fixed Flash, Redial, and Hold feature buttons and a message waiting light.

6220 telephone

The 6220 telephones are single-line analog models. These phones also have fixed Flash, Redial, Hold, Mute, and Speakerphone (Spkr) feature buttons and a message waiting light. They also have handset volume control, ringer volume control, timed switch-hook disconnect, 10 programmable dialing buttons, repertory keylock, set personalized ring, and system hold.

Figure 217: The 6210 telephone

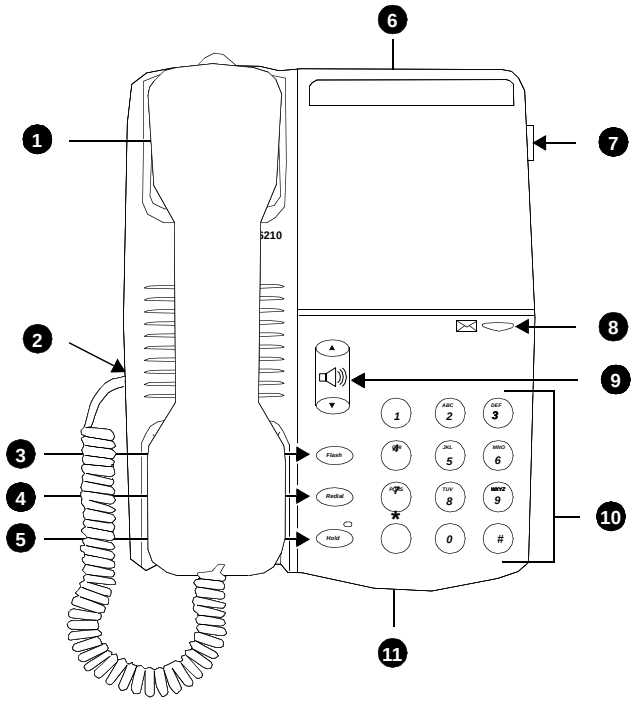


Figure notes:

1. **Handset**
2. **Handset cord jack**
3. **Flash button**
4. **Redial button**
5. **Hold button**
6. **DATA jack**
7. **Ringer volume control**
8. **Message light**
9. **Handset volume control**
10. **Dial pad**
11. **LINE jack (on bottom of telephone)**

Figure 218: The 6218 telephone

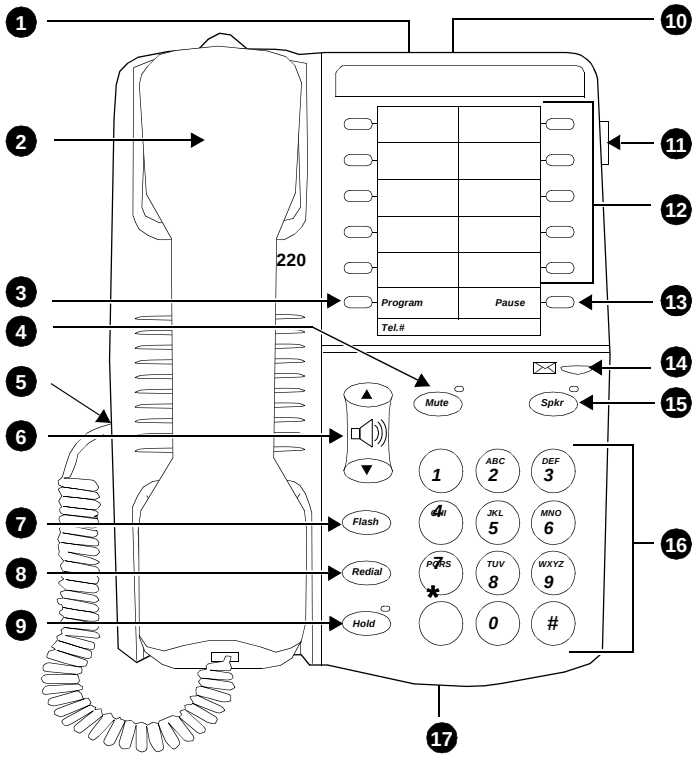


Figure notes:

1. **Handset parking tab**
2. **Handset**
3. **Program button**
4. **Handset cord jack**
5. **Flash button**
6. **Redial button**
7. **Hold button**
8. **Data jack**
9. **Personalized ring**
10. **Ringer volume control**
11. **10 programmable dialing buttons**
12. **Pause button**
13. **Message light**
14. **Handset/speakerphone volume control**
15. **Dial pad**
16. **LINE jack (on bottom of telephone)**

Figure 219: The 6220 telephone

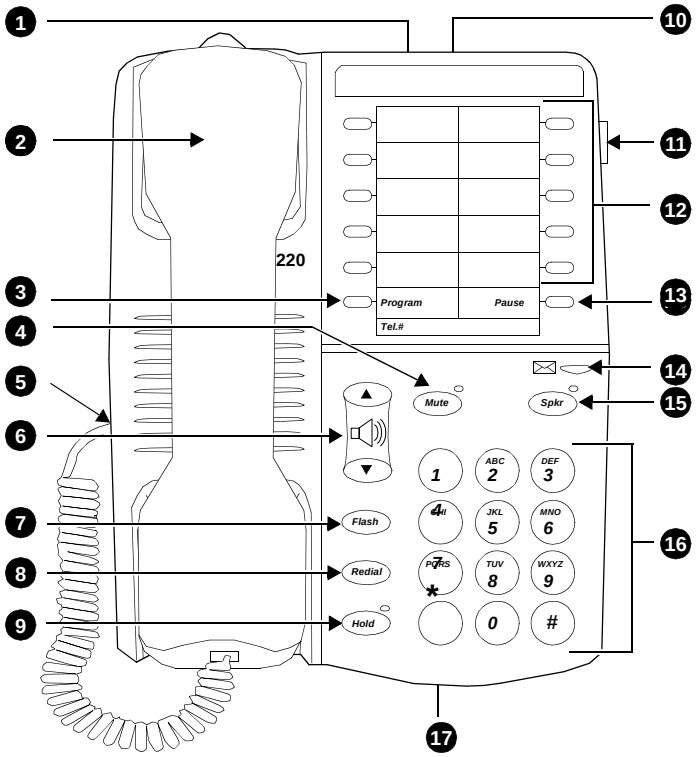


Figure notes:

1. **DATA jack**
2. **Handset**
3. **Program button**
4. **Mute button**
5. **Handset cord jack**
6. **Handset/speakerphone volume control**
7. **Flash button**
8. **Redial button**
9. **Hold button**
10. **Personalized ring**
11. **Ringer volume control**
12. **10 programmable dialing buttons**
13. **Pause button**
14. **Message light**
15. **Speakerphone button**
16. **Dial pad**
17. **LINE jack (on bottom of telephone)**

6400-series telephones

The 6400-series telephones are DCP 2-wire telephones that work with Avaya Communication Manager. The last two digits of the 6400-series model number identify the number of call appearances (2-lamp buttons) for that model. For example, the 6424D has 24 call appearances. The 6400 series includes two single-line sets (6402 and 6402D), 8-button sets, a 16-button set, a 24-button set, and a 24-button expansion module (for the 6416D+ and 6424D+ telephones).

Each of the 6400-series phones includes 6 standard feature buttons:

- SPEAKER button, which can access a 2-way speakerphone or allow group listen
- MUTE button, which mutes the handset or speakerphone microphone
- HOLD button
- REDIAL button
- TRANSFER/TEST button for transferring a call or testing the lights and display on the telephone
- CONF/RING button for setting up a conference call and for selecting a personalized ringing pattern.

These phones do not have a standard Drop button, but you can assign a drop button to any feature button. The 6400-series display phones show the date and time in Normal mode, so you do not have to assign a Date/Time button to these phones.

6402 telephones

The 6402 is a single-line telephone with six fixed feature buttons: a listen-only Speaker button, a Feature button that allows you to use the dial pad keys for up to 12 features assigned by the system manager, a red Hold button, a Redial button, a Transfer button that can also be used for the Test feature, and a Conference button that can also be used to select a personalized ringing pattern, a red Message light, and a Volume control button.

The 6402D is the same as a 6402, but also has a 2-line by 16-character LCD display.

The 6402 and 6402D can be used in only a 2-wire environment.

6408 telephones

The 6408 is a multi-appearance digital telephone with eight call appearance/feature buttons.

The 6408 telephone is available in the following four models:

- 6408 telephone—includes a 1-way, listen-only speaker, and no display
- 6408+ telephone—includes a 2-way speakerphone, and no display
- 6408D telephone—includes a 1-way, listen-only speaker, and a 2-line by 24-character display
- 6408D+ telephone—includes a 2-way speakerphone, and a 2-line by 24-character display.

With the 6408D and 6408D+ telephones, the end-user can access 12 features with the softkeys and display control buttons. These 12 features can be used in addition to the features you assign to the call appearance/feature buttons.

The 6408, 6408+, 6408D, and 6408D+ telephones can work only in 2-wire environments.

6416D+ telephone

The 6416D+ telephone is a multi-appearance digital telephone with 16 call appearance/feature buttons.

With the 6416D+ telephone the end-user can access 12 features with the softkeys and display control buttons. These 12 features can be used in addition to the features you assign to the call appearance/feature buttons.

Note:

You can connect an XM24 expansion module to the 6416D+ telephone to expand the number of buttons you can assign. However, when the expansion module is connected, you must connect an auxiliary power supply to the telephone.

6424D+ telephone

The 6424D+ telephone is a multi-appearance digital telephone with 24 call appearance/feature buttons. With the 6424D+ telephone the end-user can access 12 features with the softkeys and display control buttons. These 12 features can be used in addition to the features you assign to the call appearance/feature buttons.

The 6424D+ telephone can work in both 4-wire and 2-wire environments.

Note:

You can connect an XM24 expansion module to a 6424D+ telephone to expand the number of buttons you can assign. However, when the expansion module is connected, you must connect an auxiliary power supply to the telephone.

Figure 220: 6402D telephone

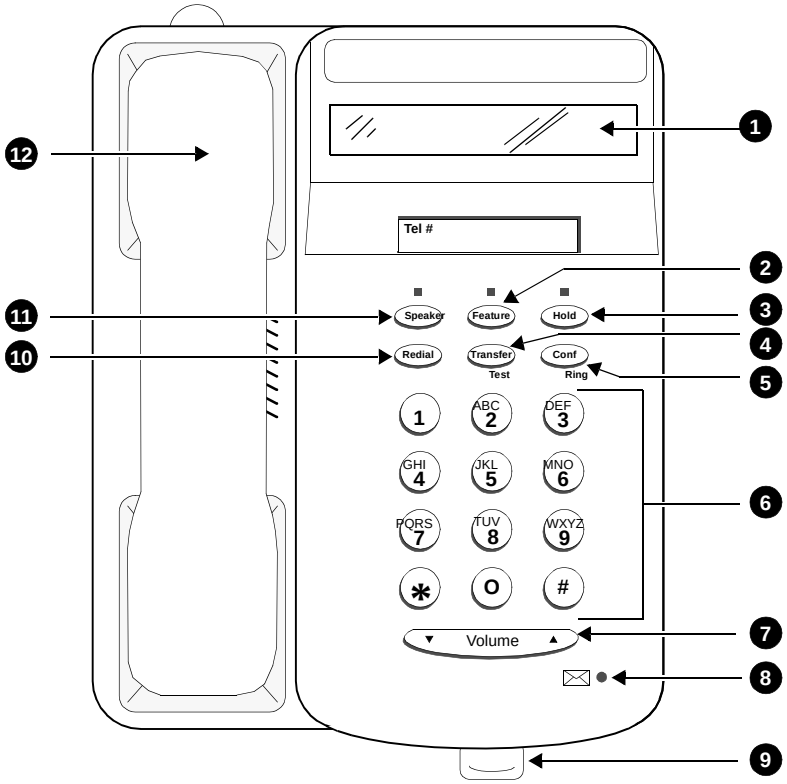


Figure notes:

1. **Display**
2. **Feature button**
3. **Hold button**
4. **Transfer/Test button**
5. **Conf/Ring button**
6. **Dial pad**
7. **Volume control button**
8. **Message light**
9. **Tray handle (includes reference cards)**
10. **Redial button**
11. **Speaker button**
12. **Handset**

Figure 221: 6408D telephone

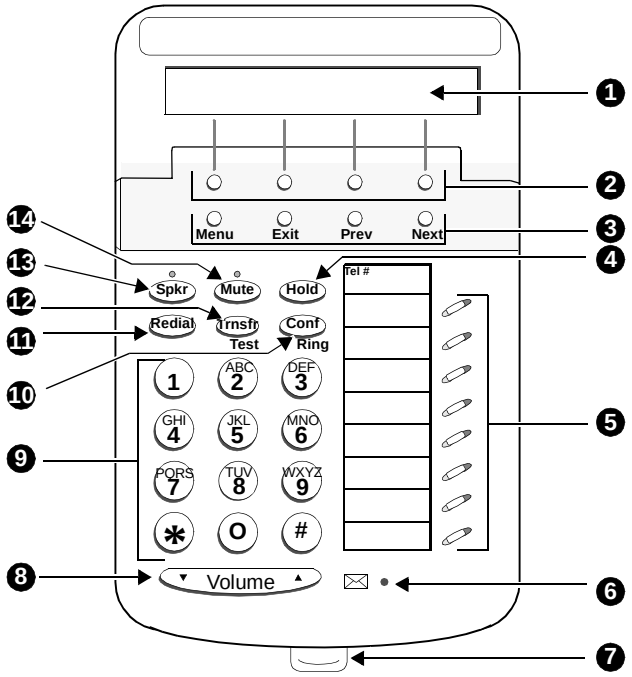


Figure notes:

1. **Display**
2. **Softkeys**
3. **Display control buttons**
4. **Hold**
5. **Call appearance/feature buttons**
6. **Message light**
7. **Tray handle (includes reference cards)**
8. **Volume control button**
9. **Dial pad**
10. **Conf/Ring button**
11. **Redial button**
12. **Transfer/Test button**
13. **Speaker button**
14. **Mute button**

7100-series telephones

7101A telephone

The 7101A telephone is a single-line analog model that is equipped with a Message Waiting light and a handy Recall button for activating the system's special features. It cannot be physically bridged to the same analog line port due to the message waiting and loop current circuitry.

7102A telephone

The 7102A (7102A01A) and 7102+ (7102A01B), called the 7102 Plus. The front of the two sets is exactly the same in appearance. The only difference is that the 7102A01B is equipped with an adjunct jack. This jack allows speakerphone/headset capability.

7103A telephone

The 7103A fixed feature telephone is a single-line analog model which has been discontinued. The feature buttons on this telephone must be programmed by the system manager. It cannot be physically bridged to the same analog line port due to the message waiting and loop current circuitry.

7104A telephone

The 7104A telephone is a single-line analog model that is equipped with a display that is used to display stored numbers. It cannot be physically bridged to the same analog line port due to the message waiting and loop current circuitry.

7300-series telephones

7302H MERLIN telephone

The 7302H is a 5-button telephone that can be desk or wall mounted. This set can no longer be ordered.

Administer 7302H telephones (5-Button) as a 7303S model.

7303H MERLIN telephone

The 7303H is a 10-button telephone that can be desk or wall mounted.

Administer 7303H telephones (10-Button) as a 7305S model. You can administer only the first 12 feature function buttons. Of these 12 buttons, 8 have two lamps and 4 have no lamps. However, the system treats the 8 double-lamp buttons as though they have a single (green) lamp. Administer the 4 buttons (with no lamps) with features that do not require status indications.

7303S telephone

The 7303S is a multi-appearance hybrid telephone which provides access to 10 line appearances or selected programmable features. The 7303S telephone is also equipped with six fixed feature buttons. It requires 3-pair wiring for operation. One wire pair is used for analog voice, while the other two pairs are used for digital control and signaling.

7305H MERLIN telephone

The 7305H series is a 34-button telephone. This telephone can be equipped with or without different features such as built-in speakerphone or display. Some versions of the 34-button series can be ordered using PEC code 3162 and the appropriate suffix.

Administer 7305H telephones (34-Button) as a 7305S model. The system treats the telephone’s 24 feature function buttons (two lamps each) as single (green) lamp function buttons.

7305S telephone

The 7305S telephone is a multi-appearance hybrid telephone which provides access to 10 line appearances. The 10 line appearance buttons can also be used as programmable feature buttons. The 7305S telephone is also equipped with 24 programmable feature buttons and six fixed feature buttons. It requires 3-pair wiring for operation. One wire pair is used for analog voice, while the other two pairs are used for digital control and signaling.

Figure 222: 7303S telephone

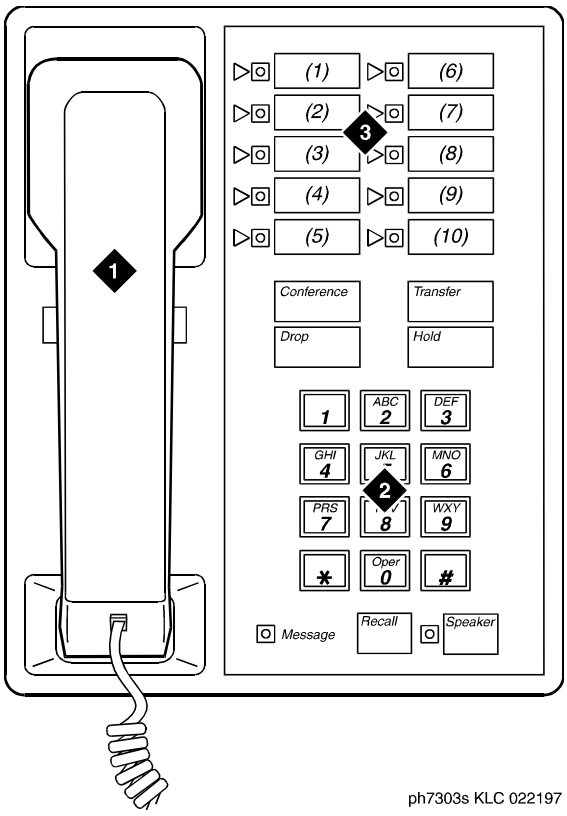


Figure notes:

1. **Handset**
2. **Dial pad**
3. **10 programmable buttons**

Figure 223: 7305S telephone

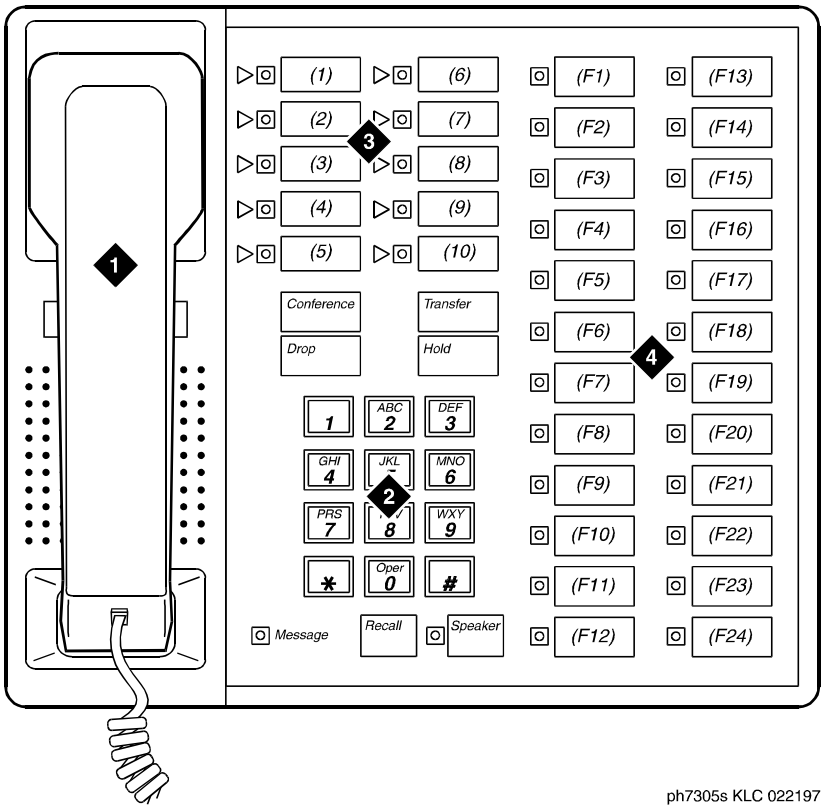


Figure notes:

1. **Handset**
2. **Dial pad**
3. **10 programmable buttons**
4. **24 feature buttons**

731x-series hybrid telephones

You should note that the following restrictions apply to administering hybrid telephones in the system:

- Hybrid telephones equipped with displays cannot be used as Automatic Call Distribution (ACD) agents and do not allow your end-users to access the Directory.
- The following fixed feature buttons do not operate on Hybrid telephones: STOP, PAUSE, RECALL, MESSAGE, HFAI, and HFAI/MIC. If you want users to have Hands Free Automatic Answer on Intercom (HFAI), assign Internal Automatic Answer (IAA) to a lamp button.
- These telephones support Leave Word Calling (LWC), but users cannot retrieve messages with the display.

Figure 224: 7313H telephone (BIS 10)

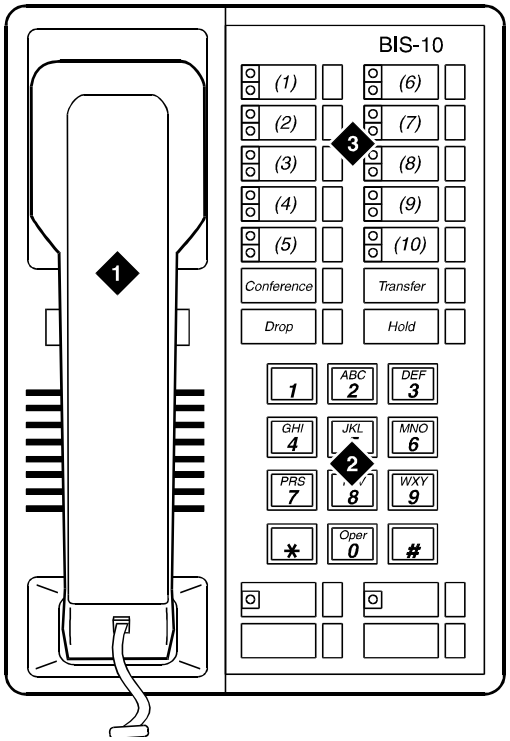


Figure notes:

1. **Handset**
2. **Dial pad**
3. **10 programmable buttons**

Figure 225: 7314H telephone (BIS 22)

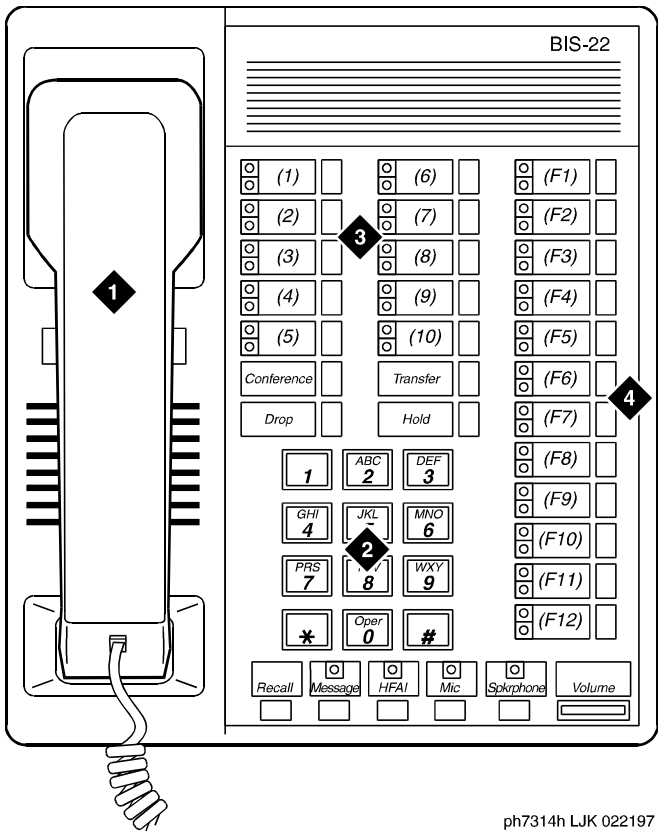


Figure notes:

1. **Handset**
2. **Dial pad**
3. **10 programmable buttons**
4. **12 feature buttons**

Figure 226: 7315H telephone (BIS 22D)

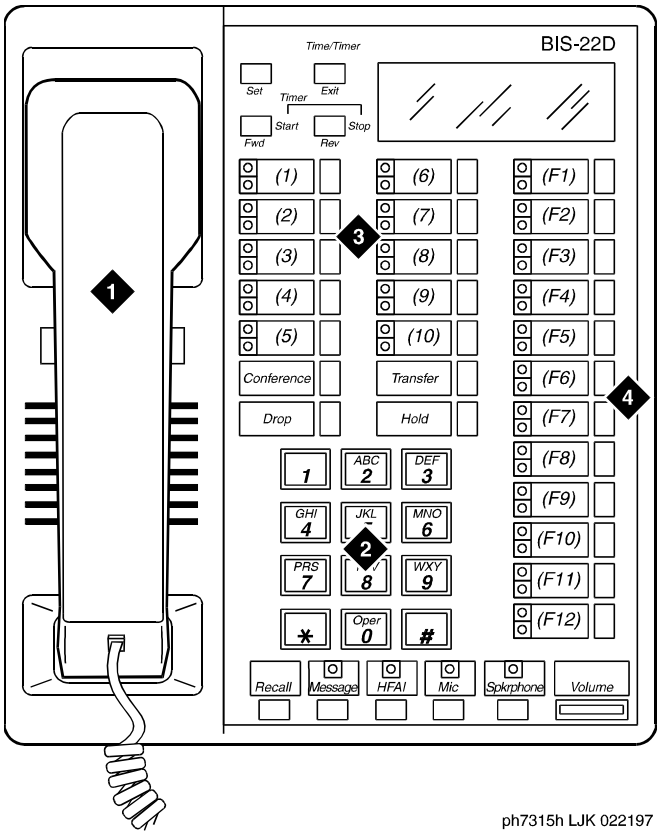


Figure notes:

1. **Handset**
2. **Dial pad**
3. **10 programmable buttons**
4. **12 feature buttons**

Figure 227: 7316H telephone (BIS 34)

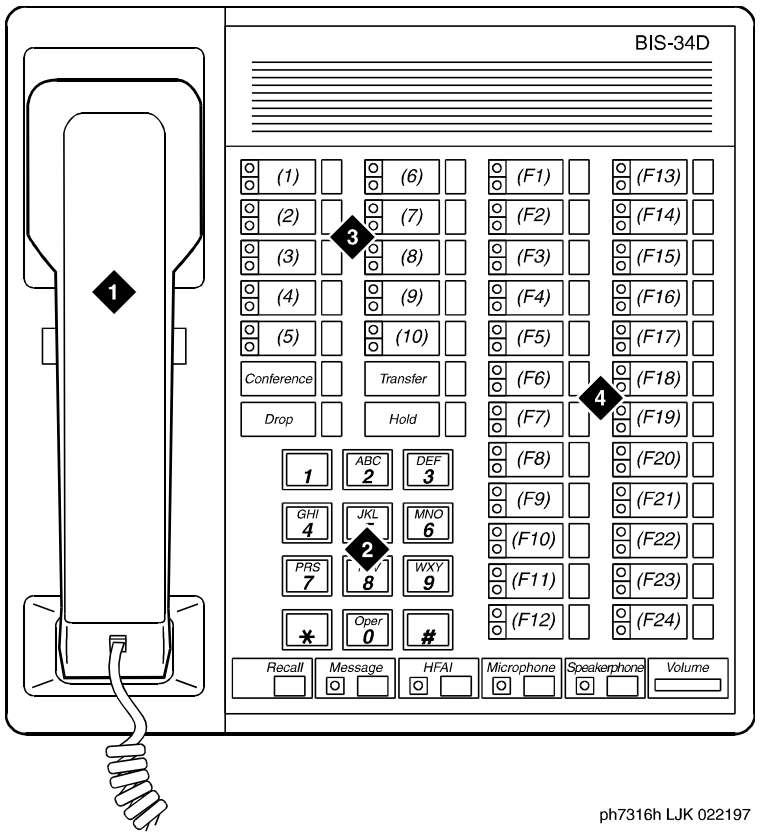


Figure notes:

1. **Handset**
2. **Dial pad**
3. **10 programmable buttons**
4. **24 feature buttons**

Figure 228: 7317H telephone (BIS 34D)

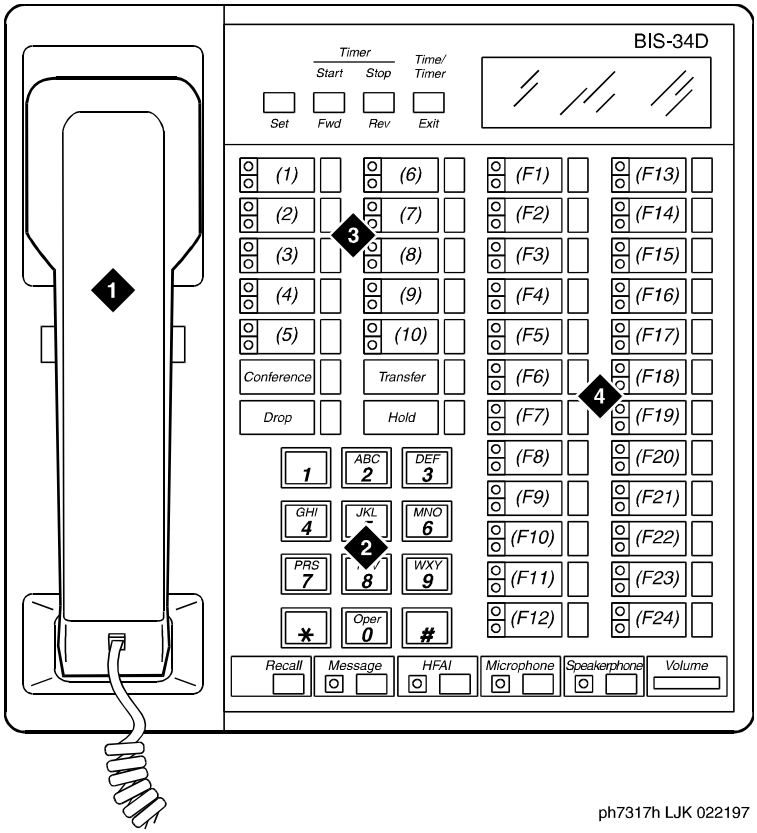


Figure notes:Figure Notes

1. **Handset**
2. **Dial pad**
3. **10 programmable buttons**
4. **24 feature buttons**

7400-series telephones

7401D telephone

The 7401D (7401D01A) and the 7401+ (7401D02A) are both single appearance digital phones which have no call appearance buttons or lights, but have two virtual call appearances. Depending on how the 7401D or the 7401+ telephone is administered, the second call appearance may be restricted to incoming priority calls and outgoing calls only.

7403D telephone

The 7403D is a multi-appearance digital telephone which has 10 buttons available for line appearances, one-touch feature access, or Speed Dialing. In addition, the 7403D telephone may be equipped with a Digital Terminal Data Module (DTDM) which attaches to the right side and allows the connection of a EIA RS-232C data terminal.

7404D telephone

The 7404D is a multi-appearance digital telephone which provides simultaneous voice/asynchronous data transmission. All the transmission is done over the same two pairs of wire.

7405D telephone

The 7405D telephone is a multi-appearance digital telephone which allows features to be added as the user needs them. The Digital Display can be added to provide access to the Message Center. A Digital Terminal Data Module or 7400B can be added to enable the user of a 7405D telephone to transmit or receive data with an associated data terminal.

The basic 7405D provides 10 call appearance/feature buttons with lights that can be assigned to call appearances or system features. It has 24 programmable feature buttons and six fixed-feature buttons. The 7405D can also have a function key module which adds 24 feature buttons and a call coverage module (when no display module is used) which adds 20 call appearance/feature buttons.

7406 telephones

The 7406D telephone (7406D01A, 7406D02A, 7406D03A, and 7406D04A models) has five call appearance/feature buttons, each with a red in-use light and a green status light, seven shiftable (2-level) programmable feature buttons with no lights, four programmable feature buttons with a green light, four fixed feature buttons (CONFERENCE, TRANSFER, DROP, and HOLD), a SHIFT button with a green light, a SPEAKER button, and a green Message light.

The 7406BIS telephone (7406D05A and 7406D06A models) has five call appearance/feature buttons, each with a red in-use light and a green status light, seven shiftable (2-level) programmable buttons with no lights, two programmable feature buttons with a green light, four fixed feature buttons (CONFERENCE, TRANSFER, DROP, and HOLD), a SHIFT button with a green light, a SPEAKER button with a green light, a MUTE button with a red light, a SPEAKER VOLUME "arrow" button, and a red Message light.

The 7406+ telephone (7406D07A and 7406D08A models) has five call appearance/feature buttons, each with a red in-use light and a green status light, three shiftable (2-level) programmable feature buttons with a green light, six shiftable (2-level) programmable feature buttons without lights, four fixed feature buttons (CONFERENCE, TRANSFER, DROP, and HOLD), a SELECT button with a green light, a SPEAKER/RESET SPKR button with a green light, a MUTE button with a red light, a VOLUME "arrow" button, and a red Message light.

7407+ telephone

The 7407D, Enhanced 7407D, and 7407+ telephones are multi-appearance digital telephones which provide digital voice, display, and data capabilities (the latter with the 7400B+ Data Module).

There are three versions of the 7407D telephone:

- 7407+ (7407D02D) — offers 10 call appearance buttons, each with a red in-use light and a green status light, four standard fixed feature buttons (CONFERENCE, DROP, HOLD, and TRANSFER), three fixed feature buttons with one light each (SELECT, SPEAKER/RESET SPKR, and MUTE), nine feature buttons with one light each (the uppermost two buttons can be used for voice or display features, the lower seven buttons for display features), 22 flexible feature buttons with no lights, a Message light, personalized ringing, a built-in speakerphone with a reset and listen-only option, and a built-in 2-line by 40-character display.
- The 7407D (the 7407D01B)—offers 10 call appearance/feature buttons, each with a red in-use light and a green status light, four standard fixed feature buttons (CONFERENCE, DROP, HOLD, and TRANSFER), three fixed feature buttons with an associated light (CALCULATOR/SELECT RING, SPEAKERPHONE, and MICROPHONE), nine programmable feature buttons with lights (the two uppermost buttons can be used for voice or display features, the lower seven for display features), 11 dual-function buttons, 22 programmable feature-only buttons without lights, a Message light, a DISPLAY button

that turns the display on and off, personalized ringing, a built-in speakerphone, a 2-line by 40-character liquid crystal display, and a built-in calculator. This set is AC powered.

- Enhanced 7407D (the 7407D02C)—offers 10 call appearance buttons, each with a red in-use light and green status light, four standard fixed feature buttons (CONFERENCE, DROP, HOLD, and TRANSFER), three fixed feature buttons with an associated light (SELECT RING, SPEAKER, and MUTE), 22 programmable feature-only buttons without lights, nine programmable feature buttons with one light each (the uppermost two buttons can be used for voice or display features, the lower seven for display features), a Message light, personalized ringing, a built-in speakerphone, a connection for an adjunct speakerphone or headset, a speakerphone with spokesman, and Mute option, and a 2-line by 40-character display.

7410D and 7410+ telephones

The 7410D (7410D01A) and 7410+ (7410D02A) are both multi-appearance digital telephones with 10 line appearances, four standard fixed feature buttons and a Select Ring button. The 7410D and 7410+ telephones also provide a Message light, Ringer Volume control, a Speakerphone/headset adapter jack. You can activate a Self-Test feature to test the lights and tone ringer on the telephone. In addition, the 7410+ provides a Speaker for listening-only functions.

7434D telephone

The 7434D is a multi-appearance digital telephone that offers 34 call appearance/feature buttons, each with a red in-use light and a green status light, four standard fixed feature buttons (CONFERENCE, DROP, HOLD, and TRANSFER), three fixed feature buttons with one light each (SELECT, SPEAKER/RESET SPKR, and MUTE), seven display feature buttons with one light each, a Message light, personalized ringing, a built-in speakerphone with a reset option, and a built-in 2-line by 40-character display. You can connect this telephone to a digital line port. The 7434D telephone supports an adjunct display module or a call coverage module.

7444D telephone

The 7444 telephone is a multi-appearance digital telephone that offers 34 call appearance/ feature buttons, each with a red in-use light and a green status light, four standard fixed feature buttons (CONFERENCE, DROP, HOLD, and TRANSFER), three fixed feature buttons with one light each (SELECT, SPEAKER/RESET SPKR, and MUTE), seven display feature buttons with one light each, a Message light, personalized ringing, a built-in speakerphone with a reset option, and a built-in 2-line by 40-character display. You can connect this telephone to a digital line port. It is powered from the switch.

Note:

The 7444 is powered by the switch, however, to use the display, you must connect an auxiliary power supply to the telephone.

Figure 229: 7401D telephone

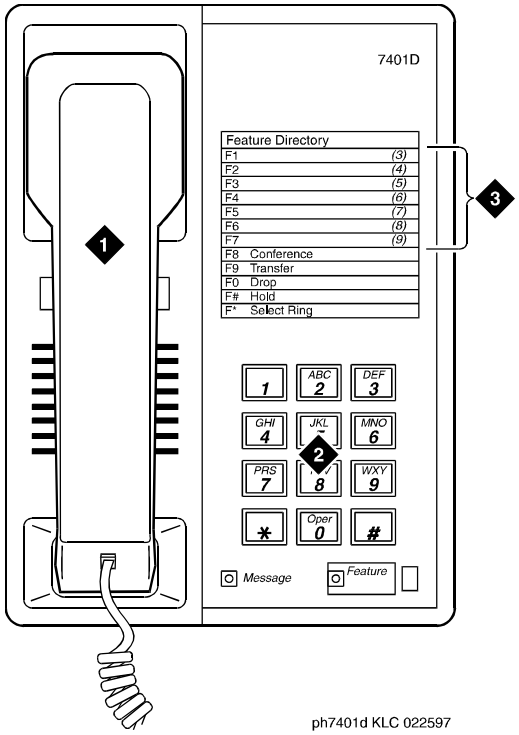


Figure notes:

1. **Handset**
2. **Dial pad**
3. **Access codes card**

Figure 230: 7404D telephone

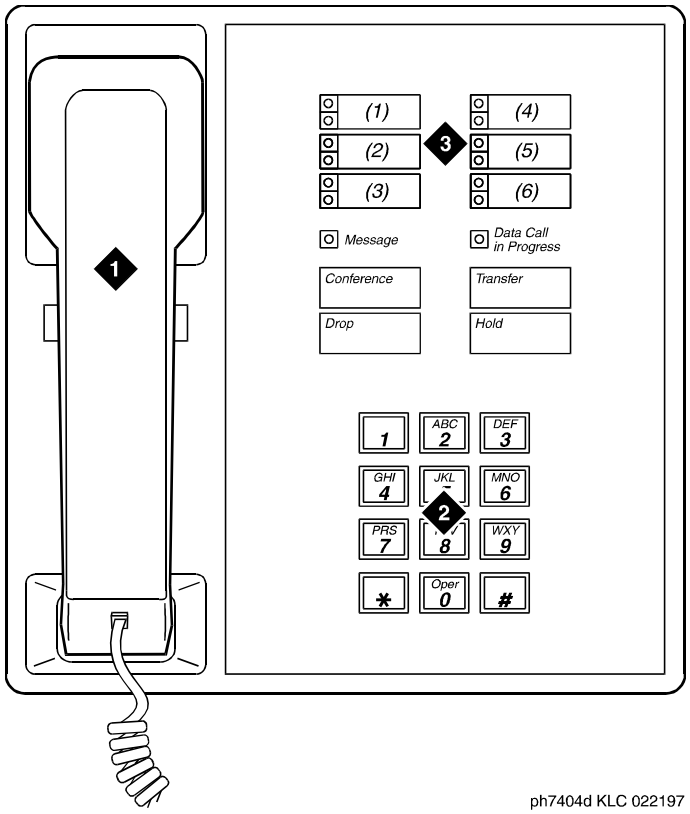


Figure notes:

1. **Handset**
2. **Dial pad**
3. **6 programmable buttons**

Figure 231: 7405D telephone

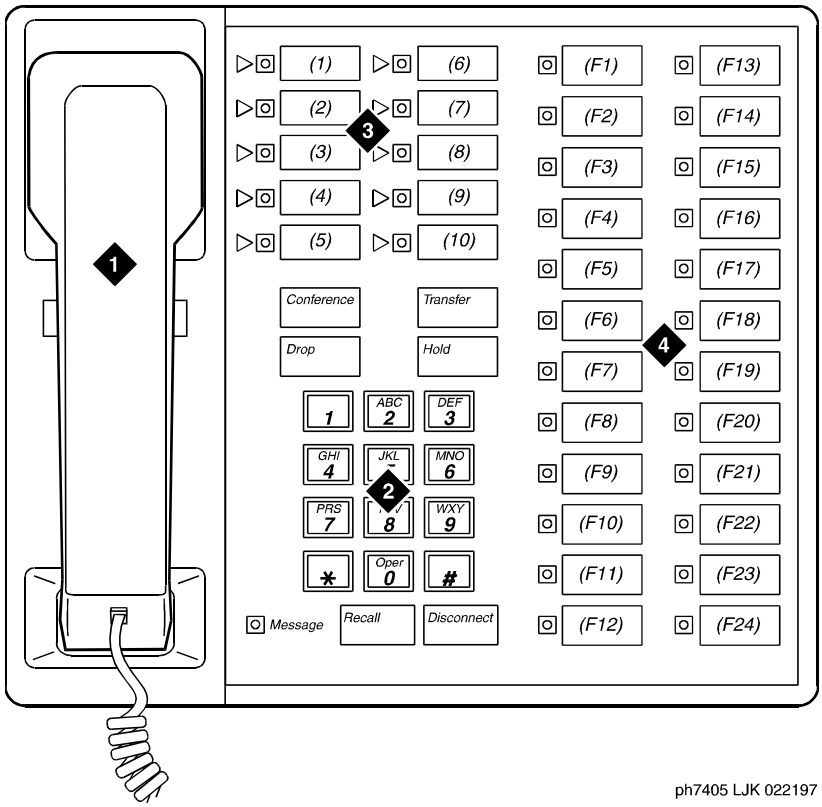


Figure notes:

1. **Handset**
2. **Dial pad**
3. **10 programmable buttons**
4. **24 feature buttons**

Figure 232: 7405D telephone with optional function key module

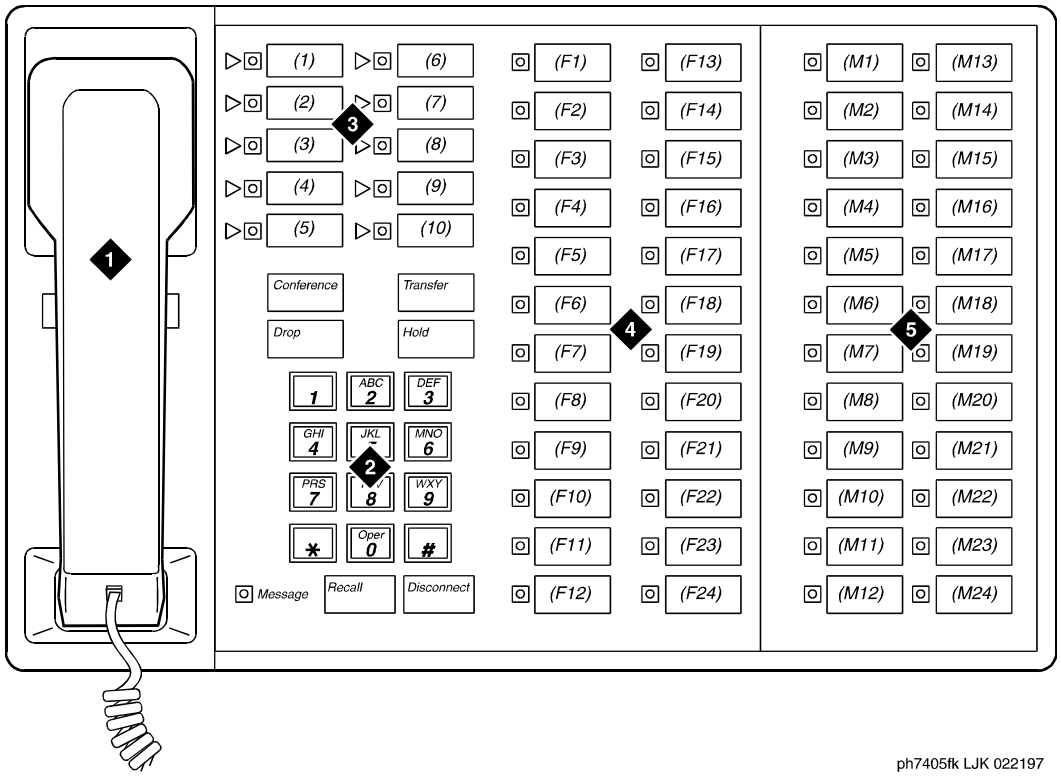


Figure notes:

1. **Handset**
2. **Dial pad**
3. **10 programmable buttons**
4. **24 feature buttons**
5. **Function key module with 24 feature module buttons**

Figure 233: 7405D telephone with optional digital display module

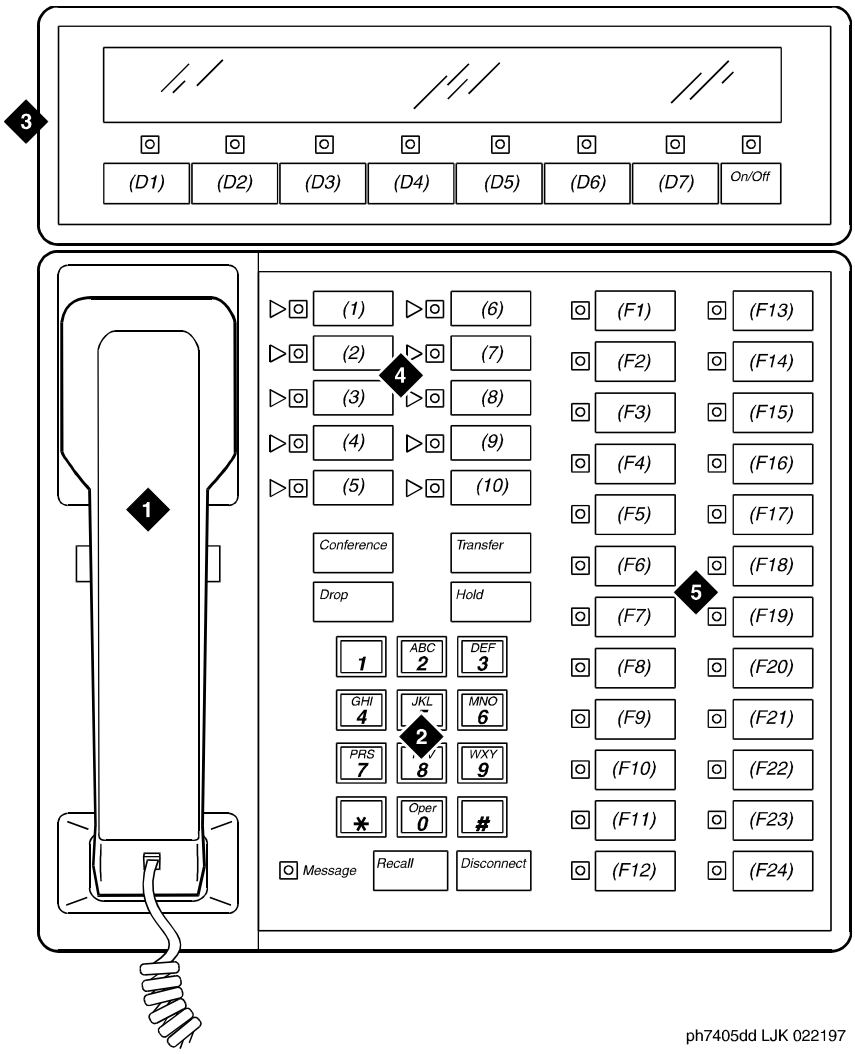


Figure notes:

1. **Handset**
2. **Dial pad**
3. **Digital display module with 7 display buttons**
4. **10 programmable buttons**
5. **24 feature buttons**

Figure 234: 7405D telephone with optional call coverage module

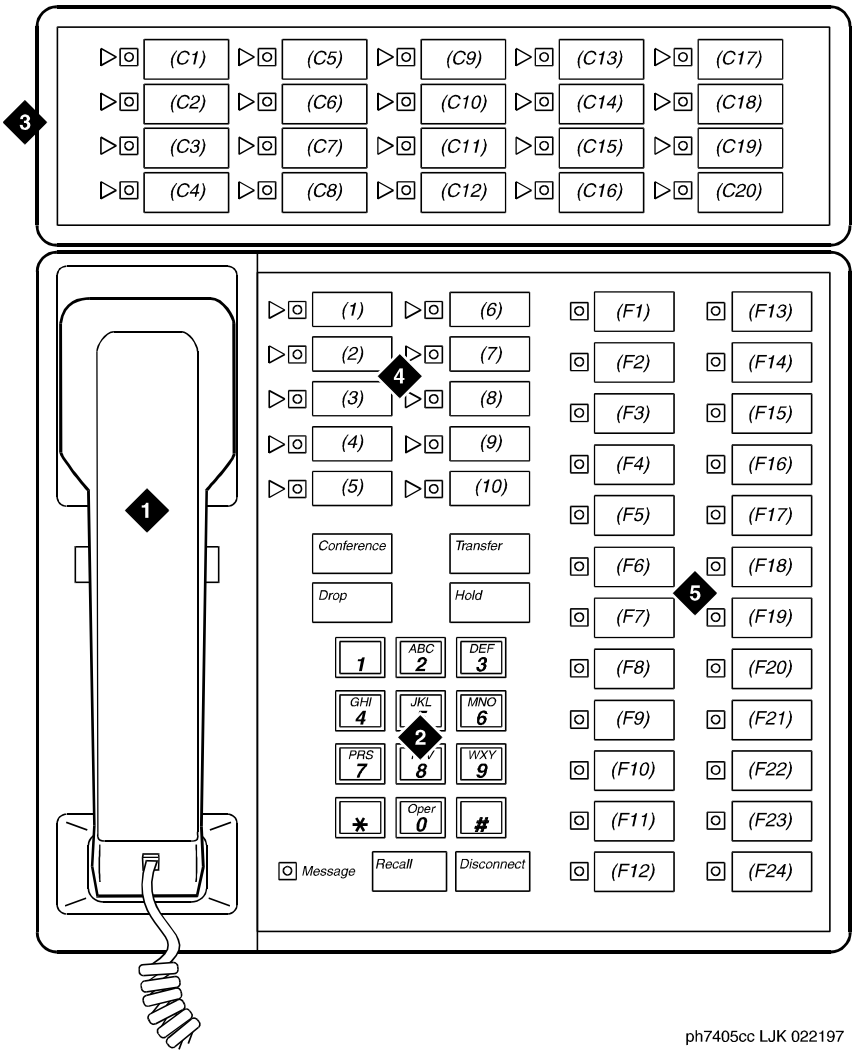


Figure notes:

1. **Handset**
2. **Dial pad**
3. **Call coverage module with 20 coverage module buttons and status lamps**
4. **10 programmable buttons**
5. **24 feature buttons**

Figure 235: 7405D telephone with optional digital terminal data module

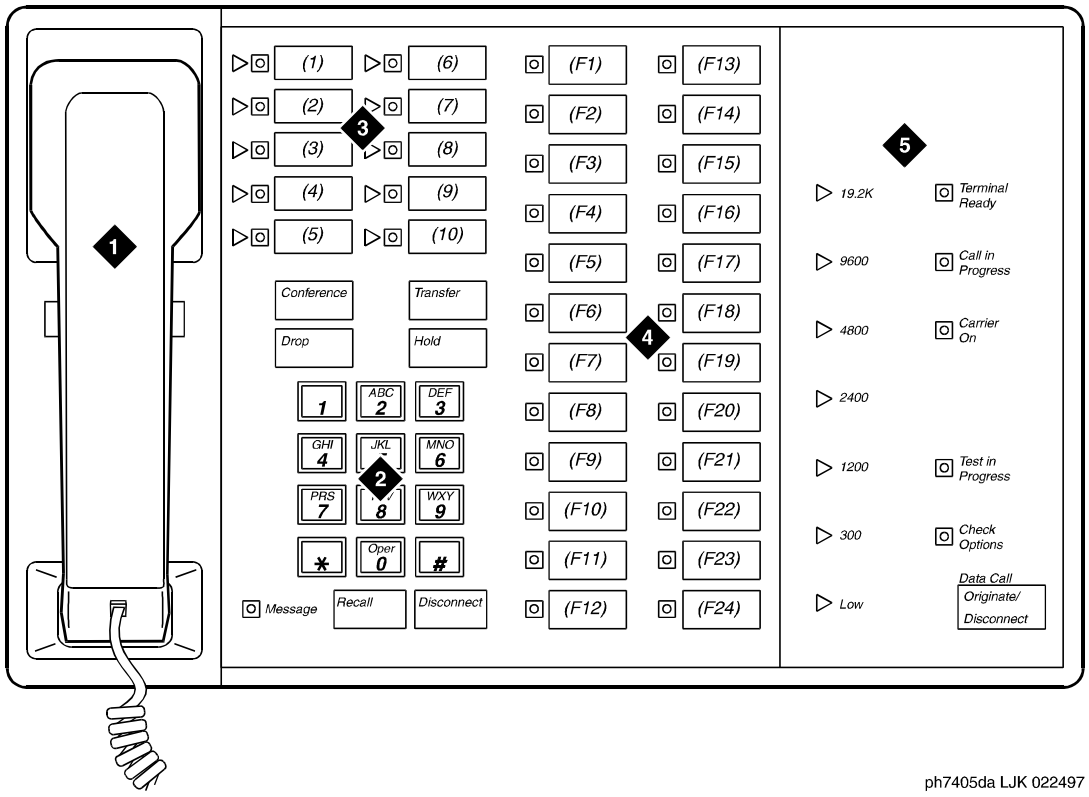


Figure notes:

1. **Handset**
2. **Dial pad**
3. **10 programmable buttons**
4. **24 feature buttons**
5. **Digital terminal data module**

Figure 236: 7406D+ telephone

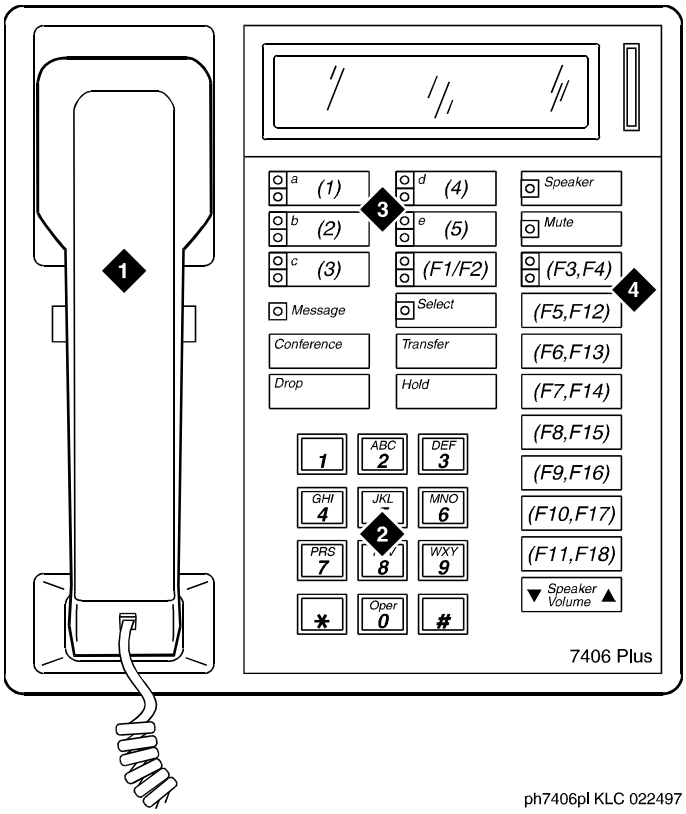


Figure notes:

1. **Handset**
2. **Dial pad**
3. **5 programmable buttons**
4. **18 feature buttons (feature buttons F2, F4, and F12 to F18 are enabled with the Shift key)**

Figure 237: 7406D telephone

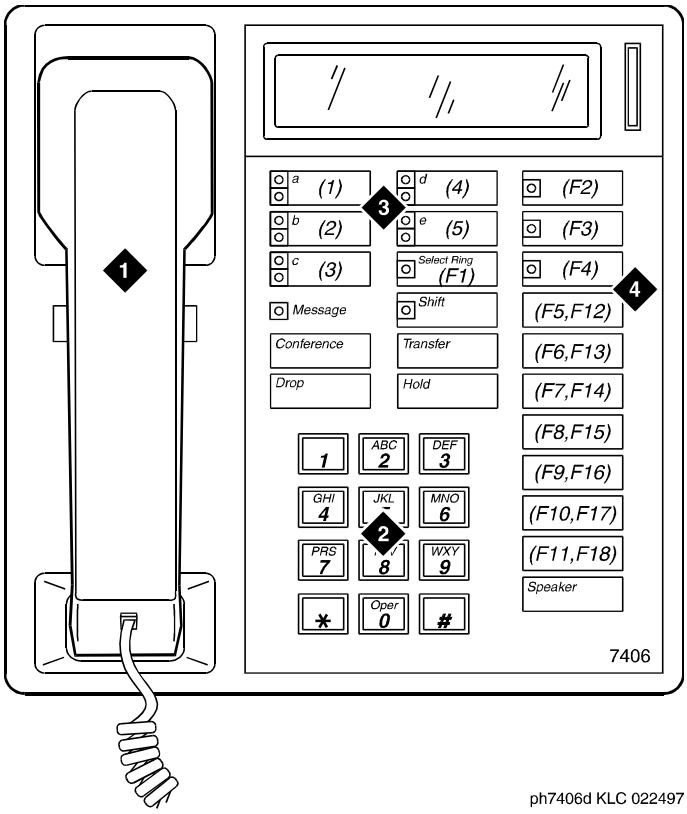


Figure notes:

1. **Handset**
2. **Dial pad**
3. **5 programmable buttons**
4. **18 feature buttons (feature buttons F12 to F18 are enabled with the Shift key)**

Figure 238: 7407D telephone

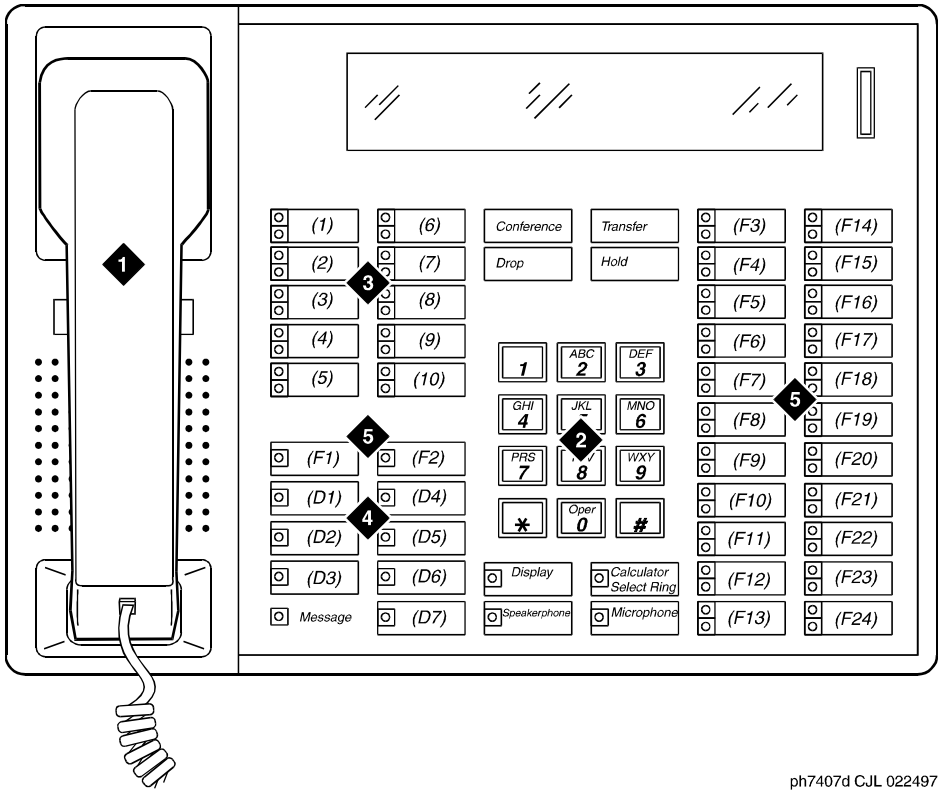


Figure notes:

1. **Handset**
2. **Dial pad**
3. **10 programmable buttons**
4. **7 display buttons**
5. **24 feature buttons**

Figure 239: 7434D telephone

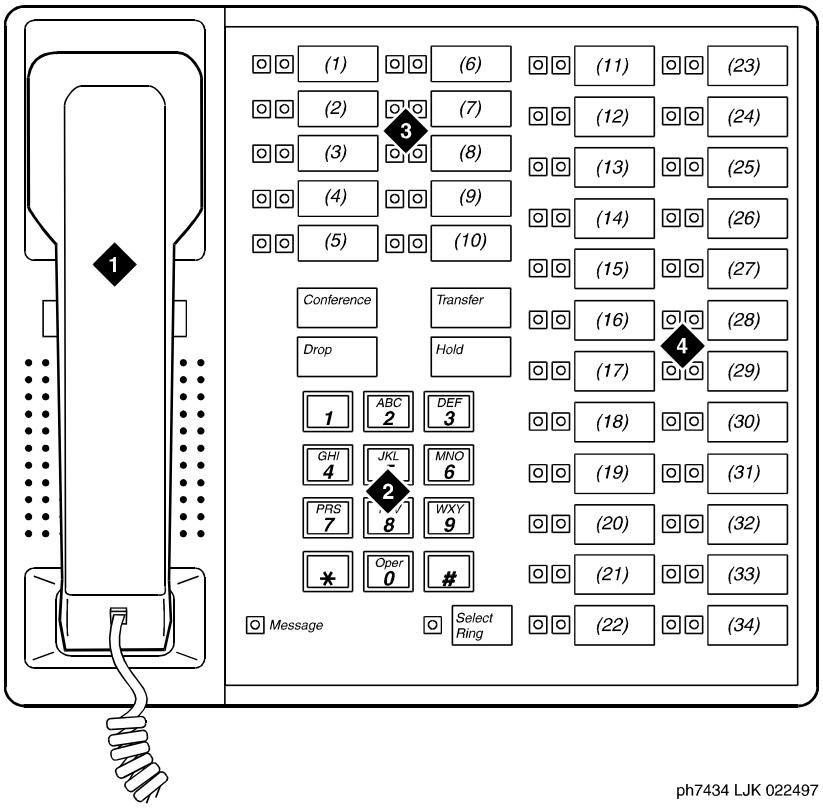


Figure notes:

1. **Handset**
2. **Dial pad**
3. **10 programmable buttons**
4. **24 feature buttons (11 to 34)**

Figure 240: 7434D telephone with optional call coverage module

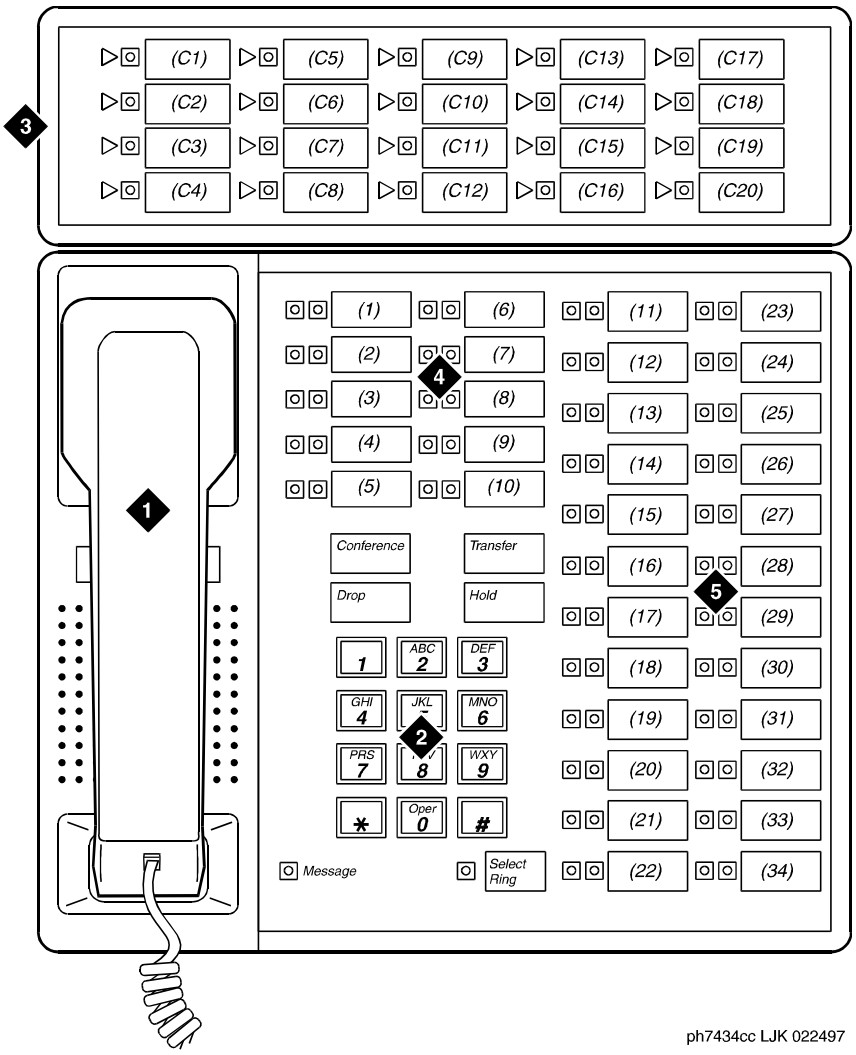


Figure notes:

1. **Handset**
2. **Dial pad**
3. **Call coverage module with 20 coverage module buttons and status lamps**
4. **10 programmable buttons**
5. **24 feature buttons (11 to 34)**

Figure 241: 7434D telephone with optional digital display module

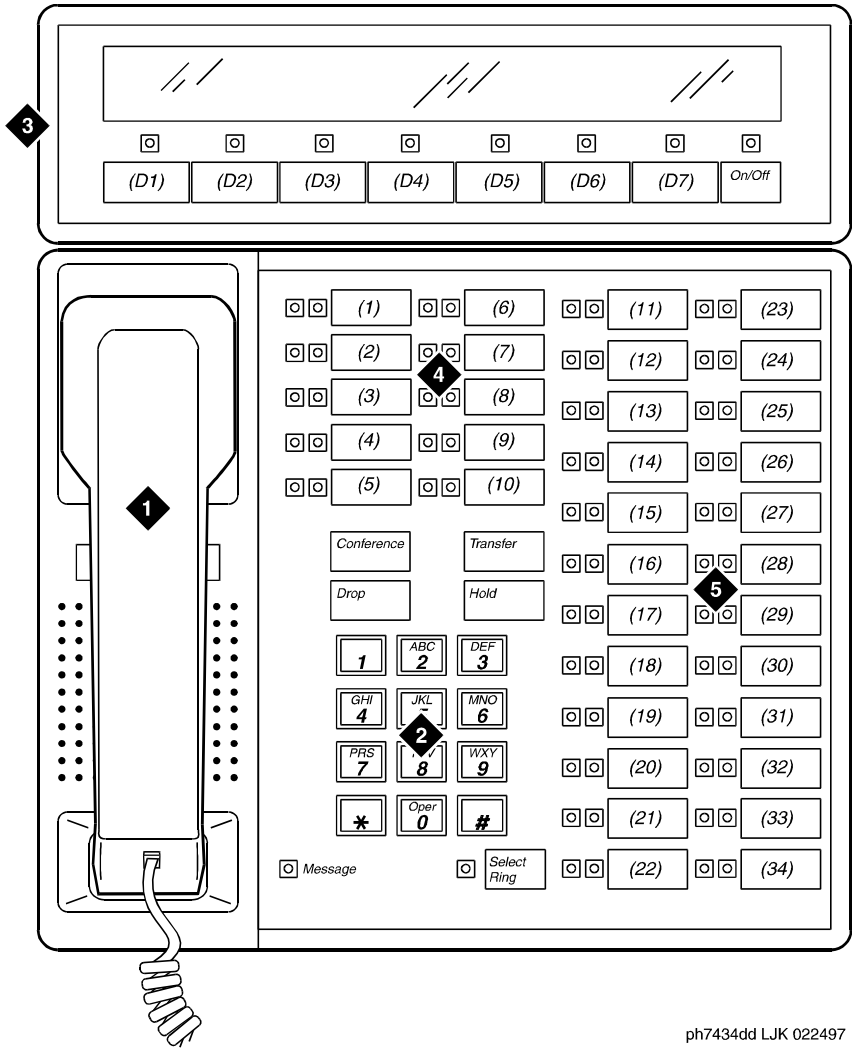


Figure notes:

1. **Handset**
2. **Dial pad**
3. **Digital display module with 7 display buttons**
4. **10 programmable buttons**
5. **24 feature buttons (11 to 34)**

ISDN telephones (7500s & 8500s)

The Integrated Services Digital Network (ISDN) phones include both the 7500-series and the 8500-series telephones. Each of these phones uses the ISDN communications through a 4-wire "T"-interface.

National ISDN (NI-1 and NI-2) BRI telephones

Phones that comply with National ISDN (NI-1 and NI-2) BRI standards may serve as phones on a Communication Manager system if they are programmed in Call Appearance Call Handling - Electronic Key Telephone Service (CACH-EKTS) mode. Examples of NI-BRI phones that are supported are:

- Nortel 5317T
- L3 STE
- 6210 and 6220 Tone Commander

NI-BRI phones with Communication Manager offer multiple call appearances, the **Conference**, **Transfer**, **Hold**, and **Drop** feature buttons, and a Message Waiting Indicator. NI-BRI telephone users must access all other features of Communication Manager using feature access codes. Additional feature buttons on an NI-BRI telephone may be assigned only as call appearances, and the number of call appearance buttons administered in Communication Manager must match the number of call appearances administered on the telephone. Bridged call appearances are not supported.

NI-BRI phones also have access to all features of the Multi-level Precedence and Preemption (MLPP) capability if MLPP has been enabled.

7505D ISDN-BRI telephone

The 7505D telephone serves as a telephone if it is equipped with a Voice Only Module (VOM). As a telephone, it offers programmable buttons, fixed feature buttons, a Message light, touch-tone dialing, and a built-in, programmable speakerphone or SPOKESMAN loudspeaker.

The 7505 can be equipped with an optional Asynchronous Data Module (ADM) that provides the user with simultaneous voice and data capabilities. The 7505 equipped with the ADM offers the same voice capabilities as the 7505 equipped with the VOM, plus it allows you to attach data terminals or personal computers to send and receive data through the digital network.

7506D ISDN-BRI telephone

The 7506D serves as a telephone if it is equipped with a VOM. As a telephone, it offers programmable buttons, fixed feature buttons, a Message light, touch-tone dialing, and a built-in, programmable speakerphone or SPOKESMAN loudspeaker.

The 7506 can be equipped with an optional Asynchronous Data Module that provides the user with simultaneous voice and data capabilities. The 7506 equipped with the ADM offers the same voice capabilities as the 7506 equipped with the VOM, plus it allows you to attach data terminals or personal computers to send and receive asynchronous data through the digital network.

7507D ISDN-BRI telephone

The 7507D serves as a telephone if it is equipped with a VOM. As a telephone it offers programmable buttons, fixed feature buttons, Message light, touch-tone dialing, and a built-in, programmable speakerphone or SPOKESMAN loudspeaker.

The 7507 can be equipped with an optional Asynchronous Data Module that provides the user with simultaneous voice and data capabilities. The 7507 equipped with the ADM offers the same voice capabilities as the 7507 equipped with the VOM, plus it allows you to attach data terminals or personal computers to send and receive data through the digital network.

8503D ISDN-BRI telephones

The 8503T telephone offers: four standard fixed feature buttons, a Message light, three call appearance/flexible feature buttons, each with a red and green light, 12 programmable memory-dialing locations on the dial pad keys, a PROGRAM button for storing numbers at the memory-dialing locations and a MEMORY button for dialing these programmed numbers or codes, the Redial feature, the Mute feature.

8510T ISDN-BRI telephone

The 8510T voice/data telephone offers: 10 call appearance/feature buttons, each with a red and green status light, buttons for the Mute, Redial, Conference, Drop, Transfer, and Hold features (the MUTE, SPEAKER, CONF, and TRANSFER buttons have a red light next to them), a Speakerphone, a Volume control, and an Adjunct jack. You can administer the softkey buttons. Four softkeys and display control buttons below the a 2-line by 24-character display can be used to access such features as a personal Directory, a Call Log, the Self-Test feature, and a personalized ringing pattern for the telephone.

8520T ISDN-BRI telephones

The 8520T voice/data telephone offers the following: 20 call appearance/feature buttons, each with a red and green status light, buttons for the Mute, Redial, Conference, Drop, Transfer, and Hold features (the MUTE, SPEAKER, CONF, and TRANSFER buttons have a red light next to them), a Speakerphone, a Volume control, and an Adjunct jack. Ten softkeys and four display control buttons located on either side of the 7-line by 24-character display can be used to access such features as a personal Directory, a Call Log, the Self-Test feature, and a personalized ringing pattern for the telephone.

Figure 242: 8503D telephone

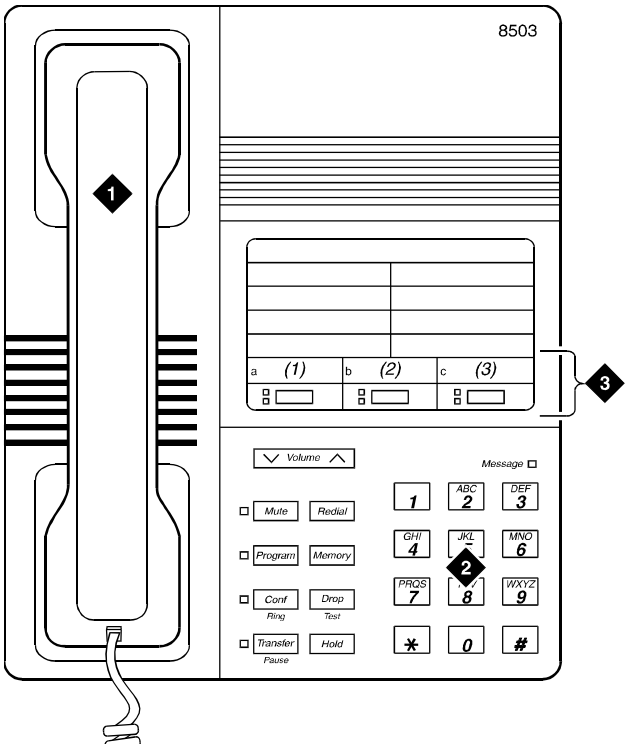


Figure notes:

1. **Handset**
2. **Dial pad**
3. **3 programmable buttons**

Figure 243: 8510T telephone

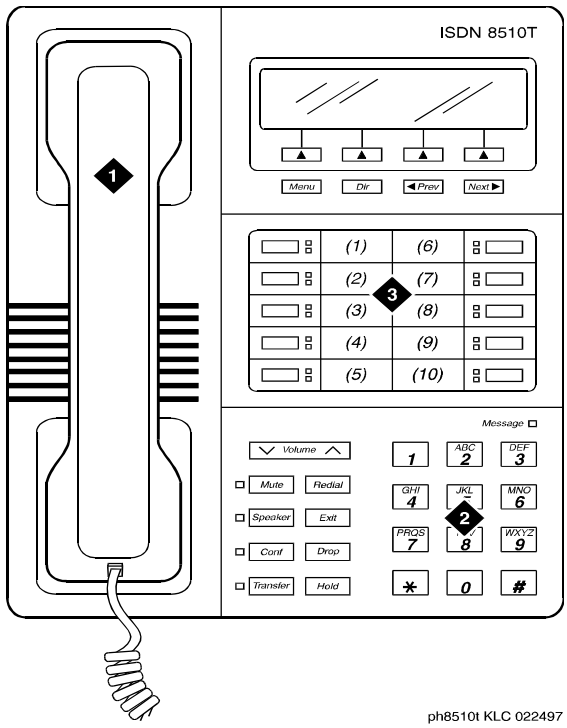


Figure notes:

1. **Handset**
2. **Dial pad**
3. **10 programmable buttons**

Figure 244: 8520T telephone

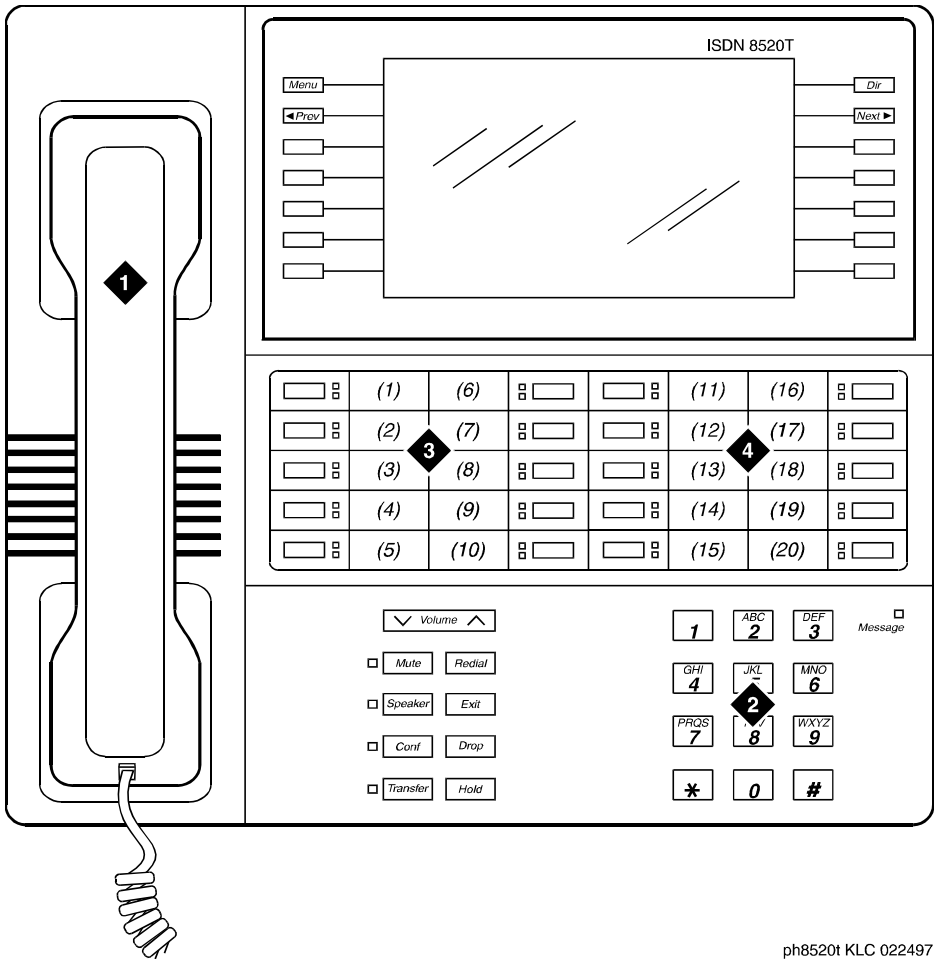


Figure notes:

1. **Handset**
2. **Dial pad**
3. **10 programmable buttons**
4. **10 programmable buttons (11 to 20)**

Note:

The 8520T telephone supports 20 call appearances. The system maximum of 10 call appearance buttons still applies. You can administer the buttons that are not used as call appearance buttons as bridged appearances.

8110 telephones

The basic 8110 (8110A01A, 8110A01B, and 811A01C) and the modified 8110M (8110A01D) telephones are single-line analog telephones. These telephones are exactly the same in appearance: each contains 12 programmable dialing buttons with PROGRAM and PAUSE buttons, automatic redial, a flashing red Message light, and a Hold button. They also have built-in speakerphones with Mute capability and the Automatic Answer feature.

8400-series telephones

8403B telephones

The 8403 telephone is a multi-appearance digital telephone with three call appearance buttons, Conference, Transfer, Drop, and Hold buttons, a TEST button, a blue FEATURE button which allows you to access 12 system features assigned by the System Manager and to choose from among eight different ringing patterns, a MUTE button, a SPEAKER button which accesses a 1-way, listen-only speaker, a red Message light, and a Volume control button.

The 8403 can be used in either a 4-wire or 2-wire environment.

8405B telephone

There are four varieties of the 8405 telephone: the 8405B and 8405B+, the 8405D and 8405D+. All four varieties are multi-appearance digital telephones with five call appearance/feature buttons.The 8405 telephones also have four standard fixed feature buttons (CONFERENCE, DROP, HOLD, and TRANSFER), a MUTE button, a SPEAKER button, a TEST button, and a Volume control button. The 8405D and 8405D+ allow you to administer 12 softkey feature buttons in addition to the call appearance and feature buttons.

The four 8405 variations have the following differences:

- The 8405B has a 1-way, listen-only speaker, with NO display.
- The 8405B+ has a 2-way speakerphone, without a display
- The 8405D has a 1-way, listen-only speaker and a 2-line by 24-character display.
- The 8405D+ has a 2-way speakerphone and a 2-line by 24-character display.

The 8405 telephones work in 4-wire or 2-wire environments.

8410B telephone

The 8410 telephone is a multi-appearance digital telephone with 10 call appearance/feature buttons, four standard fixed feature buttons (CONFERENCE, DROP, HOLD, and TRANSFER), a MUTE button, a SPEAKER button which can access either a 2-way speakerphone or a 1-way, listen-only speaker, a TEST button, and a Volume control button.

- The 8410B is the basic set, without a display.
- The 8410D (8410D03A) has a built-in 2-line by 24-character display. Those users who have an 8410D with display can access 12 features with the softkeys and display control buttons. These 12 features can be used *in addition to* the features on the call appearance/ feature buttons.

The 8410 telephone can work in both 4-wire and 2-wire environments.

8411B and 8411D telephones

The 8411 telephone is a multi-appearance digital telephone with 10 call appearance/feature buttons, four standard fixed feature buttons (CONFERENCE, DROP, HOLD, and TRANSFER), a blue SHIFT button, a MUTE button, a SPEAKER button which can access either a 2-way speakerphone or a 1-way, listen-only speaker, a TEST button, and a Volume control button.

The rear of the 8411 telephone has two jacks: The Analog Adjunct jack can be used for connecting answering machines, fax machines, PC or laptop data/fax modem cards, data sets or modems, audio teleconferencing equipment, and TTY machines commonly used by the hearing impaired. The RS-232-D Jack can be used for connecting the telephone to a COM port on an IBM®-compatible personal computer on which you can load PassageWay Solution software.

There are two varieties of the 8411 telephone: the 8411B (8411D02A) is the basic set, without a display; the 8411D (8411D01A) has a built-in 2-line by 24-character display. Those users who have an 8411D with display can access 12 features with the softkeys and display control buttons. These 12 features can be used in addition to the features on the call appearance/ feature buttons.

The 8411 telephone can work in both 4-wire and 2-wire environments.

8434D telephone

The basic 8434 (8434D01A) and the enhanced 8434DX (8434D02A) telephones are multi-appearance digital telephones which offer 34 call appearance/feature buttons, each with a red light and a green status light, four standard fixed feature buttons (CONFERENCE, DROP, HOLD, and TRANSFER), a MUTE button, a SPEAKER button which accesses either a 2-way speakerphone or a 1-way listen-only speaker, a TEST button, a SHIFT button (some 8434DX telephones will have a RING button instead), a red Message light, personalized ringing, a built-in speakerphone with a reset option, and a built-in 2-line by 40-character VFD display. The 8434 and 8434DX also have five softkeys and four display control buttons which allow the user to access 15 features. These softkey features can be used *in addition to* the features on the call appearance/feature buttons.

The 8434 and 8434DX telephones can be used in both a 4-wire and a 2-wire environment.

Note:

In order to use the display on the 8434 or 8434DX telephone and to use an 801A expansion module connected to the 8434DX, you must connect an auxiliary power supply to the telephone.

You can connect an 801A Expansion Module to the 8434DX telephone to provide 24 additional call appearance/feature buttons.

Figure 245: 8403B telephone

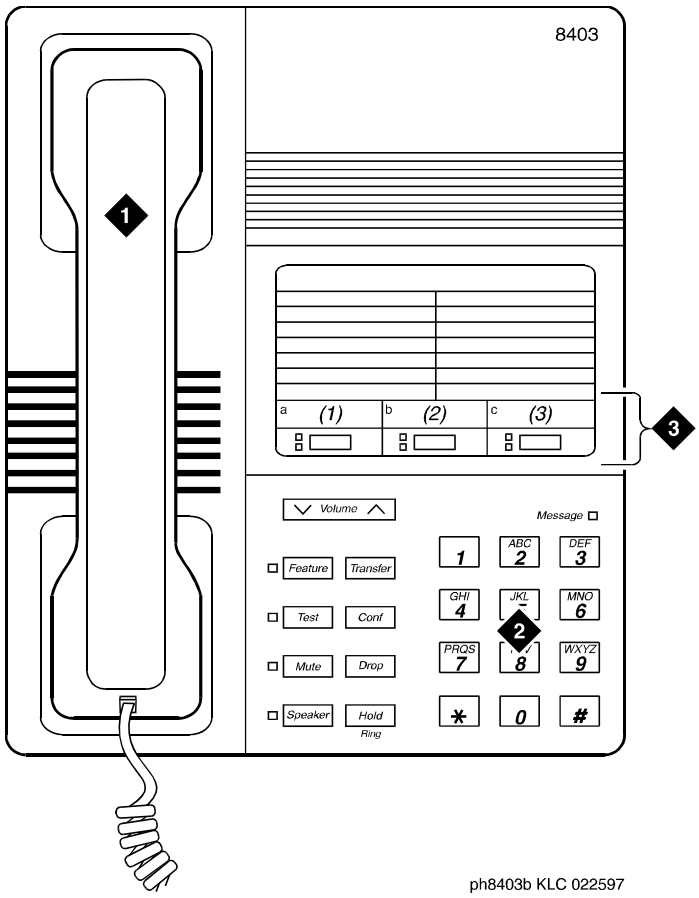


Figure notes:

1. **Handset**
2. **Dial pad**
3. **3 programmable buttons**

Figure 246: 8410B telephone

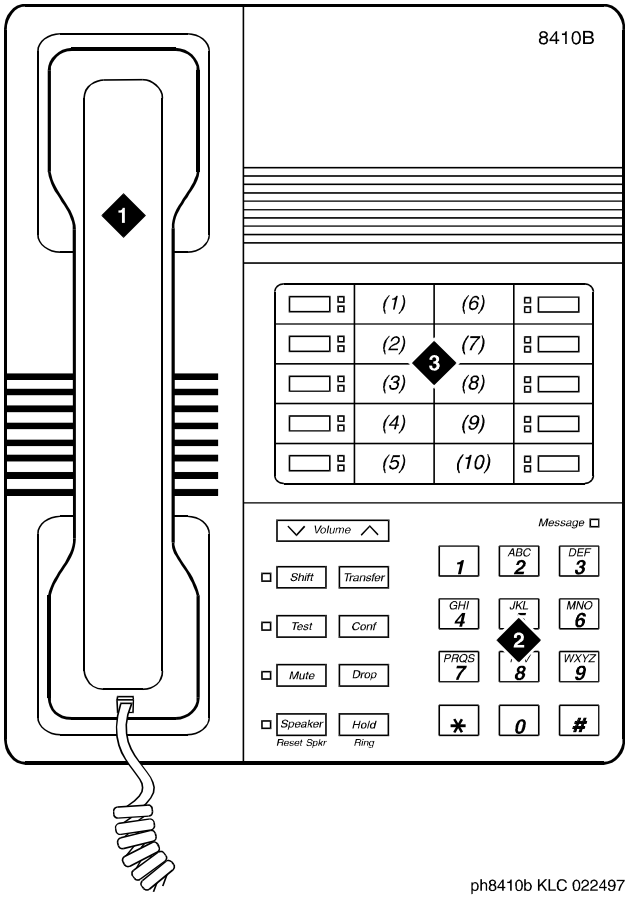


Figure notes:

1. **Handset**
2. **Dial pad**
3. **10 programmable buttons**

Note:

The 8405B and 8405B+ look like the 8410B with the exception that the 8405 series do not have the second column of line appearances.

Figure 247: 8410D telephone

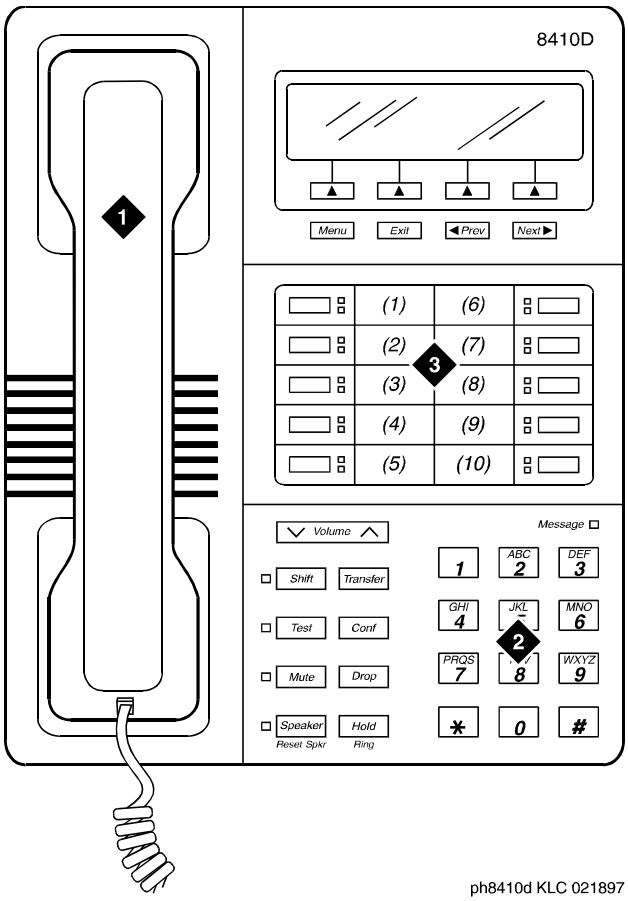


Figure notes:

1. **Handset**
2. **Dial pad**
3. **10 programmable buttons**

Note:

The 8405D and 8405D+ look like the 8410D with the exception that the 8405 series do not have the second column of line appearances.

Figure 248: 8434D telephone

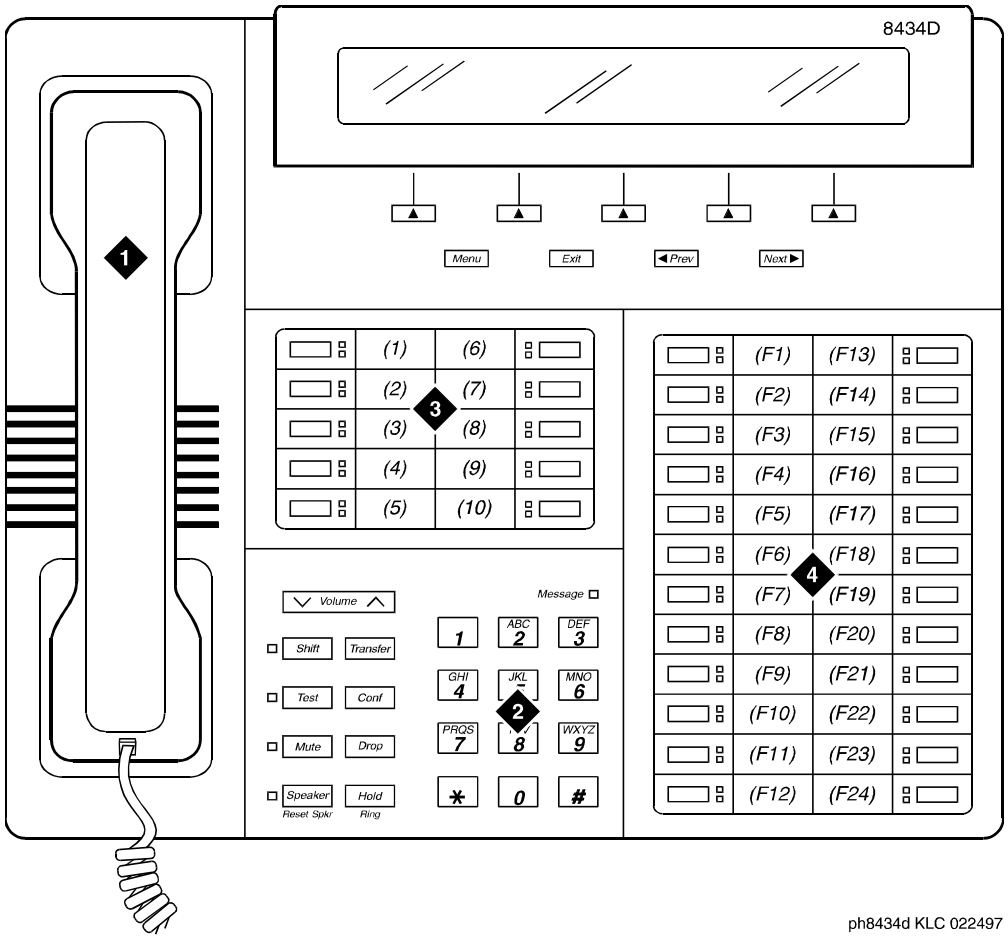


Figure notes:

1. **Handset**
2. **Dial pad**
3. **10 programmable buttons**
4. **24 feature buttons**

CALLMASTER telephones

There are several types of CALLMASTER telephones:

- 602A and 602D CALLMASTER

  The 602 CALLMASTER models have a display, a Message light, a Mute button, and four fixed feature buttons: Conference, Drop, Hold, and Transfer. You can administer its 10 call appearance/feature (2-lamp) buttons and its 17 feature-only (1-lamp) buttons.

- 603D (CALLMASTER II)

  The CALLMASTER II model has a display, a Message light, and the Mute, Select, Log In, and Release buttons. It also has four fixed features: Conference, Drop, Hold, and Transfer. You can administer its 6 call appearance/feature (2-lamp) buttons and its 15 feature-only (1-lamp) buttons.

- 603E (CALLMASTER III)

  The CALLMASTER III model has a display, a Message light, and the Select, Mute, Log In, and Release buttons. It also has four fixed features: Conference, Drop, Hold, and Transfer. You can administer its 6 call appearance/feature (2-lamp) buttons and its 15 feature-only (1-lamp) buttons. Note that you can assign any feature to the Log In and Release buttons.

  You can connect the CALLMASTER III to either a standard 4-wire DCP or a 2-wire circuit pack.

- 603F (CALLMASTER IV)

  The CALLMASTER IV model has a display, a Message light, and the Select, Mute, Log In, and Release buttons. It also has four fixed features: Conference, Drop, Hold, and Transfer. You can administer its 6 call appearance/feature (2-lamp) buttons and its 15 feature-only (1-lamp) buttons. Note that you can assign any feature to the Log In and Release buttons.

  You can connect the CALLMASTER IV to either a standard 4-wire DCP or a 2-wire circuit pack.

- 606A (CALLMASTER VI)

  The CALLMASTER VI model is a miniature, 8-button, 2-headset jack, digital telephone that is controlled by the user’s personal computer (PC) through an RS-232 serial-port connection.

- 607A (CALLMASTER V)

  The CALLMASTER V model has a display, softkeys, and the display control buttons (Menu, Exit, Previous, and Next). This model does not have a standard handset, but you can connect a handset to one of its headset jacks. The CALLMASTER V has six fixed feature buttons: Speaker, Mute, Hold, Redial, Conference, and Transfer. You can administer its 16 call appearance/feature (2-lamp) buttons, however, one of these buttons must be administered as a Headset On/Off button and a second one must be administered as a Release button. You can also administer the 12 softkey buttons.

Figure 249: CALLMASTER II/CALLMASTER III digital telephone

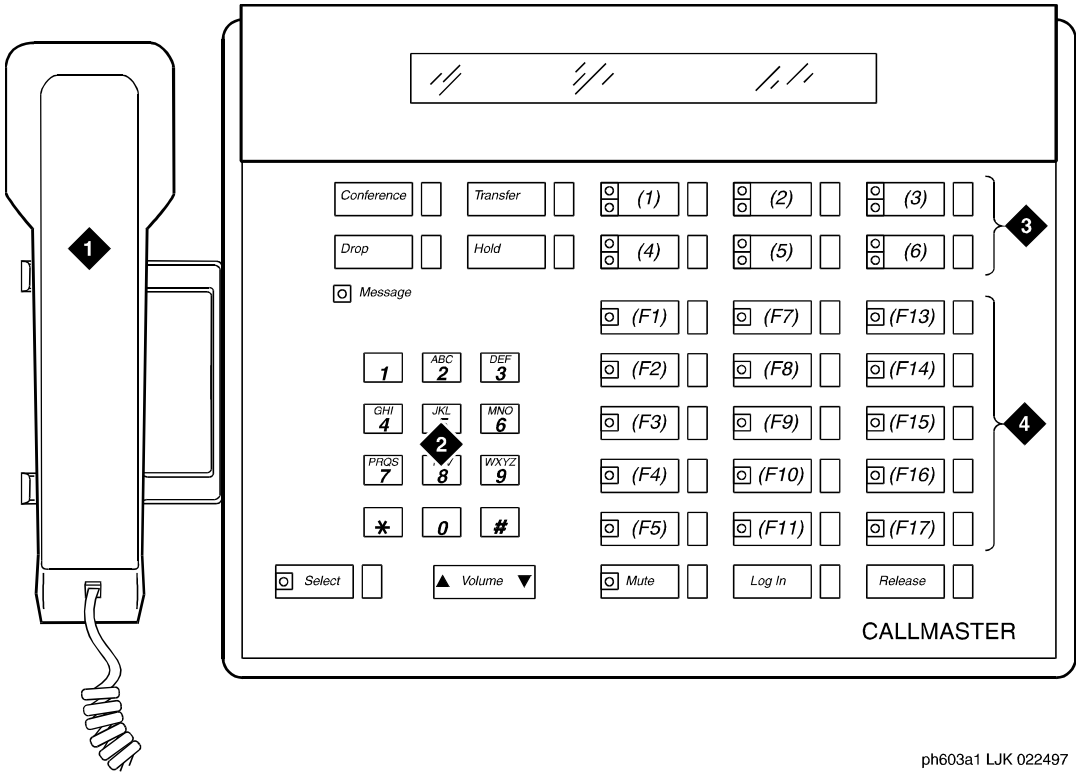


Figure notes:

1. **Handset**
2. **Dial pad**
3. **6 programmable buttons**
4. **18 feature buttons**

Figure 250: CALLMASTER IV digital telephone

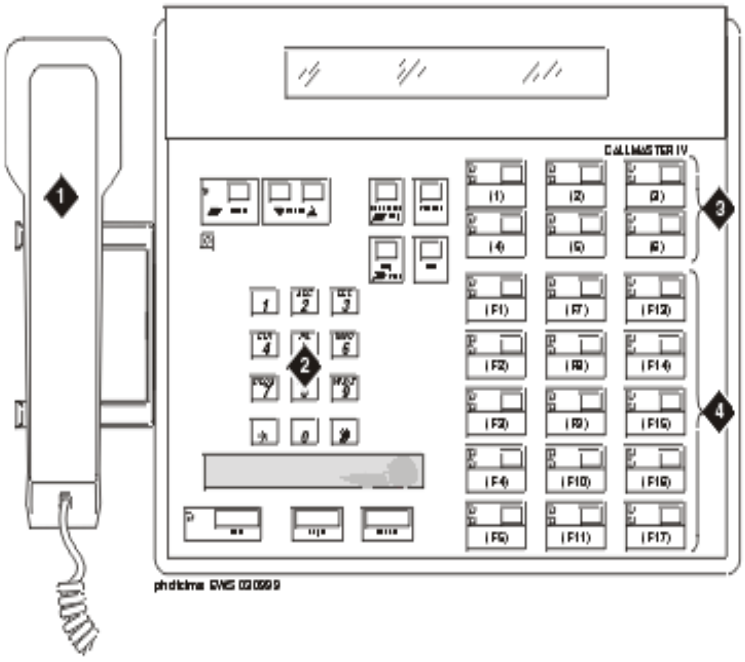


Figure notes:

1. **Handset**
2. **Dial pad**
3. **6 programmable buttons**
4. **15 feature buttons**

Figure 251: CALLMASTER V digital telephone

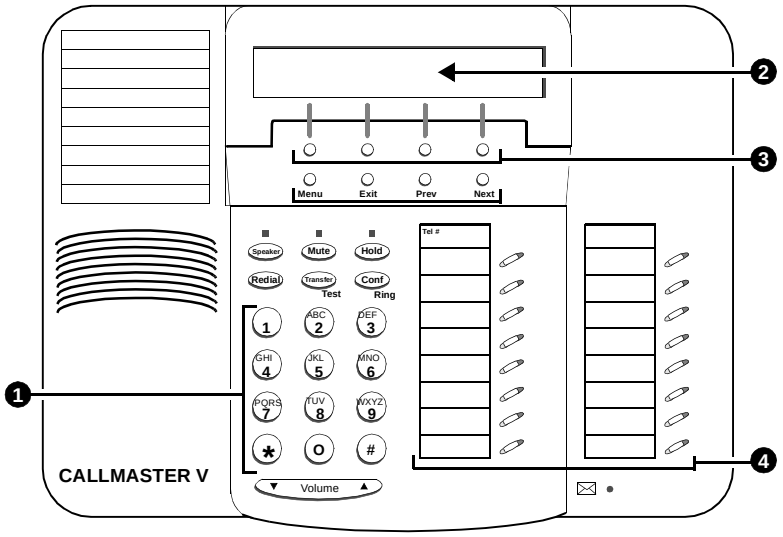


Figure notes:

1. **Dial pad**
2. **Display**
3. **4 softkey buttons**
4. **16 call appearance/feature buttons**

Cordless telephone

MDC9000 cordless telephone

The MDC 9000 Cordless Telephone has two basic parts, the handset and the charging base.

- The handset has line/programmable feature/intercom buttons, Conference, Drop, Hold, Transfer, Status, and Feature buttons, Headset On/Off and Handset On/Off buttons, a LCD display, an earpiece volume control switch, battery charging contacts, a directory card, and a headset jack.
- The charging base has a handset hook, ringer volume controls, battery charging contacts, a Talk indicator, a Charge indicator, a Message indicator, and a base ringer.

MDW9000 cordless telephone

The MDW 9000 Wireless Telephone is part of the TransTalk 9000 Digital Wireless System family of telephones. This wireless telephone has three basic parts, the handset the charging cradle, and the radio module.

- The handset has line/programmable feature/intercom buttons, Drop, Mute, Hold, Conference, and Transfer buttons, a Headset On/Off button and a Handset On/Off button, a LCD display, a Volume control switch, battery charging contacts, a flexible antenna, and a Headset jack.
- The charging cradle has a handset hook, a spare battery cover, a spare battery.
- The radio module has Power, Pass, and Radio indicator lights, a top hook, card edge, and snap lock which connect the radio module to the carrier assembly/backplane, an antenna, and power plug and line connectors.

Figure 252: MDC9000 and MDW9000 cordless telephones

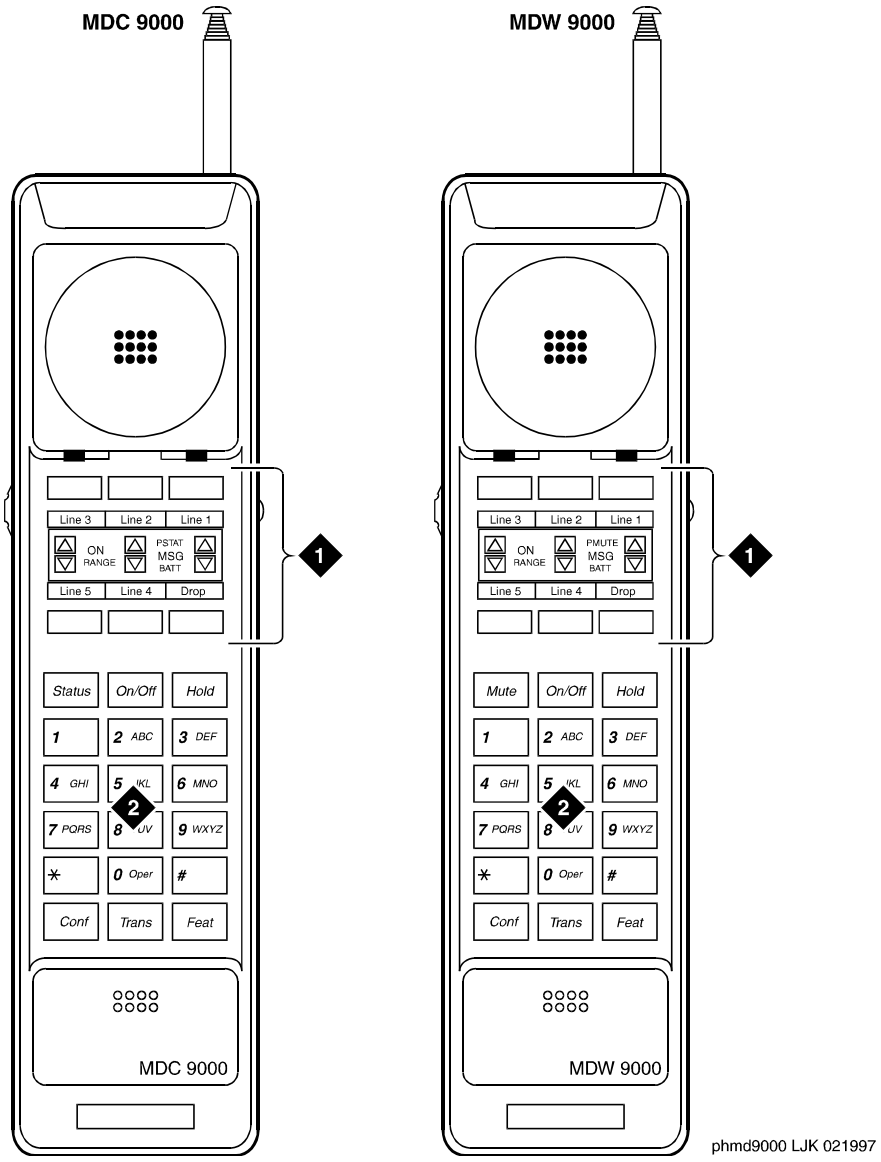


Figure notes:

1. **6 programmable buttons**
2. **Dial pad**
3. **6 programmable buttons**
4. **Dial pad**

Figure 253: MDC9000 and MDW9000 cordless telephones

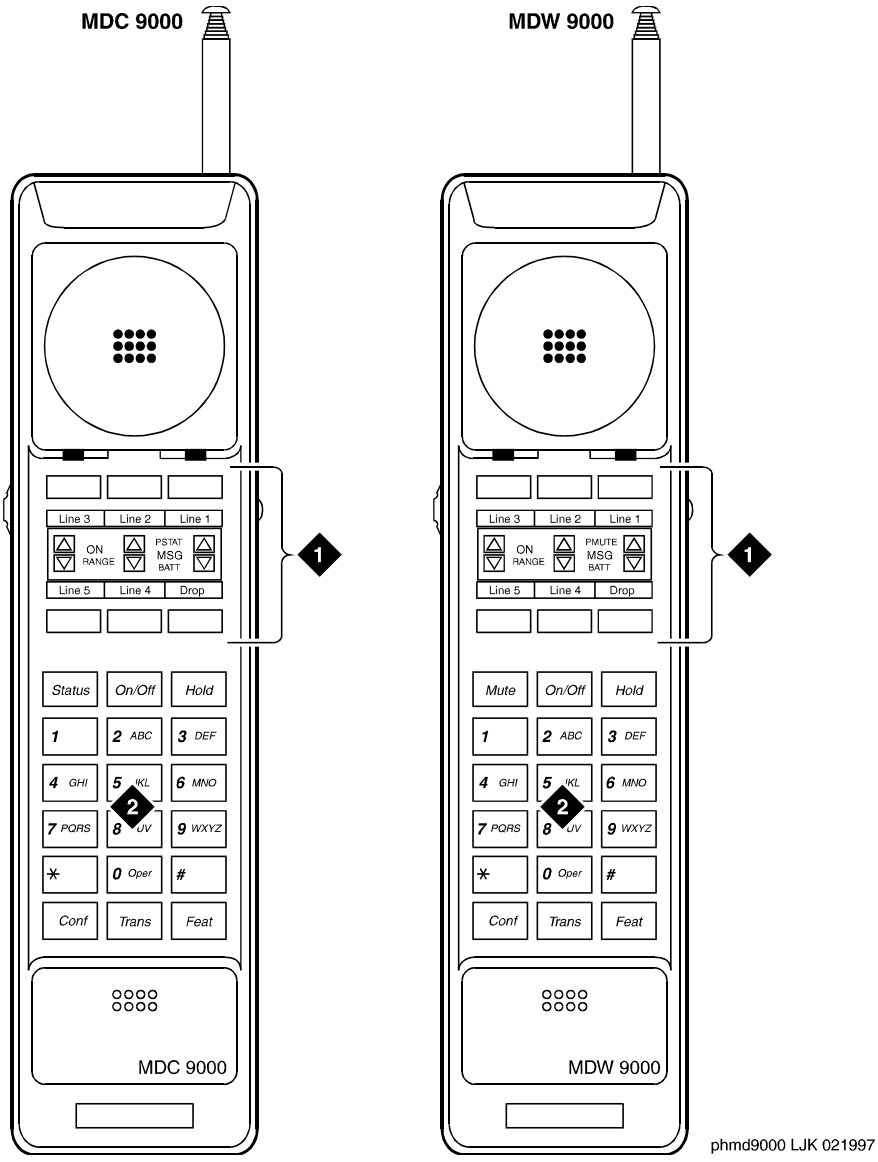


Figure notes:

1. 6 programmable buttons
2. Dial pad
3. 6 programmable buttons
4. Dial pad

Internet Protocol (IP) Softphones

IP Softphones can turn a PC or a laptop into an advanced telephone. Using the Internet Protocol (IP) network, you can place calls, take calls, and handle calls on your PC. With certain exceptions, every feature available for wired-endpoint voice calling is available for IP-based calling; it supports internetworking with conventional circuit-switched stations and trunks.

There are three softphone models that are fixed at the time of purchase and do not change afterward:

- Avaya IP Softphone (for additional information, go to http://www.avaya.com/support, click on **Telephones and End User Devices**, then scroll down to **IP Telephony**)
- Avaya IP Agent (for additional information, go to http://www.avaya.com/support, click on **Call Center/CRM**, then scroll down to **IP Agent**)
- Avaya Softconsole (for additional information, go to http://www.avaya.com/support, click on **Call Center/CRM**, then scroll down to **IP Telephony**)

The connection for the end-point is supported whether the customer is directly connected using a network inter-face card (NIC) or the customer has dialed into their network using a dial-up point-to-point (PPP) account. The connection is managed via a registration session with the Avaya server. The call server manages the voice connection directly to a circuit-switched telephone (Telecommuter configuration), directly to an IP Telephone (Control of IP Telephone configuration), directly to a DCP digital telephone (Control of DCP Telephone configuration), or over UDP/RTP to the IP Softphone's iClarity IP Audio H.323 audio end-point (Road Warrior or VoIP configuration).

At any time, an end user can put a softphone into one of the following modes by changing a software menu:

- Road-Warrior (Voice over IP (VoIP))
- Telecommuter
- Native H.323 (only available with Avaya IP Softphone R2)
- Control of IP Telephone (only available with IP Softphone R4 and later)
- Control of DCP Telephone (only available with IP Softphone R5 and later)

Note:

Beginning with the November 2003 release of Communication Manager, R1 and R2 IP Softphone and IP Agent, which use a dual connect (two extensions) architecture, are no longer supported. R3 and R4 IP Softphone and IP Agent, which use a single connect (one extension) architecture, continue to be supported. This applies to the RoadWarrior and the Telecommuter configurations for the IP Softphone. Native H.323 registrations for R1 and R2 Softphones continue to be supported.

Note:

Beginning with the January 2005 release of Communication Manager, the Message Flow Control feature is available. When this feature is enabled, previously existing IP soft clients will fail to register. The clients listed below are impacted by this condition.

- IP Softphone R3V1
- IP Softphone R3V2
- IP Softphone R3V2.1
- IP Softphone R4
- IP Softphone R5.0
- IP Softphone R5.1
- IP Agent V3
- IP Agent R4
- IP AgentR5
- Softconsole 1.0
- Softconsole 1.5

For assistance, contact your Avaya technical support representative.

Road-warrior mode

The road-warrior (VoIP) mode enables travelers to use Communication Manager features from temporary remote locations, such as a hotel room. The road-warrior configuration uses two separate software applications running on a user’s PC that is connected to Communication Manager over an IP network. The single network connection carries two channels: one for call-control signaling and one for voice. IP Softphone software handles the call signaling and an H.323 V2-compliant audio application such as Avaya iClarity, which is installed with Avaya IP Softphone Release 3 and later (or a third-party application, like Microsoft NetMeeting) handles the voice communications. The user’s PC provides the audio connection via a microphone and speakers.

Telecommuter mode

The telecommuter mode enables remote workers to use Communication Manager features from a remote location, such as while telecommuting from a home office. The telecommuter configuration uses two connections to Avaya Communication Manager:

- a connection to the PC over the IP network
- a connection to the telephone over the public-switched telephone network (PSTN)

The PC user places and takes calls with the IP Softphone interface and uses the telephone handset to speak and listen.

You can also use a variation of the telecommuter for call center agents: Avaya IP Agent. This mode uses the Avaya IP Agent interface instead of the IP Softphone interface to emulate a remote CallMaster telephone.

Stand-alone H.323

The stand-alone H.323 mode enables travelers to use some Avaya Communication Manager features from a remote location. This mode uses a PC running an H.323 v2-compliant audio application, such as Microsoft NetMeeting. The H.323 mode controls the call signaling and the voice path. However, since it does not use the IP Softphone interface, this configuration is capable of operating only as an analog or single-line telephone making one call at a time without any additional assigned features. You can provide stand-alone H.323 users only features that can they can activate with dial access codes.

Control of IP Telephone

This mode allows you to make and receive calls under the control of the IP Softphone - just like in the Telecommuter or Road Warrior mode. The big difference is that you have a true, feature-rich digital telephone under your control. In the Road Warrior mode, there is no telephone. In the Telecommuter mode, the telephone you are using (whether analog, digital, or IP telephone) is used only to carry audio and not for any features. In this mode (if you have an IP telephone), you get the best of both worlds. The terminals that are supported for this registration mode are shown in Supported Terminals for Control of Terminal/Station on page 706.

Control of DCP Telephone

This feature provides a registration endpoint configuration that will allow an IP softphone and a non-softphone telephone to be in service on the same extension at the same time. In this new configuration, the call control is done by both the softphone and the telephone endpoint. This feature supports speakerphones, handsets, and headsets. .

Table 8: Supported Terminals for Control of Terminal/Station *1 of 2*

Terminal Type	Control of the Terminal/Station			
	"Regular" Control		"Shared" Control	
	Road Warrior	Telecommuter	Via IP Telephone	Via the server
2402	Yes	Yes	N/A	Yes
2410	Yes	Yes	N/A	Yes
2420	Yes	Yes	N/A	Yes
2420 w/Expansion Module	Yes	Yes	N/A	Yes
4601	Yes	Yes	No	Yes
4602	Yes	Yes	No	N/A
4606	Yes	Yes	Yes	Yes
4610	Yes	Yes	Yes	Yes
4612	Yes	Yes	Yes	Yes
4620	Yes	Yes	Yes	Yes
4620 w/Expansion Module	Yes	Yes	Yes	N/A
4624	Yes	Yes	Yes	Yes
6402D	Yes	Yes	N/A	Yes
6408D/6408D+	Yes	Yes	N/A	Yes
6416D+	Yes	Yes	N/A	Yes
6416D+ w/Expansion Module	Yes	Yes	N/A	Yes
6424D+	Yes	Yes	N/A	Yes

1 of 2

Table 8: Supported Terminals for Control of Terminal/Station *2 of 2*

Terminal Type	Control of the Terminal/Station			
	"Regular" Control		"Shared" Control	
	Road Warrior	Telecommuter	Via IP Telephone	Via the server
6424D+ w/Expansion Module	Yes	Yes	N/A	Yes
8405D/8405D+	Yes	Yes	N/A	No
8410D	Yes	Yes	N/A	No
8411D	Yes	Yes	N/A	No
8434D	Yes	Yes	N/A	No
8434D w/Expansion Module	Yes	Yes	N/A	No
6402	Yes	Yes	N/A	N/A
6408/6408+	Yes	Yes	N/A	N/A
7403D	Yes	Yes	N/A	N/A
7404D	Yes	Yes	N/A	N/A
7405D	Yes	Yes	N/A	N/A
7406+/7406D	Yes	Yes	N/A	N/A
7407+/7407D	Yes	Yes	N/A	N/A
7410+/7410D	Yes	Yes	N/A	N/A
7434D	Yes	Yes	N/A	N/A
7444D	Yes	Yes	N/A	N/A
8403B	Yes	Yes	N/A	N/A
8405B/8405B+	Yes	Yes	N/A	N/A
8410B	Yes	Yes	N/A	N/A
8411B	Yes	Yes	N/A	N/A

2 of 2

Related topics

For instructions on how to administer an IP Softphone on your system, see Adding an IP Softphone on page 98.

You can also find information on the IP Softphone CD (see *DEFINITY IP Softphone Overview and Troubleshooting* and *DEFINITY IP Softphone Getting Started*).

Chapter 20: Screen Reference

This chapter contains descriptions of Communication Manager screens that are used in performing administrative tasks. These are most often screens that are invoked using commands such as **add**, **change**, and **remove**.

For maintenance-related screens that are invoked using commands such as **list**, **display**, and **status**, see *Maintenance Commands for Avaya Communication Manager, Media Gateways and Servers*, 03-300191.

AAR and ARS Digit Analysis Table

Your Avaya DEFINITY or media server running Avaya Communication Manager compares dialed numbers with the dialed strings in this table and determines the route pattern for the number.

Figure 254: AAR Digit Analysis Table screen

```
change aar analysis                                          Page 1 of X

                          AAR DIGIT ANALYSIS TABLE
                                                  Percent Full:  ___

     Dialed          Total    Route    Call   Node  ANI
     String        Min Max  Pattern   Type   Num   Reqd
_________________  __  __   _____    ____   ___    n
_________________  __  __   _____    ____   ___    n
_________________  __  __   _____    ____   ___    n
_________________  __  __   _____    ____   ___    n
_________________  __  __   _____    ____   ___    n
_________________  __  __   _____    ____   ___    n
_________________  __  __   _____    ____   ___    n
_________________  __  __   _____    ____   ___    n
_________________  __  __   _____    ____   ___    n
_________________  __  __   _____    ____   ___    n
```

Figure 255: ARS Digit Analysis Table screen

```
change ars analysis                                          Page 1 of X

                          ARS DIGIT ANALYSIS TABLE
                                  Location:  ___      Percent Full:  ___

     Dialed          Total    Route    Call   Node  ANI
     String        Min Max  Pattern   Type   Num   Reqd
_________________  __  __   _____    ____   ___    n
_________________  __  __   _____    ____   ___    n
_________________  __  __   _____    ____   ___    n
_________________  __  __   _____    ____   ___    n
_________________  __  __   _____    ____   ___    n
_________________  __  __   _____    ____   ___    n
_________________  __  __   _____    ____   ___    n
_________________  __  __   _____    ____   ___    n
_________________  __  __   _____    ____   ___    n
_________________  __  __   _____    ____   ___    n
```

ANI Reqd

Valid entries	Usage
y/n	Enter **y** if ANI is required on incoming R2-MFC or Russian MF ANI calls. This field applies only if the **Request Incoming ANI (non-AAR/ARS)** field on the **Multifrequency-Signaling-Related System Parameters** screen is **n**.
r	Allowed only if the **Allow ANI Restriction on AAR/ARS** field on the **Feature Related System Parameters** screen is **y**. Use to drop a call on a Russian Shuttle trunk or Russian Rotary trunk if the ANI request fails. Other types of trunks treat **r** as **y**.

Call Type (for AAR only)

Enter the call type associated with each dialed string. Call types indicate numbering requirements on different trunk networks. ISDN Protocols are listed in the table below.

Valid entries	Usage
aar	Regular AAR calls
intl	The Route Index contains public network ISDN trunks that require international type of number encodings.
pubu	The Route Index contains public network ISDN trunks that require unknown type of number encodings.
lev0 to lev2	Specify ISDN Private Numbering Plan (PNP) number formats. (See ISDN Numbering — Private on page 1200 for more information.)

ISDN Protocol

Call Type	Numbering Plan Identifier	Type of Numbering
aar	E.164(1)	national(2)
intl	E.164(1)	international(1)
pubu	E.164(1)	unknown(0)
lev0	PNP(9)	local(4)
lev1	PNP(9)	Regional Level 1(2)
lev2	PNP(9)	Regional Level 2(1)

Call Type (for ARS only)

Valid entries	Usage	China # 1 Call Type
alrt	alerts attendant consoles or other digital phones when an emergency call is placed	normal
emer	emergency call	normal
fnpa	10-digit North American Numbering Plan (NANP) call (11 digits with Prefix Digit "1")	attendant
hnpa	7-digit NANP call	normal
intl	public-network international number	toll-auto
iop	international operator	attendant
locl	public-network local number	normal
lpvt	local private	normal
natl	non-NANP	normal
npvt	national private	normal
nsvc	national service	normal
op	operator	attendant
pubu	public-network number (E.164)-unknown	normal
svcl	national(2)	toll-auto
svct	national(2)	normal
svft	service call, first party control	local
svfl	service call, first party control	toll

Dialed String

User-dialed numbers are matched to the dialed string entry that most closely matches the dialed number. For example, if a user dials 297-1234 and the AAR or ARS Digit Analysis Table has dialed string entries of 297-1 and 297-123, the match is on the 297-123 entry.

An exact match is made on a user-dialed number and dialed string entries with wildcard characters and an equal number of digits. For example, if a user dials 424, and there is a 424 entry and an X24 entry, the match is on the 424 entry.

Valid entries	Usage
0 to **9**	Enter up to 18 digits that the call-processing server analyzes.
***, x, X**	wildcard characters

Location (for the ARS Digit Analysis Table)

This is a display-only field on the **ARS Digit Analysis Table** screen.

Valid display	Usage
1 to **64**	Defines the location of the server running Avaya Communication Manager that uses this ARS Digit Analysis Table. On the **System-Parameters Customer-Options** screen, the **ARS** field and the **Multiple Locations** field must be set to **y** for values other than **all** to appear.
all	Indicates that this ARS Digit Analysis Table is the default for all port network (cabinet) locations. Appears only if the **Multiple Locations** field is **n** on the **System-Parameters Customer-Options** screen.

Max

Valid entries	Usage
Between Min and **28**	Enter the maximum number of user-dialed digits the system collects to match to the dialed string.

Min

Valid entries	Usage
Between **1** and Max	Enter the minimum number of user-dialed digits the system collects to match to the dialed string.

Node Number

Valid entries	Usage
1 to **999** or blank	Enter the number of the destination node in a private network if you are using node number routing or DCS. If you complete this field, leave the **Route Index** field blank.

Percent Full

Displays the percentage (**0** to **100**) of the system's memory resources that have been used by AAR/ARS. If the figure is close to 100%, you can free up memory resources.

Route Pattern

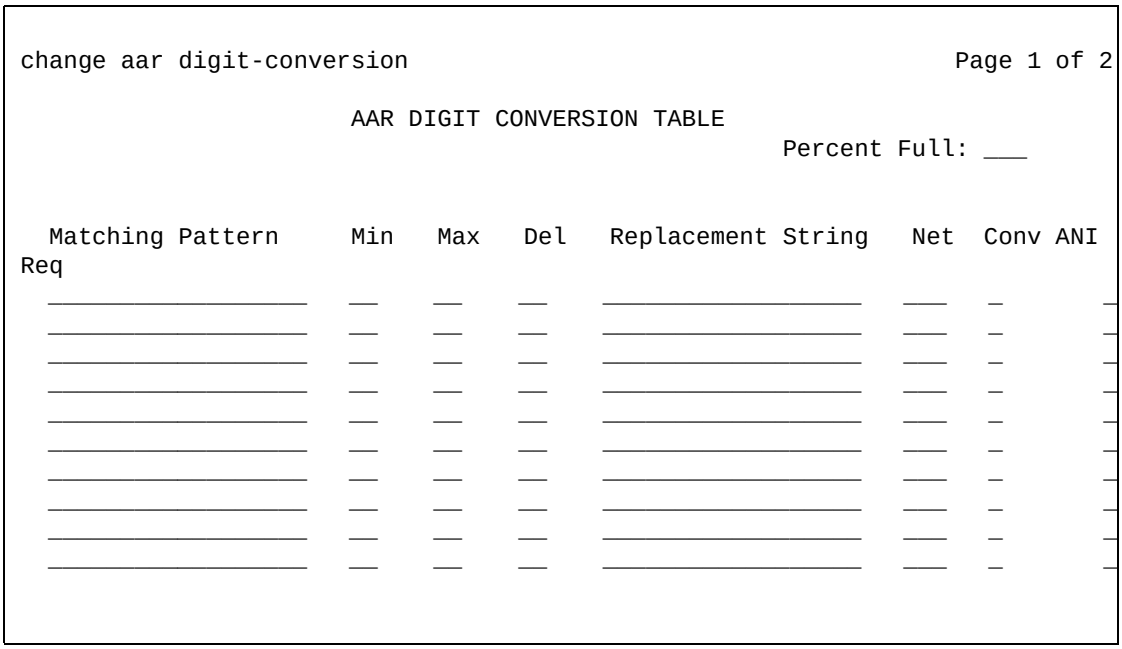
Enter the route number you want the server running Avaya Communication Manager to use for this dialed string.

Valid entries	Usage
p1 to **p2000**	Specifies the route index number established on the Partition Routing Table
1 to **640**	Specifies the route pattern used to route the call.
1 to **999**	Specifies the route pattern used to route the call. For S8300 Media Server only.
r1 to **r32**	Specifies the remote home numbering plan area table. Complete this field if RHNPA translations are required for the corresponding dialed string.
node	Designates node number routing
deny	Blocks the call

AAR and ARS Digit Conversion Table

Your system uses the AAR or ARS Digit Conversion Table to change a dialed number for more efficient routing. Digits may be inserted or deleted from the dialed number. For instance, you can tell the server running Communication Manager to delete a 1 and an area code on calls to one of your locations, and avoid long-distance charges by routing the call over your private network.

Figure 256: AAR Digit Conversion Table screen



```
change aar digit-conversion                                    Page 1 of 2

                      AAR DIGIT CONVERSION TABLE
                                                 Percent Full: ___


  Matching Pattern     Min   Max   Del   Replacement String   Net  Conv ANI
Req
  ____________________  __    __    __   ____________________  ___   _       _
  ____________________  __    __    __   ____________________  ___   _       _
  ____________________  __    __    __   ____________________  ___   _       _
  ____________________  __    __    __   ____________________  ___   _       _
  ____________________  __    __    __   ____________________  ___   _       _
  ____________________  __    __    __   ____________________  ___   _       _
  ____________________  __    __    __   ____________________  ___   _       _
  ____________________  __    __    __   ____________________  ___   _       _
  ____________________  __    __    __   ____________________  ___   _       _
  ____________________  __    __    __   ____________________  ___   _       _
  ____________________  __    __    __   ____________________  ___   _       _
```

Figure 257: ARS Digit Conversion Table screen

```
change ars digit-conversion                                  Page 1 of 2

                      ARS DIGIT CONVERSION TABLE
                          Location:  1                Percent Full: ___

  Matching Pattern     Min   Max   Del   Replacement String   Net Conv ANI Req
  ____________________ __    __    __    ____________________ ___   _       _
  ____________________ __    __    __    ____________________ ___   _       _
  ____________________ __    __    __    ____________________ ___   _       _
  ____________________ __    __    __    ____________________ ___   _       _
  ____________________ __    __    __    ____________________ ___   _       _
  ____________________ __    __    __    ____________________ ___   _       _
  ____________________ __    __    __    ____________________ ___   _       _
  ____________________ __    __    __    ____________________ ___   _       _
  ____________________ __    __    __    ____________________ ___   _       _
  ____________________ __    __    __    ____________________ ___   _       _
  ____________________ __    __    __    ____________________ ___   _       _
  ____________________ __    __    __    ____________________ ___   _       _
  ____________________ __    __    __    ____________________ ___   _       _
```

Note:

When you access the screen with the **display** or **change** command, the entries are sorted in the order of the matching pattern. Digits appear before characters.

ANI Req

This field applies only if the **Request Incoming ANI (non-AAR/ARS)** field on the **Multifrequency-Signaling-Related System Parameters** screen is **n**.

Valid entries	Usage
y/n	Enter **y** to require ANI on incoming R2-MFC or Russian MF ANI calls.
r	Allowed only if the **Allow ANI Restriction on AAR/ARS** field is **y** on the **Feature-Related System Parameters** screen. Use to drop a call on a Russian Shuttle trunk or Russian Rotary trunk if the ANI request fails. Other types of trunks treat **r** as **y**.

Conv

Valid entries	Usage
y/n	Enter **y** to allow additional digit conversion.

Del

Valid entries	Usage
0 to Min	Enter the number of digits you want the system to delete from the beginning of the dialed string.

Location (for ARS only)

This is a display-only field.

Valid display	Usage
1 to **64**	Defines the location of the server running Avaya Communication Manager for this ARS Digit Conversion Table. On the **System-Parameters Customer-Options** screen, the **ARS** field and the **Multiple Locations** field must be set to **y** for values other than **all** to appear.
all	Indicates that this ARS Digit Conversion Table is the default for all port network (cabinet) locations.

Matching Pattern

Valid entries	Usage
0 to **9** (1 to 18 digits)	Enter the number you want the server running Avaya Communication Manager to match to dialed numbers. If a Prefix Digit 1 is required for 10-digit direct distance dialing (DDD) numbers, be sure the matching pattern begins with a 1.
***, x, X**	wildcard characters

Max

Valid entries	Usage
Min to **28**	Enter the maximum number of user-dialed digits the system collects to match to this Matching Pattern.

Min

Valid entries	Usage
1 to Max	Enter the minimum number of user-dialed digits the system collects to match to this Matching Pattern.

Net

Enter the call-processing server network used to analyze the converted number.

Valid entries	Usage
ext, aar, ars	Analyze the converted digit-string as an extension number, an AAR address, or an ARS address.

Percent Full

Displays the percentage (**0** to **100**) of the system's memory resources that have been used by AAR/ARS. If the figure is close to 100%, you can free-up memory resources.

Replacement String

Valid entries	Usage
0 to **9** (1 to 18 digits)	Enter the digits that replace the deleted portion of the dialed number. Leave this field blank to simply delete the digits.
*****	
#	Use # to indicate end-of-dialing. It must be at the end of the digit-string.
blank	

Abbreviated Dialing List

This screen establishes system-wide or personal lists for speed dialing.

Enhanced List

The Enhanced Abbreviated Dialing List can be accessed by users to place local, long-distance, and international calls; to activate/deactivate features; or to access remote computer equipment.

Note:

Dialing must be enabled in your license file before you can program an Enhanced List. When the feature is enabled, the **Abbreviated Dialing Enhanced List** field on the **System-Parameters Customer-Options** screen displays **y**.

You can define two Enhanced Abbreviated Dialing Lists in the system. Before you assign numbers to a list, you must define whether you want a 3-digit or 4-digit enhanced list on the **Feature-Related System Parameters** screen. If you select 3-digit enhanced list, the list can be up to 10 separate screens numbered from 0 to 9 that allow you to define up to 1000 numbers. If you select a 4-digit enhanced list, a list can include up to 100 separate screens numbered 0 to 99 that allow you to assign up to 10,000 numbers on each list. The two Enhanced Abbreviated Dialing Lists together can support up to 20,000 entries.

If you want your attendants to use abbreviated dialing, you must also administer the **Console Parameters** screen.

Field descriptions for page 1

Figure 258: Abbreviated Dialing Enhanced List screen

```
display abbreviated-dialing enhanced                          Page 1 of 4
                                ABBREVIATED DIALING LIST
                                    Enhanced List
                     Size (multiple of 5): 5                Privileged? n
    DIAL CODE
      100: _____________________________
      101: _____________________________
      102: _____________________________
      103: _____________________________
      104: _____________________________
      105: _____________________________
```

DIAL CODE

Enter the number you want the system to dial when users enter this dial code. While the system is waiting, a call progress tone receiver is tied up, and, since there are a limited number of receivers in the system, outgoing calling capability may be impaired.

Vector Directory Number extension may also be assigned.

Valid entries	Usage
Digits **0** to **9**	Up to 24 characters
***** (star)	Part of FAC
# (pound)	Part of FAC
~p	Pause 1.5 seconds
~w	Wait for dial tone
~m	Change to outpulse DTMF digits at the end-to-end rate
~s	Start suppressing display of the digits being outpulsed
~W	Wait indefinitely for dial tone. Use this only if network response time is more than 30 seconds. Not available for S8300 Media Server.

Privileged

Indicates whether users of this list can dial any number in the list, regardless of the COR of the station from which they dial.

Valid entries	Usage
y/n	Set this field to **n** if you want the system to verify that this station is allowed to dial this number.

Size (multiple of 5)

The number of dial code list entries you want in this list.

Valid entries	Usage
5 to **100**, in multiples of 5	Up to 100 entries per screen

Group List

This screen implements the Abbreviated Dialing Group List. The Group Lists are controlled by the System Administrator. Up to 100 numbers can be entered per group list that can be accessed by users to place local, long-distance, and international calls; to activate/deactivate features; or to access remote computer equipment.

Field descriptions for page 1

Figure 259: Abbreviated Dialing Group List screen

```
change abbreviated-dialing group                                  Page 1 of X
                            ABBREVIATED DIALING LIST
                               Group List: ____
       Size (multiple of 5):   5  Program Ext: _________   Privileged? n
    DIAL CODE
         01: ____________________
         02: ____________________
         03: ____________________
         04: ____________________
         05: ____________________
```

DIAL CODE

Enter the number you want the system to dial when users enter this dial code. While the system is waiting, a call progress tone receiver is tied up, and, since there are a limited number of receivers in the system, outgoing calling capability may be impaired.

Only 1 through 5 display initially. If you enter a number greater than 5 in the **Size** field, the system increases the number of dial codes to the number you specified.

Vector Directory Number extension may also be assigned.

Valid entries	Usage
Digits **0** to **9**	Up to 24 characters
***** (star)	Part of FAC
# (pound)	Part of FAC
~p	Pause 1.5 seconds

1 of 2

Valid entries	Usage
~w	Wait for dial tone
~m	Change to outpulse DTMF digits at the end-to-end rate
~s	Start suppressing display of the digits being outpulsed
~W	Wait indefinitely for dial tone. Use this only if network response time is more than 30 seconds. Not available for S8300 Media Server.
	2 of 2

Group List

This is a display-only field when the screen is accessed using an administration command such as **add** or **change**.

Valid entries	Usage
Display-only field	Enter a group number when completing a paper screen.

Privileged

Valid entries	Usage
y	If **y** is entered, the calling telephone's class of restriction (COR) is never checked and any number in the group list will be dialed.
n	If **n** is entered, the calling telephone's COR is checked to determine if the number can be dialed.

Program Ext

Enter the extension that you want to give permission to program the Group List.

Size (multiple of 5)

Enter the number of abbreviated dialing numbers you want to assign in multiples of 5, up to 100.

Personal List

This screen establishes a personal dialing list for telephone/data module users. The personal list must first be assigned to the telephone by the System Administrator before the telephone user can add entries in the list. The lists can be accessed by users to place local, long-distance, and international calls; to activate/deactivate features; or to access remote computer equipment.

Figure 260: Abbreviated Dialing Personal List screen

```
change abbreviated-dialing personal                          Page 1 of 4
                        ABBREVIATED DIALING LIST

                  Personal List: _________ List Number: ___
                  Size (multiple of 5): 5
DIAL CODE
     01: ____________________
     02: ____________________
     03: ____________________
     04: ____________________
     05: ____________________
     06: ____________________
     07: ____________________
     08: ____________________
     09: ____________________
     00: ____________________
```

DIAL CODE

Enter the number you want the system to dial when users enter this dial code. While the system is waiting, a call progress tone receiver is tied up, and, since there are a limited number of receivers in the system, outgoing calling capability may be impaired.

Only 1 through 5 display initially. If you enter a number greater than **5** in the **Size** field, the system increases the number of dial codes to the number you specified.

Note:

Although the **Abbreviated Dialing Personal List** screen shows dial codes with a leading zero (i.e., 01, 02, 03), the user enters only the digit following the zero and not the zero itself to successfully access the extension administered on that dial code.

Vector Directory Number extension may also be assigned.

Valid entries	Usage
Digits 0 to 9	Up to 24 characters
***** (star)	Part of FAC
# (pound)	Part of FAC
~p	Pause 1.5 seconds
~w	Wait for dial tone
~m	Change to outpulse DTMF digits at the end-to-end rate
~s	Start suppressing display of the digits being outpulsed
~W	Wait indefinitely for dial tone. Only use this if network response time is more than 30 seconds.

List Number

A display-only field indicating which of the three personal lists is defined for the telephone.

Personal List

A display-only field indicating the extension of the telephone that will use this list.

Size (multiple of 5)

Enter the number of abbreviated dialing numbers you want to assign in multiples of 5, up to 100.

System List

This screen implements a system abbreviated-dialing list. Only one system list can be assigned and is administered by the System Administrator. The list can be accessed by users to place local, long-distance, and international calls; to activate/deactivate features; or to access remote computer equipment.

Page 1 of 7 of the screen

Figure 261: Abbreviated Dialing System List screen

```
add abbreviated-dialing system                                  Page  1 of 7
                        ABBREVIATED DIALING LIST

                                SYSTEM LIST
Size (multiple of 5): 100      Privileged? n      Label Language:english
DIAL CODE                                  LABELS FOR 2420/4620 STATIONS
  11:                                    11:*************
  12:                                    12:*************
  13:                                    13:*************
  14:                                    14:*************
  15:                                    15:*************
  16:                                    16:*************
  17:                                    17:*************
  18:                                    18:*************
  19:                                    19:*************
  20:                                    20:*************
  21:                                    21:*************
  22:                                    22:*************
  23:                                    23:*************
  24:                                    24:*************
  25:                                    25:*************
```

DIAL CODE

Enter the number you want the system to dial when users enter this dial code. While the system is waiting, a call progress tone receiver is tied up, and, since there are a limited number of receivers in the system, outgoing calling capability may be impaired.

Only 1 through 5 display initially. If you enter a number greater than 5 in the **Size** field, the system increases the number of dial codes to the number you specified.

Vector Directory Number extension may also be assigned.

Valid entries	Usage
Digits 0 to 9	Up to 24 characters
***** (star)	Part of FAC
# (pound)	Part of FAC
~p	Pause 1.5 seconds

1 of 2

Valid entries	Usage
~w	Wait for dial tone
~m	Change to outpulse DTMF digits at the end-to-end rate
~s	Start suppressing display of the digits being outpulsed
~W	Wait indefinitely for dial tone. Use this only if network response time is more than 30 seconds.
	2 of 2

Label Language

This field provides administration of personalized labels on the 2420/4620 telephone sets. If this field is changed to another language, all administered labels in the original language are saved and the labels for the new language are read in and displayed.

Valid entries	Usage
English, Italian, French, Spanish, user-defined, or **Unicode**	Enter the appropriate language for the 2420/4620 labels. **Note:** Unicode display is only available for Unicode-supported telephones. Currently, only the 4620SW and 4610SW support Unicode display.

LABELS FOR 2420/4620 STATIONS

This field provides the administrative capability to actually customize the labels for the system-wide Abbreviated Dial buttons on the 2420/4620 telephone sets.

Valid entries	Usage
A-Z, a-z, 0-9, and ! & * ? ; ' ^ ( ) , . : -	Up to 15 alphanumeric characters

Privileged

Valid entries	Usage
y	Enter **y** if the originating party's class of restriction (COR) is never checked and any number in the list can be dialed.
n	Enter **n** if the COR is to be checked to determine if the number can be dialed.

Size (multiple of 5)

Enter the number of abbreviated dialing numbers you want to assign in multiples of 5, up to 100.

The Figure 262 shows the last page of the **Abbreviated Dialing System** screen when, on the **System-Parameters Customer-Options** screen, the **A/D Grp/Sys List Dialing Start at 01** field is **n**.

Figure 262: Abbreviated Dialing System List screen

```
add abbreviated-dialing system                                Page  7 of 7
                        ABBREVIATED DIALING LIST

                                  SYSTEM LIST
                                         Label Language:english
DIAL CODE                                  LABELS FOR 2420/4620 STATIONS
  01:                                   01:*************
  02:                                   02:*************
  03:                                   03:*************
  04:                                   04:*************
  05:                                   05:*************
  06:                                   06:*************
  07:                                   07:*************
  08:                                   08:*************
  09:                                   09:*************
  10:                                   10:*************
```

Figure 263 shows the last page of the **Abbreviated Dialing System** screen when, on the **System-Parameters Customer-Options** screen, the **A/D Grp/Sys List Dialing Start at 01** field is **y**.

Figure 263: Abbreviated Dialing System List screen

```
add abbreviated-dialing system                                  Page  7 of 7
                         ABBREVIATED DIALING LIST

                                    SYSTEM LIST
                                           Label Language:english
DIAL CODE                                    LABELS FOR 2420/4620 STATIONS
  91:                                     91:**************
  92:                                     92:**************
  93:                                     93:**************
  94:                                     94:**************
  95:                                     95:**************
  96:                                     96:**************
  97:                                     97:**************
  98:                                     98:**************
  99:                                     99:**************
  00:                                     00:**************
```

7103A Button List

This screen assigns abbreviated dialing numbers to the 7103A telephone buttons. The entries can then be accessed by 7103A telephone users to place local, long-distance, and international calls; activate/deactivate features; or to access remote computer equipment. This screen applies only to 7103A fixed feature phones. Only one 7103A abbreviated dialing list can be implemented in the system and it applies to all 7103A fixed feature phones in the system. This list is controlled by the System Administrator.

Pages 1 through 4 of the screen

Figure 264: Abbreviated Dialing List — 7103A Button List screen

```
display abbreviated-dialing 7103A-buttons                  Page 1 of 1
                           ABBREVIATED DIALING LIST
                              7103A Button List

      DIAL CODE (FOR THE 7103A STATION BUTTONS)
       1: _____________________________      5. _____________________________
       2: _____________________________      6. _____________________________
       3: _____________________________      7. _____________________________
       4: _____________________________      8. _____________________________
```

DIAL CODE

Enter the number you want to assign to each dial code (button). Any additions or changes apply to all 7103A fixed feature phones. While the system is waiting, a call progress tone receiver is tied up, and, since there are a limited number of receivers in the system, outgoing calling capability may be impaired.

Vector Directory Number extension may also be assigned.

Valid entries	Usage
Digits 0 to 9	Up to 24 characters
***** (star)	Part of FAC
# (pound)	Part of FAC
~p	Pause 1.5 seconds
~w	Wait for dial tone
~m	Mark
~s	Start suppressing display of the digits being outpulsed
~W	Wait indefinitely for dial tone. Use this only if network response time is more than 30 seconds. Not available for S8300 Media Server.

Access Endpoint

This screen administers Access Endpoints and Wideband Access endpoints.

Note:

You can administer Wideband Access Endpoints only if, on the **System-Parameters Customer-Options** screen, the **Wideband Switching** field is **y**.

An Access Endpoint is a nonsignaling trunk that neither responds to signaling nor generates signaling. Access Endpoints eliminate the need to dedicate an entire trunk group for the access of a single trunk by providing the capability to assign an extension number to a single trunk.

An Access Endpoint can be specified as the Originator or Destination endpoint of an administered connection.

A Wideband Access Endpoint (WAE) is an endpoint application connected to line-side non-ISDN T1 or E1 facilities and, like Access Endpoints, have no signaling interface with the system. For information on endpoint applications connected to line-side ISDN-PRI facilities, see on page 1347.

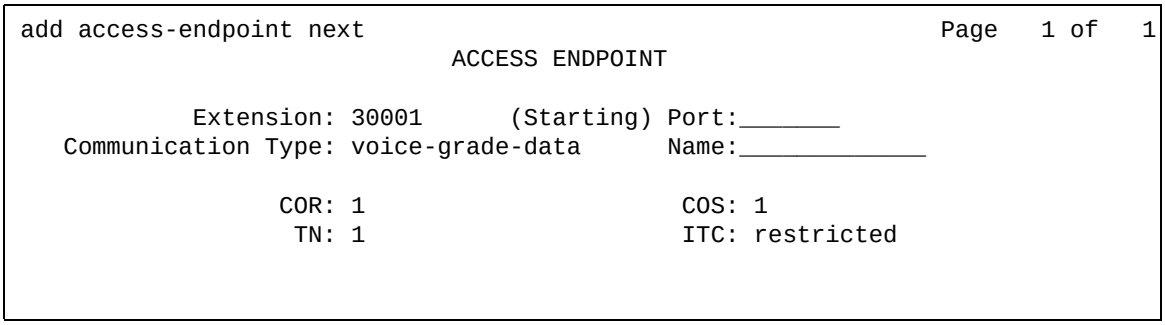
The WAE is defined by a starting port (DS0) and a width specifying the number of adjacent nonsignaling DS0s (positioned within a DS1 facility) that make up the endpoint. This width may be between 2 and 31 adjacent DS0s.

Note:

Access Endpoints and Wideband Access Endpoints consume the same resources that trunks use. Thus, the sum of Access Endpoints and trunks cannot exceed the maximum number of trunks available in your system configuration.

Field descriptions for page 1

Figure 265: Access Endpoint screen



```
add access-endpoint next                                      Page   1 of   1
                              ACCESS ENDPOINT

           Extension: 30001      (Starting) Port:_______
   Communication Type: voice-grade-data      Name:_____________

                 COR: 1                       COS: 1
                  TN: 1                       ITC: restricted
```

Communication Type

Valid entries	Usage
voice-grade-data	For an analog tie trunk access endpoint.
56k-data	For a DS1 access endpoint enter as appropriate (**64k-data** is not allowed for robbed-bit trunks).
64k-data	
wideband	For a Wideband access endpoint

COR

The COR is administered so that only an administered connection (AC) endpoint can be connected to another AC endpoint.

Valid entries	Usage
0 to **95**	Enter the appropriate class of restriction (COR) number.

COS

The COS is administered (see Class of Service on page 830) so that the use of the Call Forwarding All Calls feature for access endpoints is prohibited.

Valid entries	Usage
0 to **15**	Enter the appropriate COS number.

Extension

A display-only field showing the extension number as specified in the command line, or shows the next available extension number if **next** was entered on the command line. This is the extension number assigned to the nonsignaling trunk and used to access the trunk endpoint.

ITC (Information Transfer Capability)

This field is used to determine the type of transmission facilities to be used for ISDN calls originating from this endpoint. Displays when the **Communication Type** field is **56k-data**, **64k-data**, or **Wideband**.

When adding an access endpoint with the ITC administered as unrestricted, its associated port has to be a channel of a DS1 circuit pack with the **Zero Code Suppression** field administered as B8ZS. If the port is not a channel of a DS1 circuit pack with its **Zero Code Suppression** field administered as B8ZS, the end validation fails and the screen submission is rejected. The cursor is moved to ITC with the following error message:

An unrestricted access endpoint can only be from B8ZS DS1 circuit pack.

When adding an access endpoint with the ITC administered as restricted, its associated port can be a channel from a DS1 circuit pack with the **Zero Code Suppression** field administered as ZCS or B8ZS.

For an existing access endpoint, ITC can only be changed from restricted to unrestricted if its associated port is a channel of a DS1 circuit pack with its **Zero Code Suppression** field administered as B8ZS. If the port is not a channel of a DS1 circuit pack with its **Zero Code Suppression** field administered as B8ZS, the end validation fails and the screen submission is rejected. The cursor is moved to ITC with the following error message:

`An unrestricted access endpoint can use only B8ZS DS1 circuit pack`

Without this end validation, a user could administer an access endpoint as unrestricted when in fact it is restricted, that is, its associated port is a member of a DS1 circuit pack that uses ZCS data transmission.

Valid entries	Usage
unrestricted	When **unrestricted**, only unrestricted transmission facilities (b8zs) will be used to complete the call. An unrestricted facility is a transmission facility that does not enforce 1's density digital transmission (that is, digital information is sent exactly as is). For Wideband Access Endpoints, enter **unrestricted**.
restricted	When **restricted**, either restricted (zcs-ami) or unrestricted transmission facilities is used to complete the call. A restricted facility is a transmission facility that enforces 1's density digital transmission (that is, a sequence of eight digital zeros is converted to a sequence of seven zeros and a digital one) via zcs coding on DS1 circuit pack.

Name

Enter an name for the endpoint.

(Starting) Port

Enter the necessary characters.

Valid entries	Usage
01 to **44** (DEFINITY R) or **01** to **03** (DEFINITY CSI, SI) or **1** to **64** (S8700 or S8710 IP-Connect / S8700 or S8710 Multi-Connect / S8300 Media Server)	First and second characters are the cabinet number.
A to **E**	Third character is the carrier.
0 to **20**	Fourth and fifth characters are the slot number.
01 to **04** (Analog TIE trunks) **01** to **31** **1** to **80** (DEFINITY R, CSI) or **1** to **10** (DEFINITY SI) or **1** to **250** (S8700 or S8710 IP-Connect / S8700 or S8710 Multi-Connect / S8300 Media Server)	Six and seventh characters are the circuit number. Gateway
V1 to **V9**	Module
01 to **31**	Circuit

For example, 01A0612 is in cabinet 01, carrier A, slot 06, and circuit number (port) 12.

Note:

For Wideband Access Endpoints, analog tie trunks cannot be used and the DS1 Interface circuit pack, Version C or later, must be used.

The DS1 circuit number corresponds to the channel that will carry the data traffic. Channels 1 through 31 (DS1 Interface only) or channels 1 through 24 (DS1 Tie Trunk, DS1 Interface, or DS1 Interface (32) circuit packs) can be used when the **DS1 Signaling Type** field is **robbed-bit** or **isdn-ext**. For Common Channel or ISDN-PRI signaling, channel use is limited to channels 1 through 30 (DS1 Interface circuit pack only) or channels 1 through 23 (DS1 Interface (32) or DS1 Interface). A channel can be administered as an access endpoint regardless of the DS1 signaling type.

TN

Valid entries	Usage
1 to 100	Enter the Tenant Partition number.

Width

Appears if the **Communication Type** field is **wideband**. This field cannot be blank.

Valid entries	Usage
2 to **31**	Enter the number of adjacent DS0 ports beginning with the specified Starting Port, that make up the WAE.
6	A width of **6** defines a 384 Kbps WAE.

Administered Connection

This screen assigns an end-to-end Administered Connection (AC) between two access endpoints or data endpoints. The AC is established automatically by the system whenever the system restarts or the AC is due to be active. See "Administered Connections" in *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205, and Access Endpoint on page 730 for additional information.

Field descriptions for page 1

Figure 266: Administered Connection screen

```
change administered-connection                                  Page 1 of 1
                        ADMINISTERED CONNECTION
  Connection Number:  1                          Enable? y
         Originator:  ________
        Destination:  ____________________________________
               Name:  ____________________

AUTHORIZED TIME OF DAY

         Continuous? n
                     Sun? n Mon? n Tue? n Wed? n Thu? n  Fri? n  Sat? n
         Start Time:  00:00
           Duration:  000:00

  MISCELLANEOUS PARAMETERS

                 Alarm Type:  warning  Alarm Threshold: 5
                                        Retry Interval: 2
                   Priority:  5       Auto Restoration? y
```

Connection Number

This is a display-only field showing an unassigned AC number when the screen is accessed using an administration command such as **change** or **display**.

Destination

Used to route the AC to a desired endpoint. Enter the address of the destination access or data endpoint. This endpoint is the terminating party of the AC and need not be local to the server on which the AC is assigned. The entry must be consistent with the local Communication Manager server's dial plan (that is, the first digits are assigned as an extension, feature access code, or trunk access code, or DDD Number). If a local extension is entered, it must be assigned to either an access or data endpoint. Abbreviated Dialing entries may be used in this field.

Valid entries	Usage
Extension/string	Enter the assigned access endpoint/data module extension or valid dialed string.

Enable

Provides the administered connection.

Valid entries	Usage
y	Indicates an attempt will be made to establish the AC when the AC is due to be active.
n	The AC will not be made or if it is up, it will drop.

Name

Valid entries	Usage
Up to 27 alphanumeric characters. Up to 15 alphanumeric characters (S8300 Media Server, S8700 or S8710 IP-Connect only)	Enter a short identification of the AC.

Originator

Enter the assigned access endpoint/data module extension.

Data Line circuit pack

- Asynchronous EIA 232C compatible equipment

Digital Line circuit pack connections, including:

- MPDM (700D), MTDM (700B, 700C, 700E), 7400D data module
- 7400A, 7400B, 7400C HSL, 8400B data module
- 7401D telephone with 7400B or 8400B data module
- 7403D/7405D/7407D/7410D/7434D telephone with DTDM or 7400B or 8400B data module
- 7404D or 7406D telephone
- 510D personal terminal
- 515 BCT, 615 BCT, or 715 BCT terminal
- Connection between PC and the server running Avaya Communication Manager

ISDN-BRI Line circuit pack connections, including:

- 7500 data module
- 7505D/7506D/7507D telephone with ADM

Valid entries	Usage
Assigned access endpoint/ data module extension	The endpoint must be local to the server on which the AC is administered. Nonsignaling DS1 trunk or analog tie trunk.

Authorized Time of Day

Continuous

The connection will be up all the time or re-established if the connection goes down.

Valid entries	Usage
y	Indicates that the AC is continuous (that is, not scheduled to be active at a certain time). If **y** is entered, the seven **Start Days** and associated **Duration** fields do not appear.
n	Displays the **Start Days** fields.

Duration

Enter the period of time that the scheduled AC should remain active. This period is specified in two fields separated by a colon. The maximum duration is 167 hours and 59 minutes (that is, 1 minute less than 1 week). Only appears if the **Continuous** field is **n**.

Valid entries	Usage
000 through **167**	For the **hour** field.
00 through **59**	For the **minute** field.

Start Days (Sun through Sat)

These fields indicate only the days on which an attempt will be made to establish the AC and not necessarily the days it is active. A scheduled AC may be active over a number of days, and, in this situation, these fields should be used only to specify the days on which the AC starts and not other days on which the AC may be active. Only appears if the **Continuous** field is **n**.

Valid entries	Usage
y	Enter **y** in each of the required days of the week fields to indicate that an attempt will be made to establish the AC.
n	Displays the day fields.

Start Time

Only appears if the **Continuous** field is **n**.

Valid entries	Usage
00:00 through **23:59**	Enter the time of the day when an attempt should begin to establish a scheduled AC. The time is specified in two fields separated by a colon.

Miscellaneous Parameters

Alarm Threshold

Only appears if an entry in the **Alarm Type** field is other than **none**. Enter the number of times an attempt to establish or reestablish an AC must fail consecutively before an AC alarm generates. (An alarm will be generated after the fourth retry has failed; thus, with the retry interval of 2 minutes, an alarm will be generated approximately 8 minutes after the first failure occurs.)

Valid entries	Usage
1 through **10**	An alarm generates on the first failure if this field is **1**.

Alarm Type

Enter the type of alarm to be generated if the AC cannot be initially established, or fails and cannot be reestablished, and the number of consecutive failures equals the alarm threshold. All AC alarms and the errors that caused the alarms are recorded in the system's alarm and error log. In addition, a status lamp associated with an attendant console or telephone feature button (ac-alarm) may be used to indicate the AC alarm.

Valid entries	Usage
major	Failures that cause critical degradation of service and require immediate attention.
minor	Failures that cause some degradation of service, but do not render a crucial portion of the system inoperable. This condition requires action, but its consequences are not immediate. Problems might be impairing service to a few trunks or stations or interfering with one feature across the entire system.
warning	Failures that cause no significant degradation of service or failures in equipment external to the system. Warning alarms are not reported to the attendant console or INADS.
none	The alarm notification may be disabled for this AC.

Auto Restoration

Valid entries	Usage
y	Enter **y** to indicate an attempt is to be made to reestablish an AC that failed. Auto restoration is available only for an AC that is established over an ISDN Software Defined Data Network (SDDN) trunk group. A **y** in this field is ignored in all other situations.

Priority

Enter a number that is to be used to determine the order in which ACs are to be established.

Valid entries	Usage
1 to **8**	1 is the highest and 8 the lowest priority.

Retry Interval

Valid entries	Usage
1 to **60**	Enter the number of minutes between attempts to establish or reestablish the AC.

Agent Login ID

Use this screen to add or change agent login IDs and skill assignments. If you add or change skills on the media server, the agent must log out and then log in again before the changes take effect. For more information, see the *Automatic Call Distribution (ACD) Guide*.

Field descriptions for page 1

Figure 267: Agent Login ID screen

```
add agent-loginID 9011                                          Page 1 of 1
                                  AGENT LOGINID

                 Login ID: 9011_                                    AAS? _
                     Name:_____________________________           AUDIX? _
                       TN: 1__                                LWC Reception: spe
                      COR: 1                          AUDIX Name for Messaging: _______
            Coverage Path: ____          Messaging Server Name for Messaging: _______
            Security Code: ____                    LoginID for ISDN Display? n
       Direct Agent Skill: ___                                  Password: _______
 Call Handling Preference: skill-level                Password (enter again): _______
        Service Objective? _                                   Auto Answer: ______

   SN RL SL PA         SN RL SL PA           SN RL SL PA           SN RL SL PA
1: __ _  __ ___    6:  __ _  __ ___     11: __ _  __ ___     16: __ _  __ ___
2: __ _  __ ___    7:  __ _  __ ___     12: __ _  __ ___     17: __ _  __ ___
3: __ _  __ ___    8:  __ _  __ ___     13: __ _  __ ___     18: __ _  __ ___
4: __ _  __ ___    9:  __ _  __ ___     14: __ _  __ ___     19: __ _  __ ___
5. __ _  __ ___   10:  __ _  __ ___     15: __ _  __ ___     20: __ _  __ ___

            WARNING:  Agent must log in again before skill changes take effect
```

AAS

Enter **y** if this extension is used as a port for an Auto Available Split/Skill. Default is **n**.

Entering **y** in the **AAS** field clears the password and requires execution of the remove agent-loginid command. To set AAS to **n**, remove this logical agent and add it again. This option is intended for switch adjunct equipment ports only, not human agents.

Audix

Enter **y** if this extension is used as a port for AUDIX. Default is **n**. The **AAS** and **AUDIX** fields cannot both be y.

Audix Name for Messaging

Do one of the following actions:

- Enter the name of the messaging system used for LWC Reception, or
- Enter the name of the messaging system that provides coverage for this Agent LoginID.

Auto Answer

When using EAS, the agent's auto answer setting applies to the station where the agent logs in. If the auto answer setting for that station is different, the agent's setting overrides the station's setting. The following entries are valid:

- **all** - immediately sends all ACD and non ACD calls to the agent.
- **acd** - only ACD split /skill calls and direct agent calls go to auto answer. If this field is **acd**, non ACD calls terminated to the agent ring audibly.
- **none** - all calls terminated to this agent receive an audible ringing treatment. This is the default.
- **station** - auto answer for the agent is controlled by the auto answer field on the **Station** screen.

Call Handling Preference

Options are **skill-level** (default), **greatest-need**, and **percent-allocation**. When calls are in queue and an agent becomes available, **skill-level** delivers the oldest, highest priority call waiting for the agent's highest level skill. **Greatest-need** delivers the oldest, highest priority call waiting for any of the agent's skills. **Percent allocation** delivers a call from the skill that will otherwise deviate most from its administered allocation. **Percent-allocation** is available only with Avaya Business Advocate software. For more information, see the *Avaya Business Advocate User Guide*.

COR

Enter the Class of Restriction for the agent. Valid entries are **0** to **95**. Default is **1**.

Coverage Path

Enter the number of the coverage path used by calls to the LoginID. Valid entries are a path number between **1** and **999**, time of day table **t1** to **t999**, or blank (default). This is used when the agent is logged out, busy, or does not answer.

Direct Agents Calls First

This field replaces the **Service Objective** field when percent-allocation is entered in the **Call Handling Preference** field. Enter **y** if you want direct agent calls to override the percent-allocation call selection method and be delivered before other ACD calls. Enter **n** if you want direct agent calls to be treated like other ACD calls. For more information, see the *Avaya Business Advocate User Guide*.

Direct Agent Skill

Enter the number of the skill used to handle Direct Agent calls. Valid entries are **1** to **99**, or blank (default).

Login ID

Display-only field. Contains the identifier for the Logical Agent as entered on the command line.

Login ID for ISDN Display

Enter **y** if the **Agent LoginID CPN** and **Name** field is to be included in ISDN messaging over network facilities. In this case, the physical station extension CPN and Name is sent. Default is **n**.

LWC Reception

Valid entries are **audix**, **msa-spe** (default), and **none**.

Messaging Server Name for Messaging

Do one of the following actions:

- Enter the name of the Messaging Server used for LWC Reception.
- Enter the name of the Messaging Server that provides coverage for this Agent LoginID.
- Leave blank (default).

Name

Enter up to a 27-character string naming the agent. Any alpha-numeric character is valid. Default is blank.

PA

Percent Allocation. If the call handling preference is percent-allocation, you must enter a percentage for each of the agent's skills. Enter a number between 1 and 100 for each skill. Your entries for all of the agent's skills together must total 100%. Do not use target allocations for reserve skills. Percent Allocation is available as part of the Advocate software.

Password

Appears only if both the **AAS** and **AUDIX** fields are **n**. Enter up to nine digits as the password the Agent must enter upon login. Valid entries are the digits **0** through **9**. Enter the minimum number of digits in this field specified by the **Minimum Agent-LoginID Password Length** field on the **Feature-Related System Parameters** screen. Default is blank. Values entered in this field are not displayed on the screen.

Password (enter again)

Appears only if both the **AAS** and **AUDIX** fields are n. Reenter the same password exactly as it was entered in the **Password** field. Default is blank. Values entered in this field are not displayed on the screen.

Port Extension

Appears only if either the **AAS** or **AUDIX** field is **y**. Enter the assigned extension for the AAS or AUDIX port. This extension cannot be a VDN or an Agent LoginID. Default is blank.

RL

Reserve Level. Enter any reserve levels assigned to this agent with the Service Level Supervisor feature. You can assign a reserve level of 1 or 2. When this skill reaches the corresponding EWT threshold set on the **Hunt Group** form, this agent will automatically be logged into the skill and will take calls until the skill's EWT drops below the preassigned overload threshold. Service Level Supervisor is available as part of the Advocate software.

SL

Skill Level. Enter a skill level for each of an agent's assigned skills. If EAS-PHD is not optioned, 2 priority levels are available. If EAS-PHD is optioned, 16 priority levels are available. In releases prior to R3V5, level 1 was the primary skill and level 2 was the secondary skill.

Security Code

Enter the 4-digit security code (password) for the Demand Print messages feature. This field can be blank (default).

Service Objective

Appears only when **Call Handling Preference** is **greatest-need** or **skill-level**. Valid entries are **y** or **n**. Service Objective is administered on the **Hunt Group** screen and the **Agent LoginID** screen. The server selects calls for agents according to the ratio of Predicted Wait Time (PWT) or Current Wait Time (CWT) and the administered service objective for the skill. Service Objective is a feature that is part of the Advocate software.

SN

Skill Number. Enter the Skill Hunt Group(s) that this agent handles. The same skill may not be entered twice. Consider the following options:

- If EAS-PHD is not optioned, enter up to four skills.
- If EAS-PHD is optioned, enter up to 20 or 60 skills depending on the platform.
- Assigning a large number of skills to agents can potentially impact system performance. Review system designs with the ATAC when a significant number of agents have greater than 20 skills per agent.

TN

Enter the Tenant Partition number. Valid entries are **1** to **20**. Default is **1**.

Alias Station

This screen allows you to configure the system so that you can administer new telephone types that are not supported by your system software. This screen maps new telephone models to a supported telephone model. This mapping does not guarantee compatibility, but allows nonsupported models to be administered and tracked by their own names.

Some administrators also use this screen to "name" non-telephone devices. For example, you know that you can add a modem to your system by simply administering the extension as the standard analog type 2500. But, if you listed your stations, how would you know which extensions are modems? Instead, you could use the **Alias** screen to create a ‘modem’ alias to type 2500 and enter modem in the **Type** field for every modem you add to your system. For more information, see Adding a Fax or Modem on page 96.

Tip:

When you upgrade a system that uses an alias set type to a new release, the system determines if the aliased type is supported in the new release (is now a native set type). When you review the **Alias Station** screen, you may see alias types that have become native. If the type is now native, the last character of the aliased set type becomes a "#."

Field descriptions for page 1

Figure 268: Alias Station screen

```
change alias station                                          Page 1 of 1
                                ALIAS STATION
                        Alias Set Type    Supported Set Type
                             _______           _______
                             _______           _______
                             _______           _______
                             _______           _______
                             _______           _______
                             _______           _______
                             _______           _______
                             _______           _______
                             _______           _______
                             _______           _______
                 '#' indicates previously aliased set type is now native
```

Alias Set Type

Enter up to a 5-character name for the non-supported telephone type that you want to alias to a similar supported telephone type. Do not use blank characters.

Supported Set Type

Enter a supported telephone type that you want to map (or alias) to the alias set type. Valid supported telephone types are listed in Telephones on page 1450.

Note:

Data Communication Protocol (DCP) telephone types must be aliased to DCP telephone types, hybrid types to hybrid types, and analog to analog types.

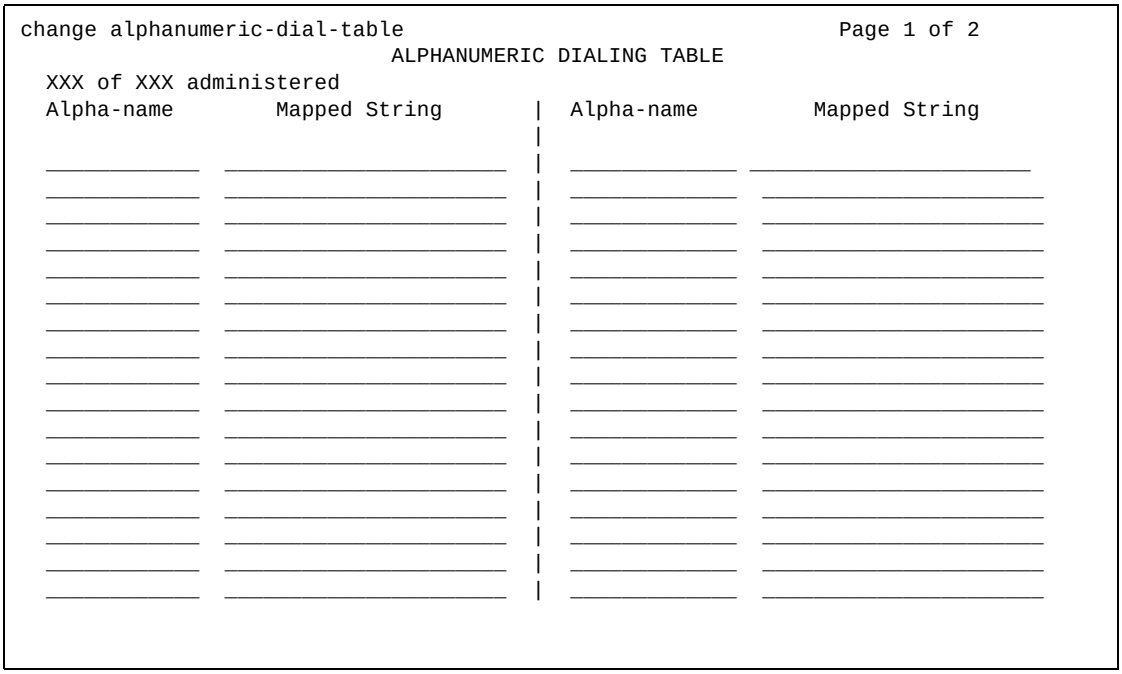
Alphanumeric Dialing Table

This screen associates alpha-names to dialed digit strings. This allows telephone users to place a data call by simply typing the alpha-name. Users need only remember far-end alpha-names instead of the actual digit strings.

The screen consists of paired **Alpha-name/Mapped String** fields. Entries may be made in any order on the screen. However, before the screen is displayed for changing or reviewing, the entries in the table are sorted alphanumerically by the alpha-name. All entries will be moved to the beginning of the table, leaving all blank entries at the end.

Field descriptions for page 1

Figure 269: Alphanumeric Dial Table screen



Alpha-name

All alpha-names in the table must be unique and cannot be referenced in their own "Mapped String." The alpha-names can be used in any other "Mapped String" and multiple times in a "Mapped String."

Valid entries	Usage
From 1 to 8 alphanumeric characters	The entry must start with an alphabetic character and may not have blank spaces between characters.

Mapped String

Enter from 1 to 24 characters that may contain alphanumeric, readability, delimiters, and/or special characters. The entry is used to generate the final dialing string and can include Facility Access Codes.

Note:

A Mapped String may not contain an Alpha-Name whose Mapped String also contains an Alpha-Name.

Valid entries	Usage
Digits 0 to 9	Numeric
A through **Z**, **a** through **z**	Alpha (note uppercase entries are mapped to lowercase)
(	Character delimiters used for easy reading of the dial string Delimiters consisting of both readability and special characters are used to separate tokens of "alpha-names" or numeric sub-strings. For example, "(205) mt-1234+0000"
)	
/	
-	
+	
%	
","	
space	
#	Each treated as a numeric
*	
^	Treated as a readability character

Announcements/Audio Sources

Use this screen to assign announcements to circuit packs and port locations.

Figure 270: Announcements/Audio Sources screen

```
change announcements                                          Page 1 of X
                          ANNOUNCEMENTS/AUDIO SOURCES

 Ann.
 No.  Ext.      Type       COR  TN    Name                    Q QLen Pr Rt Port
 1    _________ __________  1_  1_  ________________________ _ _    _  __ _____
 2    _________ __________  1_  1_  ________________________ _ _    _  __ _____
 3    _________ __________  1_  1_  ________________________ _ _    _  __ _____
 4    _________ __________  1_  1_  ________________________ _ _    _  __ _____
 5    _________ __________  1_  1_  ________________________ _ _    _  __ _____
 6    _________ __________  1_  1_  ________________________ _ _    _  __ _____
 7    _________ __________  1_  1_  ________________________ _ _    _  __ _____
 8    _________ __________  1_  1_  ________________________ _ _    _  __ _____
 9    _________ __________  1_  1_  ________________________ _ _    _  __ _____
 10   _________ __________  1_  1_  ________________________ _ _    _  __ _____
 11   _________ __________  1_  1_  ________________________ _ _    _  __ _____
 12   _________ __________  1_  1_  ________________________ _ _    _  __ _____
 13   _________ __________  1_  1_  ________________________ _ _    _  __ _____
 14   _________ __________  1_  1_  ________________________ _ _    _  __ _____
 15   _________ __________  1_  1_  ________________________ _ _    _  __ _____
 16   _________ __________  1_  1_  ________________________ _ _    _  __ _____
```

COR

Valid entries	Usage
0 to **95**	Enter the class of restriction (COR) you want associated with this announcement.

Ext

Valid entries	Usage
1 to 8 digits	Enter the extension you assign to this announcement. The following screens can reference this extension: **Hunt Group**, **Coverage Path**, **Trunk Group (Incoming Destination and Night Destination)**, **Vector**, **Feature-Related System Parameters** (**DID/Tie/ISDN Intercept Treatment**, **Controlled Restriction**).

CAUTION:

When entering a Multi-Location Dial Plan shortened extension in a field designed for announcement extensions, certain administration end validations that are normally performed on announcement extensions are not done, and resultant warnings or submittal denials do not occur. The shortened extensions also do not appear in any display or list that shows announcement extensions. Extra care should be taken to administer the correct type of announcement for the application when assigning shortened extensions.

Name

Valid entries	Usage
up to 27-character alpha-numeric filename (no ., /, :, *, ?, <, >, \, .wav, or blanks in this field for VAL circuit packs only)	Describe the announcement message. For TN2501AP announcements, this field is required. The value in this field becomes the filename of the announcement. The **.wav** file extension, which is part of the filename stored on the circuit pack, does not appear. Do not enter **.wav** as part of the filename. Names on a single VAL circuit pack must be unique. The system checks for duplicate filenames on the same VAL circuit pack.

Port

For External Announcements (analog, Lineside T1/E1, or Aux. Trunk). Enter the necessary characters in the **aaxssyy** format (where **aa** = the cabinet number, **x** = the carrier, **ss** = the slot number, and **yy** = the circuit number).

Valid entries	Usage
01 through **44** (DEFINITY Server R) or **01** through **03** (DEFINITY CSI, SI) or **1** through **64** (S8700 or S8710 IP-Connect / S8700 or S8710 Multi-Connect)	First and second characters are the cabinet number.
A through **E**	Third character is the carrier.
0 through **20**	Fourth and fifth characters are the slot number.
00 through **32**	Six and seventh characters are the circuit number.

For example, 01A0612 is in cabinet 01, carrier A, slot 06, and circuit number (port) 12.

For Integrated Announcements (TN750 or TN2501AP (VAL) circuit pack). Enter the necessary characters in the **aaxss** format (where **aa** = the cabinet number, **x** = the carrier, and **ss** = the slot number).

Valid entries	Usage
01 through **44** (DEFINITY Server R) or **01** through **03** (DEFINITY CSI, SI) or **1** through **64** (S8700 or S8710 IP-Connect / S8700 or S8710 Multi-Connect)	First and second characters are the cabinet number.
A through **E**	Third character is the carrier.
0 through **20**	Fourth and fifth characters are the slot number.

For example, 01A06 is in cabinet 01, carrier A, and slot 06.

For G700 Media Gateway (VVAL) circuit pack. Enter the necessary characters in the **ggV9** format (where **gg** = the gateway number and **V9** = embedded announcements).

Valid entries	Usage
01 through **50**	Media gateway number.

For example, 10V9 is gateway 10.

Pr

When the **Type** is field **analog**, **ds1**, or **aux-trunk**, **N/A** appears in this field.

Valid entries	Usage
y	Enter **y** to protect the integrated announcement from being deleted or changed by any user. For VAL, after an announcement file resides on the circuit pack (recorded or FTP transfer), you can set this field to **y** to protect the file (read-only).
n	Enter **n** to allow only users with a console permission COS to change or delete an announcement. For VAL the default is **n**, meaning that the announcement file is read-write. Use this value when you initially administer the announcement.

Q (Queue)

Valid entries	Usage
y	Enter **y** to queue calls for the announcement if the **Type** field is **integrated**, **integ-rep** or **aux-trunk**. The caller is always connected to the beginning of the announcement. Enter **y** for ACD and vectoring delay announcements. Call centers should always use this option.
n	No queue and no barge-in. The caller is always connected to the beginning of the announcement. The announcement does not play if a port is not available.
b	Enter **b** to set up barge-in if the **Type** field is **integrated**, **integ-rep** or **aux-trunk**. Callers are connected to the announcement at any time while it is playing. NOTE: The same non-barge-in announcement can be played through more than one port (or all ports) of an integrated circuit pack. The initial request to play an announcement selects an available port on the board on which the announcement resides. If there are additional requests to play the announcement while it is playing on another port(s), another port is selected. If all ports are busy, new requests to play announcements go to the integrated announcement system queue (**Q** field must be **y**). Otherwise, the request to play is denied, and processing continues without the caller hearing the announcement. When a port becomes available, all queued calls (up to the platform "calls connected" limit) are connected at the same time to hear the announcement play from the beginning. A barge-in announcement starts playing when first requested and continues playing through a port, repeating until there are no more requests. Call processing simultaneously connects calls to the playing barge-in announcement. Each call remains connected until the requesting feature operation removes the call (for example, wait step times out). Barge-in type announcements never select another port to play the same announcement once it is playing on a specific port.

QLen (Queue Length)

The queue length is the number of calls that can queue for this announcement. The maximum number of queues allowed depends on your system configuration.

The **QLen** field applies if the **Q** field is **y** and the **Type** field is **analog**, ds1 or **aux-trunk**. When the **Type** field is **integrated** or **integ-rep**, **N/A** appears in this field. Integrated announcements have a pre-set queue length

.

Valid entries	Usage
The maximum number your system allows	Number of calls that can be queued for this announcement

Rt

Enter the recording rate speed (in 1000 bits/second) for integrated announcements. A different recording speed may be used for each integrated announcement. When the **Type** field is **analog**, **ds1** or **aux-trunk**, **N/A** appears in this field.

TN750 or ISSPA.

Valid entries	Usage
16	16 kbps (8 minutes and 32 seconds of announcement time per circuit pack or 1 hour and 24 minutes for 10 circuit packs for the TN750; for the ISSPA, there are 240 minutes of storage time). This rate does not provide a high-quality recording. Avaya does not recommend this for customer announcements, but it is adequate for VDN of Origin announcements.
32	32 kbps (4 minutes and 16 seconds of total announcement time for the TN750; for the ISSPA, there are 120 minutes of storage time).
64	64 kbps (for 2 minutes and 8 seconds of announcement time per circuit pack or 42 minutes for 10 circuit packs for the TN750; for the ISSPA, there are 60 minutes of storage time). This is the default for VAL.

VAL or VVAL

Valid entries	Usage
64	VAL provides 10 minutes storage as the default and 60 minutes with the **VAL Maximum Capacity** field. VVAL provides 20 minutes regardless of the setting of the **VAL Maximum Capacity** field.

TN

Valid entries	Usage
1 to **100**	Enter the Tenant Partition number, if any.

Type

Enter the type of announcement you want to assign to this extension number.

If you enter **integrated** or **integ-rep**, complete the **Q**, **Protect**, **Rate**, and **Port** fields. If you enter **analog**, **ds1-fd**, **ds1-sa**, **ds1-ops**, or **aux-trunk**, complete **QLen** (if **Q** is **y**) and **Port**.

Valid entries	Usage
analog	Use to play announcements from an external device for a specific period and hang up when finished. When the device hangs up, the caller hears a click. Connects to the server running Avaya Communication Manager through an analog port. Ringing starts playback.
analog-m	Use for continuous playing music or audio source from an external announcement device.
analog-fd	Use to play announcements from an external device for a specific period and hang up when finished. When the device hangs up, the caller hears a click. Connects to the server running Avaya Communication Manager through an analog port. Ringing starts playback. Sends forward disconnect signal to stop playback.
aux-trunk	Auxiliary trunk. Use with an external announcement device with a 4-wire "aux" interface.
aux-trk-m	Auxiliary trunk. Use with continuously playing music or audio sources that do not indicate playback is active.

1 of 2

Valid entries	Usage
ds1-fd	Assigned to DS1 ports on circuit packs. Callers do not hear a click when the device hangs up. Provides a disconnect to stop playback when the announcement is done.
ds1-ops	Callers do not hear a click when the device hangs up.
ds1-sa	Provides a disconnect to stop playback when the announcement is done. Callers do not hear a click when the device hangs up.
integrated	Stored internally on the Avaya DEFINITY or media server on a special integrated announcement circuit pack. Use for general announcements and VDN of Origin Announcements.
integ-rep	integrated repeating
	2 of 2

See Appendix A, "Recorded Announcements," in *Avaya Communication Manager Contact Center Guide to ACD Contact Centers*, for more information about the **Type** field.

ARS Toll Table

This screen assigns ARS Toll Tables used by Subnet Trunking. Use it to specify whether calls to CO codes listed on the table are toll or non-toll calls. You specify non-toll calls based on the last 2 digits of the distant-end of the trunk group.

Figure 271: ARS Toll Table screen

```
change ars toll
                              ARS TOLL TABLE: __              Page 1 of 1

                             OFFICE CODES: x00-x99
   00: y   10: y   20: y   30: y  40: y  50: y  60: y  70: y  80: y  90: y
   01: y   11: y   21: y   31: y  41: y  51: y  61: y  71: y  81: y  91: y
   02: y   12: y   22: y   32: y  42: y  52: y  62: y  72: y  82: y  92: y
   03: y   13: y   23: y   33: y  43: y  53: y  63: y  73: y  83: y  93: y
   04: y   14: y   24: y   34: y  44: y  54: y  64: y  74: y  84: y  94: y
   05: y   15: y   25: y   35: y  45: y  55: y  65: y  75: y  85: y  95: y
   06: y   16: y   26: y   36: y  46: y  56: y  66: y  76: y  86: y  96: y
   07: y   17: y   27: y   37: y  47: y  57: y  67: y  77: y  87: y  97: y
   08: y   18: y   28: y   38: y  48: y  58: y  68: y  78: y  88: y  98: y
   09: y   19: y   29: y   39: y  49: y  59: y  69: y  79: y  89: y  99: y
```

00: through 99:

These fields represent the last 2 digits of the codes within the 100-block of numbers. Designate each as a number toll or non-toll call.

Valid entries	Usage
y/n	Enter **n** to designate a non-toll CO code.

ARS TOLL TABLE

Valid entries	Usage
2 through **9**	Identify the number of the ARS Toll Table.

OFFICE CODES

Valid entries	Usage
200 to **299** through **900** to **999**	Identify the block of numbers on this screen.

Attendant Console

This screen assigns an Attendant Console to the system.

Field descriptions for page 1

Figure 272: Attendant Console screen

```
change attendant 1                                          Page   1 of   3
                            ATTENDANT CONSOLE 1

         Type: console           Name: 27 character attd cons name
    Extension: 1000             Group: 1                Auto Answer: none
 Console Type: principal           TN: 1                Data Module? y
         Port: 01C1106             COR: 1          Disp Client Redir? n
Security Code:                     COS: 1           Display Language: english
                                                   H.320 Conversion? n

DIRECT TRUNK GROUP SELECT BUTTON ASSIGNMENTS (Trunk Access Codes)
    Local Remote           Local Remote             Local Remote
 1: 9                   5:                       9:
 2: 82                  6:                      10:
 3:                     7:                      11:
 4:                     8:                      12:

HUNDREDS SELECT BUTTON ASSIGNMENTS
 1:           5:            9:           13:           17:
 2:           6:           10:           14:           18:
 3:           7:           11:           15:           19:
 4:           8:           12:           16:           20:
```

Attendant Console x

This is a display-only field when the screen is accessed using an administration command such as **add** or **change**.

Auto Answer

Valid entries	Usage
all	Entering **all** in this field indicates an incoming call to an idle attendant will be answered automatically without any action (no button presses required) by the attendant.
acd	Entering **acd** indicates only ACD split/skill calls and direct agent calls can auto answer. Non-ACD calls terminated to an attendant console with Auto Answer set to **acd** ring audibly.
none	Entering **none** causes all calls terminated to this attendant console to receive some sort of audible ringing treatment.

Console Type

Enter this console's intended use. There can only be one **night-only** or one **day/night** console in the system unless Tenant Partitioning is administered. Night Service is activated from the principal console or from the one station set per-system that has a **nite-serv** button.

Valid entries	Usage
principal	Puts the attendant console into night service.
day-only	Will not get night service calls.
night-only	Handles only night service calls.
day/night	Handles day or night service calls.

COR

Valid entries	Usage
0 through **95**	Enter the class of restriction that reflects the desired restriction.

COS

Valid entries	Usage
0 through **15**	Enter the class of service (COS) for this attendant console.

Data Module

Valid entries	Usage
y/n	Enter **y** if the console is to be connected to a data terminal via 7400B or 8400 Data Module. If **y** is entered, complete the **Data Module** screen (page 4).

Disp Client Redir

This field is administrable only if the Hospitality feature has been enabled on the **System-Parameters Customer-Options** screen. This field affects the station's display on calls originated from a station with Client Room Class of Service.

Valid entries	Usage
y	When the field is **y**, the redirection information for a call originating from a Client Room and terminating to this station displays. Note: For stations with an **audix** station type, AUDIX Voice Power ports, or ports for any other type of messaging that needs display information, this field must be **y**.
n	When the field is **n**, then for all calls originating from a Client Room (even redirected calls) that terminate to this station, this station's display will not show the redirection information. Only the client name and extension (or room, depending on what is administered on the **Hospitality** screen) displays.

Display Language

Enter the language in which you want console messages displayed.

Valid entries	Usage
English	Enter the language in which you want messages to be displayed.
French	
Italian	
Spanish	
user-defined	
Unicode	**Note:** Unicode display is only available for Unicode-supported telephones. Currently, only the 4620SW and 4610SW support Unicode display.

Extension (Optional)

Enter the extension for the individual attendant console. Individual attendant extensions allow attendants to use features that an attendant group cannot use. For example, extensions can be members of a DDC or UCD group. An individual attendant extension can have its own Class of Restriction and Class of Service.

If you give your attendants an individual extension, users can call the attendant by dialing the extension or you can assign them an abbreviated-dialing button for fast access to the attendant.

Valid entries	Usage
An unassigned extension or blank	If an extension is not assigned, the attendant can only be addressed as a member of the attendant group. If the attendant has a data module, the **Extension** field cannot be blank.

Group

Valid entries	Usage
1 to **28**	Enter the Attendant Group number.

H.320 Conversion

Allows H.320 compliant calls made to this telephone to be converted to voice-only. Because the system can handle only a limited number of conversion calls, you may need to limit the number of phones with H.320 conversion.

Valid entries	Usage
y/n	Enter **y** for H.320 compliant calls.

Name

Enter the name of this console.

Valid entries	Usage
Up to 27 alphanumeric characters	Any entry is accepted.

Port

Enter the necessary characters.

Valid entries	Usage
01 to **44** (DEFINITY R) or **01** to **03** (DEFINITY CSI, SI) or **1** to **64** (S8700 or S8710 IP-Connect / S8700 or S8710 Multi-Connect / S8300 Media Server)	First and second characters are the cabinet number.
A to **E**	Third character is the carrier.

1 of 2

Valid entries	Usage
0 to **20**	Fourth and fifth characters are the slot number.
01 to **04** (Analog TIE trunks) **01** to **31** **1** through **80** (DEFINITY R, CSI) or **1** through **10** (DEFINITY SI) or **1** through **250** (S8700 or S8710 IP-Connect / S8700 or S8710 Multi-Connect / S8300 Media Server)	Six and seventh characters are the circuit number. Gateway
V1 through **V9**	Module
01 through **31**	Circuit
ip	SoftConsole IP attendant. You also must have the **Type** field as **302B** and enter a security code. ip is allowed only if, on the **System-Parameters Customer-Options** screen, the **IP Attendant Consoles** field is **y**.
x	Indicates that there is no hardware associated with the port assignment. An individual attendant extension must be assigned in the **Extension** field.
	2 of 2

For example, 01A0612 is in cabinet 01, carrier A, slot 06, and circuit number (port) 12.

Each attendant console requires a port on a digital line circuit pack. For reliability, the attendant consoles should not be assigned to ports on the same digital line circuit pack. For example, if three attendant consoles are to be provided, assign each console to a port on three different digital line circuit pack, if possible. However, if required, all attendant consoles can be assigned to ports on the same digital line circuit pack.

Security Code

Does not apply to S8700 or S8710 IP-Connect. Enter the security code required by the SoftConsole IP attendant. The required security code length is determined by **Minimum Security Code Length** on the **Feature-Related System Parameters** screen.

TN

Valid entries	Usage
1 to **100**	Enter the Tenant Partition number.

Type

Valid entries	Usage
console	Indicates the type of attendant console being administered.
302	Use for 302B/C/D or SoftConsole IP attendant.

DIRECT TRUNK GROUP SELECT BUTTON ASSIGNMENTS (Trunk Access Codes)

Enter the trunk access codes (TACs) for local and remote servers. (There are fields for one local TAC and one remote TAC per button labeled **Local** and **Remote**.) The local TAC (1 to 4 digits) refers to a trunk group or Loudspeaker Paging zone on this server. Remote TACs are only useful in a private network (including DCS) network. The remote TAC (1 to 3 digits) refers to a trunk group on the remote server. If a remote TAC is given, then the local TAC must see a trunk group that connects directly to the remote server running Avaya Communication Manager and is also limited to 1 to 3 digits.

Avaya recommends a DCS trunk be specified as the local TAC between the local and remote servers. If the TAC specified as local between the local and remote servers is not a DCS trunk, the remote trunk cannot be monitored by the local server running Avaya Communication Manager.

Valid entries	Usage
1 to 4 digit number	Enter the trunk access codes (TACs) for local and remote servers.
***** or #	May be used as first digit

HUNDREDS SELECT BUTTON ASSIGNMENTS

Enter in the appropriate field (1 through 20), the hundreds group to be associated with a **Hundreds Group Select** button located on an optional selector console.

Valid entries	Usage
1 to 5 digit hundreds group (plus prefix, if needed)	Fields 1 through 8 are used when the selector console is a 24A-type console and fields 1 through 20 are used for a 26A-type console. Enter a hundreds group number that represents all but the last two digits of an extension number (for example, the **Hundreds Select** Button — on the selector console — for extension 3822 would be "38").

Field descriptions for page 2 (SoftConsole IP Attendant)

Figure 273: Attendant Console Data Module screen (page 2)

```
change attendant 1                                          Page 2 of 4
                                ATTENDANT
IP FEATURE OPTIONS


Remote Softphone Emergency Calls: as-on-local   Direct IP-IP Audio Connections?y
Emergency Location Ext: 1001       Always use? n          IP Audio Hairpinning?y
```

IP FEATURE OPTIONS

Always Use

This field does not apply to SCCAN wireless telephones, or to extensions administered as type **h.323**.

Valid entries	Usage
y	When this field is **y**: ● The **Remote Softphone Emegency Calls** field is hidden. A softphone can register no matter what emergency call handling settings the user has entered into the softphone. If a softphone dials 911, the **Emergency Location Extension** administered on the **Station** screen is used. The softphone's user-entered settings are ignored. ● If an IP telephone dials 911, the **Emergency Location Extension** administered on the **Station** screen is used.
n	Default value. For more information, see the description for the **Emergency Location Extension** field on the **Station** screen.

Direct IP-IP Audio Connections

Allows direct audio connections between IP endpoints.

Valid entries	Usage
y/n	Enter **y** to save on bandwidth resources and improve sound quality of voice over IP transmissions.

Emergency Location Ext

The **Emergency Location Ext** field defaults to the telephone's extension. This extension is the starting point for identifying the street address or nearby location when an emergency call is made. The entry in this field is manipulated by CAMA Numbering Format before being sent over CAMA trunks; or similarly by ISDN Numbering — Public/ Unknown before being sent over ISDN trunks. For more information about this field, see the **Usage** description for the **Remote Softphone Emergency Calls** field on the next page.

Valid entries	Usage
1 to 8 digits	Enter the Emergency Location Extension for the SoftConsole IP Attendant.

IP Audio Hairpinning

Allows IP endpoints to be connected through the server's IP circuit pack.

Valid entries	Usage
y/n	Enter **y** to allow IP endpoints to be connected through the IP circuit pack for Avaya Communication Manager in IP format, without going through the Communication Manager TDM bus.

Remote Softphone Emergency Calls

Use this field to tell Communication Manager how to handle emergency calls from the IP telephone. This field appears when the **IP Softphone** field is set to **y** on the **Station** screen.

CAUTION:

An Avaya IP endpoint can dial emergency calls (for example, 911 calls in the U.S.). It only reaches the local emergency service in the Public Safety Answering Point area where the telephone system has local trunks. Please be advised that an Avaya IP endpoint cannot dial to and connect with local emergency service when dialing from remote locations that do not have local trunks. You should not use an Avaya IP endpoint to dial emergency numbers for emergency services when dialing from remote locations. Avaya Inc. is not responsible or liable for any damages resulting from misplaced emergency calls made from an Avaya endpoint. Your use of this product indicates that you have read this advisory and agree to use an alternative telephone to dial all emergency calls from remote locations. Please contact your Avaya respresentative if you have questions about emergency calls from IP phones.

Valid entries	Usage

as-on-local	Type **as-on-local** to achieve the following results:
	• If the administrator chooses to leave the **Emergency Location Extension** fields (that correspond to this station's IP address) on the **IP Address Mapping** screen blank, the value **as-on-local** sends the extension entered in the **Emergency Location Extension** field in the **Station** screen to the Public Safety Answering Point (PSAP).
	• If the administrator populates the **IP Address Mapping** screen with emergency numbers, the value **as-on-local** functions as follows:
	  • If the **Emergency Location Extension** field in the **Attendant** screen is the same as the **Emergency Location Extension** field in the **IP Address Mapping** screen, the value **as-on-local** sends the extension to the Public Safety Answering Point (PSAP).
	  • If the **Emergency Location Extension** field in the **Attendant** screen is different from the **Emergency Location Extension** field in the **IP Address Mapping** screen, the value **as-on-local** sends the extension in the **IP Address Mapping** screen to the Public Safety Answering Point (PSAP).

block	Enter **block** to prevent the completion of emergency calls. Use this entry for users who move around but always have a circuit-switched telephone nearby, and for users who are farther away from the media server or switch than an adjacent area code served by the same 911 Tandem office. When users attempt to dial an emergency call from an IP Telephone and the call is blocked, they can dial 911 from a nearby circuit-switched telephone instead.

1 of 2

Valid entries	Usage
cesid	Enter **cesid** to allow Communication Manager to send the CESID information supplied by the IP Softphone to the PSAP. The end user enters the emergency information into the IP Softphone. Use this entry for IP Softphones with road warrior service that are near enough to the media server or switch that an emergency call routed over the it's trunk reaches the PSAP that covers the server or switch. If the media server or switch uses ISDN trunks for emergency calls, the digit string is the telephone number, provided that the number is a local direct-dial number with the local area code, at the physical location of the IP Softphone. If the media server or switch uses CAMA trunks for emergency calls, the end user enters a specific digit string for each IP Softphone location, based on advice from the local emergency response personnel.
option	Enter **option** to allow the user to select the option (extension, block, or cesid) that the user selected during registration and the IP Softphone reported. Use this entry for extensions that may be swapped back and forth between IP Softphones and a telephone with a fixed location. The user chooses between **block** and **cesid** on the softphone. A DCP or IP telephone in the office automatically selects **extension**.
	2 of 2

Field descriptions for Attendant Console Data Module screen

This page displays if the **Data Module** field on Page 1 is **y**. If this is a non-IP attendant console this is page 2 of the **Attendant** screen.

Figure 274: Attendant Console Data Module screen

```
change attendant 1                                      Page 2 of x
ATTENDANT
DATA MODULE
     Data Extension: ____          Name: _________________   BCC: 2
                                    COS: 1_
                                    COR: 1_
                ITC: restricted      TN: 1_

ABBREVIATED DIALING
 List1: _________

SPECIAL DIALING OPTION: default


ASSIGNED MEMBER (Station with a data extension button for this data module)
        Ext      Name
     1:
```

DATA MODULE

BCC

A display-only field that appears when the **ISDN-PRI** or **ISDN-BRI Trunks** field is enabled on the **System-Parameters Customer-Options** screen.

Note:

The **BCC** value is used to determine compatibility when non-ISDN facilities are connected to ISDN facilities (ISDN Interworking feature).

COR

Valid entries	Usage
0 to **95**	Enter the desired class of restriction (COR) number.

COS

Valid entries	Usage
0 to **15**	Enter the desired (COS) number to designate allowed features. See Class of Service on page 830 for additional information on the allowed features.

Data Extension

Enter the extension number assigned to the data module.

Valid entries	Usage
1 to 5-digit number	Must agree with the system's dial plan

ITC

Valid entries	Usage
restricted	
unrestricted	

Name

Enter the name of the user associated with the data module. The name is optional; it can be left blank.

TN

Enter the Tenant Partitioning number.

Valid entries	Usage
1 to **100**	

ABBREVIATED DIALING

List1

Valid entries	Usage
s	System
g	Group. If **g** is entered, a group number is also required.
p	Personal. If **p** is entered, a personal list number also is required.
e	Enhanced

SPECIAL DIALING OPTION

Valid entries	Usage
hot-line	Enter one of the dialing options that are available. This identifies the destination of all calls when this data module originates calls.
default	

HOT LINE DESTINATION — Abbreviated Dialing Dial Code

Only displays when the **Special Dialing Option** field is **hot-line** or **default** (S8700 or S8710 IP-Connect only). The associated AD number is dialed when the user goes off-hook on a Data Hot Line call.

Hot Line Service allows single-line telephone users, by simply lifting the handset, to automatically place a call to a preassigned destination (extension, telephone number, or feature access code).

The Hot Line Service destination number is stored in an Abbreviated Dialing List.

A Direct Department Calling (DDC), a Uniform Call Distribution (UCD), a Terminating Extension Group (TEG) extension, or any individual extension within a group can be a Hot Line Service destination. Also, any extension within a DDC group, UDC group, or TEG can have Hot Line Service assigned.

Use Hot Line Service when very fast service is required and when you use a telephone only for accessing a certain facility. Loudspeaker Paging Access can be used with Hot Line Service to provide automatic access to paging equipment.

Valid entries	Usage
A dial code	Within the range of the abbreviated dialing list type

DEFAULT DIALING Abbreviated Dialing Dial Code

The associated AD number is dialed when the user goes off-hook and enters a carriage return following the "DIAL" prompt. The data call originator also can perform data terminal dialing by specifying a dial string that may or may not contain alphanumeric names. Only displays when the **Special Dialing Option** field is **default**.

Valid entries	Usage
A dial code	Within the range of the abbreviated dialing list type

Ext

This is the extension number of a previously administered user who has an associated **Data Extension** button and who will share the use of the module.

Name

Contains the name assigned to the above extension number.

Field descriptions for page 3

Figure 275: Attendant Console screen

```
change attendant 1                                          Page 3 of 4
                                ATTENDANT CONSOLE
FEATURE BUTTON ASSIGNMENTS
  1: split_____                          13: ___________
  2: ___________                         14: ___________
  3: ___________                         15: ___________
  4: ___________                         16: ___________
  5: ___________                         17: ___________
  6: hold _____ *                        18: ___________
  7: ___________                         19: forced-rel
  8: aux-work      RC:    Grp:           20: ___________
  9: ___________                         21: ___________
 10: ___________                         22: ___________
 11: ___________                         23: night-serv *
 12: ___________                         24: pos-busy__ *
```

If this is a non-IP attendant console this is page 3 of the **Attendant** screen.

FEATURE BUTTON ASSIGNMENTS

Enter the feature buttons from Attendant Console Feature Buttons on page 166 that you want to assign to the attendant console. The fixed buttons that cannot be changed (that is, split and forced release) are shown on the screen. The **hold**, **night-serv**, and **pos-busy** buttons are shown in the system default locations. These buttons can be administered elsewhere on the screen. The following table provides descriptions of feature buttons that are unique to the attendant console. See the Attendant Console Feature Buttons and to the Telephone feature Buttons sections for more information about the buttons.

Valid entries	Usage
Audible Tones On/Off	
cw-ringoff	Call waiting ringer off; turns on/off the audible tone for call waiting on attendant console (1 per console).
in-ringoff	Incoming call ringer off; turns on/off the audible tone for incoming call ringer (1 per console).
re-ringoff	Timed reminder ringer off; turns on/off the audible tone for timer reminder ringer (1 per console).

Valid entries	Usage
alt-frl	Alternate FRL. Alternate facility restriction level; allows the attendant to activate or deactivate the AFRL feature. When activated, this allows the originating device (lines or trunks) to use an alternate set of the facility restriction levels to originate a call (1 per console).
Attendant Control of Trunk Group Access	
act-tr-grp	Activate trunk group access; allows the attendant to control a trunk group. All calls going to the trunks are routed to the attendant (1 per console).
deact-tr-g	Deactivate trunk group access; allows the attendant to release control of a trunk group (1 per console).
class-rstr	Display Class of Restriction. Used to display the COR associated with a call (1 per console).
em-acc-att	Emergency Access to the Attendant. The associated status lamp is flashed when there are one or more calls on the emergency attendant queue (1 per console).
hold	Hold. When the **Hold** button is pressed while the attendant is active on a loop, the party on the loop is put on hold and the **call type** button associated with the loop is lit (1 per console).
pos-busy	Position Busy. When this button is pushed, the attendant is put into position busy mode, the "Pos Avail" light is turned off, and the light associated with the **pos-busy** button is lit. Pushing the **pos-busy** button a second time takes the console out of "position busy" mode, turns on the "Pos Avail" light and turns off the light associated with the **pos-busy** button. If the **pos-busy** button is administered on a 2-LED button, the top LED flashes when the last attendant goes into "Position Busy" mode. Otherwise, if the button has only one LED, the single LED associated with the **pos-busy** button flashes (1 per console).
serial-cal	Serial Call. This button allows the attendant-extended calls to return to the same attendant if the trunk remains off-hook (1 per console).
override	Attendant Override. This button enables the attendant to override diversion features such as, Call Forwarding, Call Coverage, and so on (1 per console).
intrusion	Call Offer. Depression of this button allows the attendant to extend a call when the called party is active on another call (1 per console).
dont-split	Don’t Split. This button allows the attendant to not split away a call when dialing (1 per console).

Valid entries	Usage
vis	Visually Impaired Attendant Service (**vis**) — This button activates visually impaired service for the attendant. When this service is activated, the attendant can listen to console status or messages by pressing buttons that have been translated as follows: • "con-stat" repeats the console status. • "display" calls out display contents. • "dtgs-stat" calls out the DTGS status. • "last-mess" repeats the last message. • "last-op" calls out the last operation.
Trunk Group Select — In addition to the 12 Direct Trunk Group Selection (DTGS) Button Assignments on Field descriptions for page 1, up to 12 single lamp **DTGS** buttons can be administered on this page. The status lamp associated with the feature button is used to monitor the busy/idle status of the trunk. Trunk groups administered on these buttons cannot be controlled using **Attendant Control of Trunk Group Select** buttons. The single lamp **DTGS** buttons can be administered as follows:	
local-tgs	Local trunk group select; allows the attendant to access trunk groups on the local server running Avaya Communication Manager (combination of 12 local-tgs/remote-tgs per console).
remote-tgs	Remote trunk group select; allows the attendant to access trunk groups on a remote server running Avaya Communication Manager (combination of 12 local-tgs/remote-tgs per console).
hundrd-sel	Hundreds group select; in addition to the fixed **HGS** buttons on Field descriptions for page 1, a user can administer hundreds group select feature buttons on this page. When a feature button is administered as "hundrd-sel," a subfield appears that must then be administered in the same manner as the fixed **HGS** button fields (a 1 to 3 digit hundreds group plus prefix, if needed). Administered "hundrd-sel" feature buttons operate in the same manner as fixed **HGS** buttons. The total number of hundreds group select buttons (fixed and administered) allowed on a console is 20. Thus, if all 20 fixed **HGS** buttons have been administered, no "hundrd-sel" feature buttons can be administered. Note: If no fixed **HGS** buttons are administered, 19 "hundrd-sel" feature buttons are available. This is because 5 of the 24 feature buttons must be used for required feature buttons (**hold**, **pos-busy**, **night-serv**, **forced-rel**, and **split**)
group-disp	Group Display. Allows the attendant to see a display of extensions currently being tracked on the DXS module.
group-sel	Group Select. Allows the attendant to select a specific group of hundreds by dialing the first 2 or 3 digits of the hundreds group.

3 of 4

Valid entries	Usage
Attendant Room Status	
occ-rooms	Occupied rooms; allows the attendant to see which rooms are occupied.
maid-stat	Maid status; allows the attendant to see which rooms are in one of six specified states.
vu-display	VuStats (**vu-display**) — This button allows users with display telephones and attendants to turn on the VuStats display. The limit to the number of VuStats feature buttons you can administer depends on how many feature buttons are available on the attendant console you are administering. The system is designed to allow you to set up a separate VuStats display format for each feature button. Therefore, agents can change the type of measurements on their display by selecting a different VuStats feature button.
	4 of 4

- If 12 HGS buttons are assigned on field descriptions for page 2, Avaya recommends that the **night**, **pos-busy**, and **hold** buttons be reassigned to locations 20, 21, and 3, respectively. The **HGS** buttons should then be assigned to the right-most three columns, as required.

Field descriptions for page 4

Figure 276: Attendant Console screen

```
change attendant 1                                          Page 4 of 4
                                ATTENDANT CONSOLE
DISPLAY BUTTON ASSIGNMENTS
  1: normal____                                  5: delete-msg
  2: inspect___                                  6: call-disp_
  3: cov-msg-rt                                  7: date-time_
  4: next______                                  8: timer_____
```

- **DISPLAY MODULE BUTTON ASSIGNMENTS** - Display-type buttons obtain display functions on the associated alphanumeric display. These buttons are noted as [display button] in the **Feature** or **Function** column on the table. Also, several feature buttons can be administered so that their associated status lamps can be used to provide visual indications of the associated feature or function. In some cases, the button itself is not operational. These buttons are noted as [status lamp]. If a Call Cover Msg Rt (**cov-msg-rt**) button is assigned, a Leave Word Calling Delete Msg (**delete-msg**) button and a Next (**next**) button must also be assigned.

Audix-MSA Node Names

Figure 277: Audix-MSA Node Names screen

```
change node-names audix-msa                                    Page 1 of x

                        AUDIX-MSA NODE NAMES

 Audix Name        IP Address     MSA Names      IP Address
 audixA_           __.__.__.__    ________       __.__.__.__
 audixB_           __.__.__.__    ________       __.__.__.__
 ________          __.__.__.__    ________       __.__.__.__
 ________          __.__.__.__    ________       __.__.__.__
 ________          __.__.__.__    ________       __.__.__.__
 ________          __.__.__.__    ________       __.__.__.__
 ________          __.__.__.__    ________       __.__.__.__
 ________          __.__.__.__    ________       __.__.__.__
```

Audix Names

Identifies the name of the AUDIX node.

Valid entries	Usage
1 to 7 character string	Used as a label for the associated IP address. The node names must be unique on each server running Avaya Communication Manager.

IP Address

The IP address associated with the node name. This field can be blank for X.25 connections.

MSA Names

Identifies the name of the MSA node.

Valid entries	Usage
1 to 7 character string	Used as a label for the associated IP address. The MSA names must be unique on each server running Avaya Communication Manager.

Authorization Code — COR Mapping

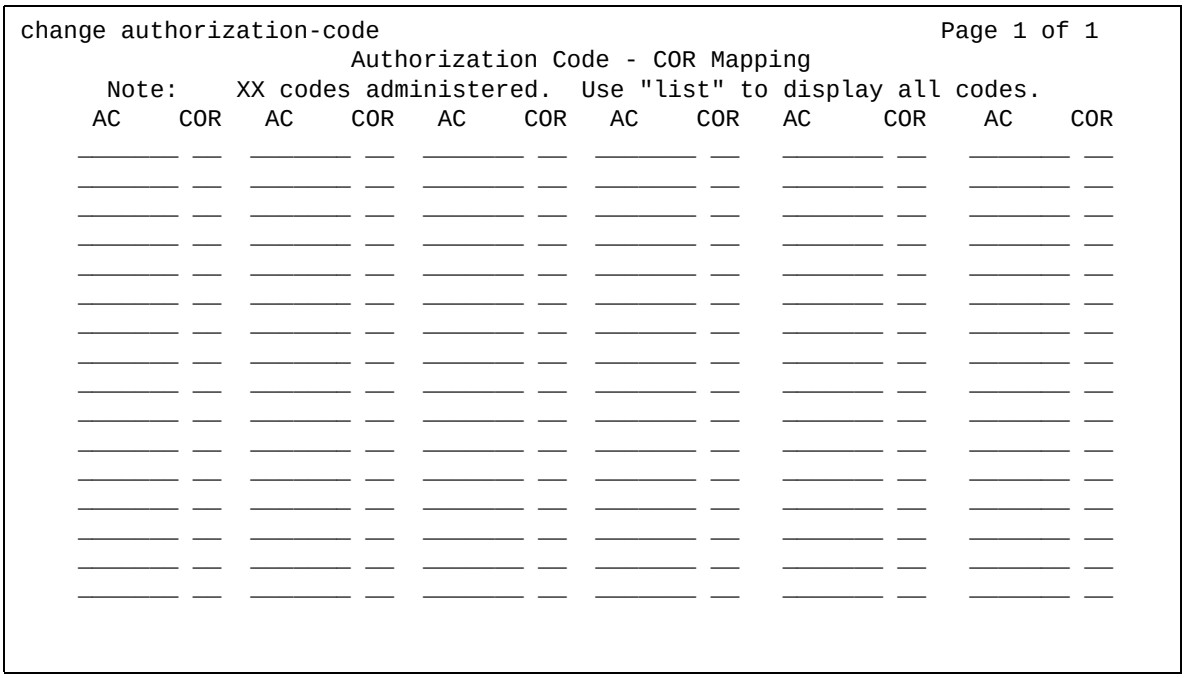
You use this screen to assign authorization codes and the class of restriction (COR) that is associated with a given authorization code. See "Authorization Codes" and "Class of Restriction" in *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205*,* for more information on how Authorization Codes work with COR.

To maximize the security of your system:

- Administer authorization codes to the maximum length allowed by the system
- Create random (nonconsecutive) authorization codes
- Change authorization codes at least quarterly
- Deactivate authorization codes immediately if a user leaves the company or changes assignments
- Assign each authorization code the minimum level of calling permissions required

Field descriptions for page 1

Figure 278: Authorization Code - COR Mapping screen



AC

Valid entries	Usage
Any combination of between 4 and 13 digits	The number of digits must agree with the number assigned to the **Authorization Code Length** field on the **Feature-Related System Parameters** screen. To enhance system security, choose Authorization Codes of 13 random digits.

COR

Valid entries	Usage
0 to **95**	When a user dials the associated authorization code, this is the COR that the telephone or other facility will assume for that call.

Number of Codes Administered

Displays the number of Authorization Codes already administered using the **Authorization Codes** screen. There is a maximum number of authorization codes that you can use. To find out what this number is for your system, type **`display capacity`**, and page down to find the authorization code information. For details on the **System Capacity** screen, see *Maintenance Commands for Avaya Communication Manager, Media Gateways and Servers*, 03-300191.

Best Service Routing

This screen administers the routing numbers for each location associated with a particular application. This allows the Avaya DEFINITY server or media server to compare specified skills, identify the skill that will provide the best service to a call, and deliver the call to that resource.

For information on setting up Best Service Routing (BSR), see *Avaya Communication Manager Contact Center Guide to ACD Contact Centers*.

Field descriptions for page 1

Figure 279: Best Service Routing screen

```
change best-service-routing 1                                  Page 1 of 15

                            BEST SERVICE ROUTING

 Number: 1      Name: ARS               Maximum Suppression Time: 30     Lock? n

Num   Location Name     Switch Node    Status Poll VDN   Interflow VDN  Net Redir?
1     st10     auto                    95022011          3035552121         y
4     st10     auto                    95022014          3035551110         n
```

Interflow VDN

When a given remote Avaya server is the best available, the origin Avaya server interflows the call to this vector on the remote server. Each remote Avaya server in a given application has to have a dedicated interflow server.

Valid entries	Usage
0 to **9**, *****, #, **~p** (pause) **~w/~W** (wait) **~m** (mark) **~s** (suppress) or blank (DEFINITY R, SI, CSI)	

Location Name

Indicates the location.

Valid entries	Usage
Up to 15 alphanumeric characters. (DEFINITY R, SI, CSI)	

Lock

Indicates whether this application is locked.

Valid entries	Usage
y/n (DEFINITY R, SI, CSI)	Set to **y** to prevent this application from being sent to Call Management System (CMS).

Maximum Suppression Time

Prevents callers from connecting to a VDN within a certain time period after receiving a busy signal.

Valid entries	Usage
0 to **60** (DEFINITY R, SI, CSI)	Enter time in seconds.

Name

Contains the name assigned to the BSR number.

Valid entries	Usage
Up to 15 alphanumeric characters. (DEFINITY R, SI, CSI)	Assign a descriptive name for the physical location.

Net Redir

Valid entries	Usage
y/n (DEFINITY R, SI, CSI)	Set to y for each to location to which calls are to be redirected using Network Call Redirection. For more information, see *Avaya MultiVantage Call Center Software Release 11 — Network Call Redirection.*

Num

This field corresponds to the "consider location x" step from the **Call Vector** screen.

Valid entries	Usage
1 to **255** (DEFINITY R, SI, CSI)	Enter the number.

Number

This display-only field corresponds to the **BSR Application** field on the **Vector Directory Number** screen.

Status Poll VDN

This field specifies the AAR or ARS pattern that routes over an IP trunk. The status poll vector on the remote Avaya server compares resources on that server and replies to the origin server with information on the best of these. Each remote Avaya server in a given application has to have a dedicated status poll vector.

Valid entries	Usage
0 to **9**, *****, **#**, **~p** (pause) **~w/~W** (wait) **~m** (mark) **~s** (suppress) or blank (DEFINITY R, SI, CSI)	Specify the AAR or ARS pattern that routes over an IP trunk

Switch Node

Enter a number unique to the switch in a network of switches.

Valid entries	Usage
1 to **32767** or blank (DEFINITY R, SI, CSI)	This number is an important part of the UCID tag and must be unique to the server running Avaya Communication Manager.

Bulletin Board

Use the bulletin board to post and receive information. There are three pages of message space within the bulletin board. The first page has 19 lines, but you can only enter text on lines 11 to 19. The first 10 lines on page 1 are for high-priority messages from Avaya personnel and are noted with an asterisk (*). The second and third pages each have 20 lines, and you can enter text on any line. The system automatically enters the date the message was posted or last changed to the right of each message line.

You can enter up to 40 characters of text per line. You can also enter one blank line. If you enter more than one blank line, the system consolidates them and displays only one. The system also deletes any blank line if it is line one of any page. You cannot indent text on the bulletin board. The TAB key moves the cursor to the next line.

Field descriptions for page 1

Figure 280: Sample Bulletin Board screen

```
change bulletin-board                                      Page 1 of 3

             Message (* indicates high-priority)                 Date
          *Avaya is in the process of                          03/02/93
          *investigating your trunk lockup problem.            03/02/93
          *The Bulletin Board will be updated as               03/02/93
          *we find information.                                03/02/93
          *  We have identified the problem.                   03/04/93
          *The trunk you added does not provide                03/04/93
          *disconnect supervision. However, the                03/04/93
          *trunk group was administered as such.               03/04/93
          *Please call Pat J. for details.                     03/04/93

          We recently added a new trunk group (14)             03/02/93
          and have had many of the members getting             03/02/93
          locked up.                                           03/02/93
          We see the error - thanks for checking.              03/05/93
```

Date

This display-only field contains the date the information was entered or last changed.

Lines 1 through 10

These lines are reserved for high priority messages and are noted with an asterisk (*) in the first column on the left. If you have an init or inads login you can enter high-priority information to trigger the high-priority message at login time.

Valid entries	Usage
A to **Z** **a** to **z** Blank **0** to **9** !@#$%^&*()_+-=[]{}|\'~;:',"<.>/?	Enter any information.

Lines 11 through 19

These lines can be used by anyone with access.

Valid entries	Usage
A to **Z** **a** to **z** Blank **0** to **9** !@#$%^&*()_+-=[]{}|\'~;:',"<.>/?	Enter any information.

Field descriptions for pages 2 and 3

Date

This display only field contains the date the information was entered or last changed.

Lines 1 through 20

These lines can be used by anyone with access.

Valid entries	Usage
A to **Z** **a** to **z** Blank **0** to **9** !@#$%^&*()_+-=[]{}|\'~;:',"<.>/?	Enter any information.

Call Vector

This screen programs a series of commands that specify how to handle calls directed to a Vector Directory Number (VDN). See the *Avaya Communication Manager Contact Center Call Vectoring and Expert Agent Selection (EAS) Guide* for additional information.

Field descriptions for page 1

Figure 281: Call Vector screen

```
 change vector 129                                          Page 1 of 3
                                CALL VECTOR
   Number: 129              Name: ________________

Multimedia? n     Attendant Vectoring? n     Meet-me Conf? y           Lock? n
     Basic? y  EAS? n   G3V4 Enhanced? y    ANI/II-Digits? y   ASAI Routing? n
 Prompting? y  LAI? n  G3V4 Adv Route? y    CINFO? y   BSR? n      Holidays? n

  01 ____________
  02 ____________
  03 ____________
  04 ____________
  05 ____________
  06 ____________
  07 ____________
  08 ____________
  09 ____________
  10 ____________
  11 ____________
```

Field descriptions for page 2

Figure 282: Call Vector screen

```
change vector 129                                        Page 2 of 3
                                CALL VECTOR

 12 _____________
 13 _____________
 14 _____________
 15 _____________
 16 _____________
 17 _____________
 18 _____________
 19 _____________
 20 _____________
 21 _____________
 22 _____________
```

Field descriptions for page 3

Figure 283: Call Vector screen

```
change vector 129                                        Page 3 of 3
                                CALL VECTOR

 23 _____________
 24 _____________
 25 _____________
 26 _____________
 27 _____________
 28 _____________
 29 _____________
 30 _____________
 31 _____________
 32 _____________
```

01 through XX

Enter vector commands as required (up to the maximum allowed in your configuration). For more information, see *Avaya Communication Manager Contact Center Call Vectoring and Expert Agent Selection (EAS) Guide*

Valid entries	Usage
adjunct	Causes a message to be sent to an adjunct requesting routing instructions based on the CTI link number.
announcement	Provides the caller with a recorded announcement.
busy	Gives the caller a busy signal and causes termination of vector processing.
check	Checks the status of a split (skill) for possible termination of the call to that split (skill).
collect	Allows the user to enter up to 16 digits from a touch-tone telephone, or allows the vector to retrieve Caller Information Forwarding (CINFO) digits from the network.
consider	Defines the resource (split, skill, or location) that is checked as part of a Best Service Routing (BSR) consider series and obtains the data BSR uses to compare resources.
converse-on	Delivers a call to a converse split (skill) and activates a voice response script that is housed within a Voice Response Unit (VRU).
disconnect	Ends treatment of a call and removes the call from the server running Avaya Communication Manager. Also allows the optional assignment of an announcement that will play immediately before the disconnect.
goto	Allows conditional or unconditional movement (branching) to a preceding or subsequent step in the vector.
messaging	Allows the caller to leave a message for the specified extension or the active or latest VDN extension.
queue-to	Unconditionally queues a call to a split or skill and assigns a queueing priority level to the call in case all agents are busy.

1 of 2

Valid entries	Usage
reply-best	Used only in status poll vectors in multi-site Best Service Routing applications, where it "returns" best data for its location to the primary vector on the origin server.
route-to	Routes calls either to a destination that is specified by digits collected from the caller or an adjunct (route-to digits), or routes calls to the destination specified by the administered digit string (route-to number).
stop	Halts the processing of any subsequent vector steps.
wait-time	Delays the processing of the next vector step if a specified delay time is included in the command's syntax. Also provides feedback (in the form of silence, ringback, or music) to the caller while the call advances in queue.

2 of 2

ANI/II-Digits

A display-only field indicating whether you can use ANI and II-Digits Vector Routing Commands. ANI/II-Digits Routing requires that the **G3V4 Enhanced** field be **y**.

ASAI Routing

A display-only field indicating whether, on the **System-Parameters Customer-Options** screen, the **CallVisor Adjunct/Switch Applications Interface (ASAI Link Core Capabilities)** field is **y**.

Attendant Vectoring

This field appears only if, on the **System-Parameters Customer-Options** screen, the **Attendant Vectoring** field is **y**. **Attendant Vectoring** and **Meet-me Conference** cannot be enabled at the same time. This field does not appear for S8700 or S8710 IP-Connect.

Valid entries	Usage
y	Enter **y** so the vector is an attendant vector. Default when, on the **System-Parameters Customer-Options** screen, the **Vectoring (Basic)** and **Vectoring (Prompting)** fields are **y**.
n	Default when, on the **System-Parameters Customer-Options** screen, the **Vectoring (Basic)** and **Vectoring (Prompting)** fields are **n**. No changes allowed to this field.

Basic

A display-only field indicating whether, on the **System-Parameters Customer-Options** screen, the **Vectoring (Basic)** field is **y**.

BSR

A display-only field indicating that on the **System-Parameters Customer-Options** screen, the **Vectoring (Best Service Routing)** field is **y**. Thus, you can use BSR commands and command elements in your vectors. An **n** indicates that the BSR option is not enabled.

CINFO

A display-only field indicating whether, on the **System-Parameters Customer-Options** screen, the **Vectoring (CINFO)** field is **y**.

EAS

A display-only field indicating whether, on the **System-Parameters Customer-Options** screen, the **Expert Agent Selection (EAS)** field is **y**.

Note:

When **Expert Agent Selection (EAS)** field is **y**, the help messages and error messages associated with this screen will reflect a terminology change from "Split" to "Skill." In addition, the vector commands entered also will be affected by this terminology change (for example, *check backup split* becomes *check backup skill* when EAS is enabled).

G3V4 Adv Route

A display-only field indicating whether you can use the G3V4 Advanced Vector Routing commands.

G3V4 Enhanced

A display-only field indicating whether you can use G3V4 Enhanced Vector Routing commands and features.

Holidays

A display-only field that appears when, on the **System-Parameters Customer-Options** screen, the **Vectoring (Holidays)** field is **y**.

LAI

A display-only field indicating whether **Look-Ahead Interflow** is enabled.

Lock

This field controls access to the vector from Avaya CentreVu products.

Note:

Always lock vectors that contain secure information (for example, access codes).

Valid entries	Usage
y	You do not want this vector to be accessible to these client programs. Locked vectors can only appear and be administered through the SAT or a terminal emulator. If **Meet-me Conference** is **y**, the **Lock** field also must be **y**.
n	Gives CentreVu CMS and CentreVu Control Center users the ability to administer this vector from these client programs.

Meet-me Conf

This field appears only if, on the **System-Parameters Customer-Options** screen, the **Enhanced Conferencing** field is **y**. This field designates the VDN as a Meet-me Conference VDN.

Valid entries	Usage
y/n	Enter **y** to enable Meet-me Conference for this vector. If **Meet-me Conference** is **y**, the **Lock** field also must be **y**. When the **Lock** field is **y**, the vector cannot be changed by adjunct vectoring programs such as Visual Vectors. **Attendant Vectoring** and **Meet-me Conference** cannot be enabled at the same time.

Multimedia

Indicates whether the vector should receive early answer treatment for multimedia calls. This only applies if the **Multimedia Call Handling** field is **y**. This field does not appear for S8700 or S8710 IP-Connect.

Valid entries	Usage
y/n	If you expect this vector to receive multimedia calls, set this field to **y**. If this value is **y**, the call is considered to be answered at the start of vector processing, and billing for the call starts at that time. See Managing MultiMedia Calling on page 289 for more information.

Name

Represents the vector name.

Valid entries	Usage
Up to 27 alphanumeric characters. Up to 15 alphanumeric characters (for S8300 Media Server, S8700 or S8710 IP-Connect only)	This is an optional field. If ~r can be used to activate Network Call Redirection if, on the **System-Parameters Customer-Options** screen, the **ISDN Network Call Redirection** field is **y**.

Number

A display-only field when the screen is accessed using a **change** or **display** administration command.

Prompting

A display-only field indicating whether, on the **System-Parameters Customer-Options** screen, the **Vectoring (Prompting)** field is **y**.

CAMA Numbering Format

This screen administers the Centralized Automatic Message Accounting (CAMA) trunks and provides Caller's Emergency Service Identification (CESID) information to the local community's Enhanced 911 system through the local tandem office.

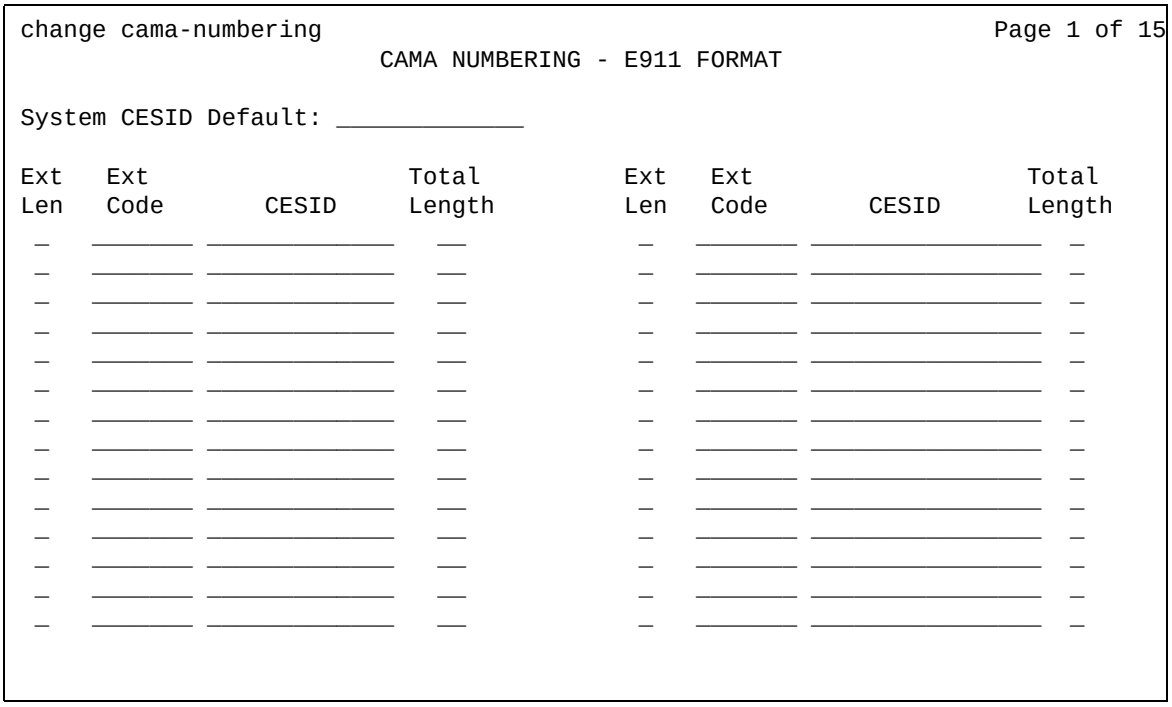
This screen provides the CESID format by extension number or number blocks. This allows for multiple CESID formats to be sent over multiple CAMA trunk groups allowing for mixed station numbering plans and some limited conversion from non-DID to DID numbers typically required by the Private Switch/Automatic Location Interface (PS/ALI) database.

The default CESID defines the CESID for all extensions that are not defined in the **Ext Code** field.

There are 446 CESID entries over 15 pages. The first page contains the Default CESID and 26 extensions to CESID entries. The second through fifteenth pages each contain 30 extensions to CESID entries.

Field descriptions for page 1

Figure 284: CAMA Numbering Format screen



```
change cama-numbering                                              Page 1 of 15
                          CAMA NUMBERING - E911 FORMAT

System CESID Default: _____________

Ext   Ext                  Total          Ext   Ext                   Total
Len   Code       CESID     Length         Len   Code       CESID      Length
 _   _______ _____________   __           _   _______ ________________  _
 _   _______ _____________   __           _   _______ ________________  _
 _   _______ _____________   __           _   _______ ________________  _
 _   _______ _____________   __           _   _______ ________________  _
 _   _______ _____________   __           _   _______ ________________  _
 _   _______ _____________   __           _   _______ ________________  _
 _   _______ _____________   __           _   _______ ________________  _
 _   _______ _____________   __           _   _______ ________________  _
 _   _______ _____________   __           _   _______ ________________  _
 _   _______ _____________   __           _   _______ ________________  _
 _   _______ _____________   __           _   _______ ________________  _
 _   _______ _____________   __           _   _______ ________________  _
 _   _______ _____________   __           _   _______ ________________  _
 _   _______ _____________   __           _   _______ ________________  _
 _   _______ _____________   __           _   _______ ________________  _
```

CESID

Valid entries	Usage
1 to 11 digits or blank 1 to 16 digits or blank (S8300 Media Server, S8700 or S8710 IP-Connect)	Enter the number that will be used to identify the calling terminal within an emergency service system. This field may represent a prefix to an extension or the entire CESID.

Ext Code

Valid entries	Usage
Up to 5 digits or blank Up to 7 digits or blank (S8300 Media Server, S8700 or S8710 IP-Connect)	Enter the leading digits or all of the digits in the extension for the specified CESID. If the extension length is greater than the number of digits in the extension code, the extension code will be interpreted as a block of digits. For example, if the extension length is 4 and the extension code is 11, the CESID will serve extensions 1100 through 1199. The Ext Code 11 is for a DID block. An Ext Code of 126 might point a non-DID block to a nearby DID extension 5241666.

Ext Len

Valid entries	Usage
1 to **7** or blank	Enter the number of digits in the extension.

System CESID Default

Valid entries	Usage
1 to 16 digits	Enter a default CESID. This number will be sent over the CAMA trunk if the **Ext Code** field does not have an entry.

Total Length

Valid entries	Usage
1 to **16** or blank	Enter the total number of digits to send.

CDR System Parameters

Use the **Call Detail Recording (CDR) System Parameters** screen to set parameters for the types of calls you want to record and how to format the information. You can use CDR records to determine call costs, diagnose problems, detect abuse, and optimize your network.

Field descriptions for page 1

Figure 285: CDR System Parameters screen

```
change system-parameters cdr                                  Page   1 of   1
                           CDR SYSTEM PARAMETERS

 Node Number (Local PBX ID):                      CDR Date Format: month/day
      Primary Output Format: printer       Primary Output Endpoint: CDR1
    Secondary Output Format:
           Use ISDN Layouts? n                        EIA Device Bit Rate: 9600
       Use Enhanced Formats? n      Condition Code 'T' for Redirected Calls? n
Modified Circuit ID Display? n                   Remove # From Called Number?n
                  Record Outgoing Calls Only? y            Intra-switch CDR? n
  Suppress CDR for Ineffective Call Attempts? y          CDR Call Splitting? y
      Disconnect Information in Place of FRL? n    Attendant Call Recording? y
                                                     Interworking Feat-flag? n
 Force Entry of Acct Code for Calls Marked on Toll Analysis Form? n
                                  Calls to Hunt Group - Record: member-ext
Record Called Vector Directory Number Instead of Group or Member? n
Record Agent ID on Incoming? n             ecord Agent ID on Outgoing?n       Inc
Trk Call Splitting? n
  Record Non-Call-Assoc TSC? n
      Record Call-Assoc TSC? n   Digits to Record for Outgoing Calls: dialed
   Privacy - Digits to Hide: 0               CDR Account Code Length: 4
```

Call Record Handling Option

Displays on DEFINITY R only. Used to control call routing when new calls come in, the CDR link is down, and the buffer is filled.

Note:

Changing this field from the default **warning** may cause ACD & vector calls that are measured by CDR to be redirected.

Valid entries	Usage
reorder	Enter **reorder** to block calls with reorder tone when the buffer is full. This applies to all calls. If you choose this option, no one will be able to make or receive calls if CDR is unable to record them.
warning	Enter **warning** to stop call recording when the buffer is full. *This generates a minor alarm.*
attendant	Enter **attendant** to route all calls to the attendant as non-CDR calls.

Calls to Hunt Group — Record

Valid entries	Usage
member-ext	Enter **member-ext** to record the extension of the telephone or data terminal where the call terminated.
group-ext	Enter **group-ext** to record the extension that was dialed.

CDR Account Code Length

Valid entries	Usage
1 to **15**	Enter the number of digits to record when a user enters an account code. For some record formats, a long account code overwrites spaces on the record that are usually assigned to other fields.

CDR Date Format

Use this field to select the format for the date stamp that begins each new day of call records.

Valid entries	Usage
month/day **day/month**	Choose the format that is most appropriate for your situation. If your company has many different sites, you may need to use the same format as the other locations.

Condition Code ‘T’ for Redirected Calls

You can elect to identify CDR records of calls that have been redirected automatically off the server running Avaya Communication Manager.

Valid entries	Usage
y	The Condition Code of both CDR records for the call will be ‘T.’
n	The Condition Codes normally associated with the **Record Outgoing Calls Only** field are generated.

Digits to Record for Outgoing Calls

Valid entries	Usage
dialed	Use **dialed** to record the digits a user actually dials.
outpulsed	Use **outpulsed** to record the digits that Communication Manager actually sends out over the trunk, *including any additions or deletions that take place during routing*.

Disconnect Information in Place of FRL

Valid entries	Usage
y	Enter **y** to replace the **Facility Restriction Level (FRL)** field with information about why a call disconnects. You can use this information to isolate problems between the DEFINITY R and the telephone network.
n	Enter **n** to record the call’s FRL.

EIA Device Bit Rate

Applies to DEFINITY SI only. Displays if either **Primary** or **Secondary Output Format** field is **eia**.

Valid entries	Usage
300 **1200** **2400** **4800** **9600**	Enter the baud rate of the CDR device connected to the EIA port.

Force Entry of Acct Code for Calls Marked on Toll Analysis Form

Specifies whether an account code will be required when making a toll call. This will not necessarily be all chargeable calls and it may even include some non-chargeable calls. See Forcing Users to Enter Account Codes on page 597 for more information.

Valid entries	Usage
y	Enter **y** to deny all toll calls unless the user dials an account code. Forced Entry of Account Codes must be **y** on the **System-Parameters Customer-Options** screen.
n	Enter **y** to allow calls without an account code. *This does not override other calling restrictions.*

Inc Attd Call Record

Appears when **Inc Trk Call Splitting** is **y**.

Valid entries	Usage
y/n	Enter **y** to enable separate recording of attendant portions of outgoing calls that are transferred or conferenced.

Inc Trk Call Splitting

Appears when the **Record Outgoing Calls Only** field on the **System Parameters CDR** screen is **n**.

Valid entries	Usage
y/n	Enter **y** to create separate records for each portion of incoming calls that are transferred or conferenced.

Interworking Feat-flag

See "Call Detail Recording" in *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205, for more information.

Valid entries	Usage
y	Enter **y** if you want the feature flag to indicate interworked outgoing ISDN calls. *An interworked call is one that passed through more than one ISDN node.*
n	Enter **n** if you want the feature flag to indicate no answer supervision for interworked calls.

Intra-Switch CDR

Valid entries	Usage
y/n	Enter **y** to record calls within Avaya Communication Manager. If you choose this option, you must complete the **Intraswitch CDR** screen to indicate which extensions should be monitored.

Modified Circuit ID Display

This affects the "printer," "teleseer," and "59-character" output formats.

Valid entries	Usage
y	Enter **y** to display the circuit ID in its actual format (100's, 10's, units). For example, circuit ID 123 displays as 123. *You might need to verify that your output device can accept this format.*
n	Enter **n** to display the circuit ID in its default format (10's, units, 100's). For example, circuit ID 123 appears as 231.

Node Number (Local PBX ID)

A display-only field indicating the DCS switch node number in a network of switches.

Outg Attd Call Record

Only appears if **Outg Trk Call Splitting** is **y**.

Valid entries	Usage
y/n	Enter **y** to enable separate recording of attendant portions of outgoing calls that are transferred or conferenced.

Outg Trk Call Splitting

See "Call Splitting" in *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205, for more information.

Valid entries	Usage
y/n	Enter **y** to create separate records for each portion of outgoing calls that are transferred or conferenced.

Primary Output Endpoint

This field determines where the server running Avaya Communication Manager sends the CDR records, and is required if you specify a Primary Output Format.

Valid entries	Usage
eia	If you use the EIA port to connect the CDR device, enter **eia**. This is not a valid option on DEFINITY R systems.
Extension number	This is the extension of the data module (if used) that links the primary output device to the server running Avaya Communication Manager.
CDR1, CDR2	Use this value if the CDR device is connected over a TCP/IP link, and this link is defined as either CDR1 or CDR2 on the **IP Services** screen.

Primary Output Format

Controls the format of the call records sent to the primary output device.

Valid entries	Usage
customized	Use this option if you have special call accounting needs that standard record formats do not accommodate. If you use a customized record format, you need to have call accounting software that is also customized to receive these records. Consult with your call accounting vendor before using this option.
printer	Use **printer** if you are sending the call detail records to a printer rather than to a record collection or call accounting system.
59-char **expanded** **lsu** **lsu-expand** **int-direct** **int-isdn** **int-process** **teleseer** **unformatted**	The remaining formats are standard record formats. The one you use must be compatible with your call accounting software. Verify this through your vendor or the accounting system documentation.

Privacy — Digits to Hide

If you enable **CDR Privacy** on the **Station** screen for a given telephone, use this field to indicate how much of the dialed number to hide on the CDR record.

Valid entries	Usage
0 to **7**	Enter the number of digits to hide, counting from the end (right to left). For example, if you enter **4** in this field and the user dials 555-1234, only "555" would appear in the **Dialed Number** field of the CDR record.

Record Agent ID on Incoming

Only displays if the **Expert Agent Selection (EAS)** field is **y** on the **System-Parameters Customer-Options** screen. You cannot use both the **Called VDN** and the **Agent Login ID Instead of Group or Member**. Only one of these fields can be **y**.

Valid entries	Usage
y/n	Enter **y** to include the EAS agent's LoginID instead of the physical extension in the **Dialed Number** field of a CDR record.

Record Agent ID on Outgoing

Only displays if the **Expert Agent Selection (EAS)** field is **y** on the **System-Parameters Customer-Options** screen.

Valid entries	Usage
y/n	Enter **y** to include the EAS agent's LoginID instead of the physical extension in the **Dialed Number** field of a CDR record.

Record Call-Assoc TSC

Valid entries	Usage
y/n	Enter **y** to create records for call-associated temporary signaling connections. *If you have a lot of data connections this could increase the number of records. You may want to consider the capacity of your call collection device*.

Record Called Vector Directory Number Instead of Group or Member

If this option is enabled, the called VDN overrides the group or member information that normally appears in the **Dialed Number** field of the CDR record. If a call is directed through more than one VDN, the first VDN used for the call is stored. This applies only to calls routed to a hunt group by a vector, not to calls routed directly to an extension by a vector.

You cannot use both the **Called VDN** and the **Agent Login ID instead of Group or Member**. Only one of these fields can be **y**.

Valid entries	Usage
y/n	Enter **y** to include the Vector Directory Number (VDN) in the **Dialed Number** field of a CDR record.

Record Non-Call-Assoc TSC

A temporary signaling channel (TSC) is a virtual connection established within an ISDN D-channel. For more information, see the *DEFINITY Communications System Generic 2.2 and Generic 3 V2 DS1/CEPT1/ISDN PRI Reference*.

Valid entries	Usage
y/n	Enter **y** to create records for non-call-associated temporary signaling connections. *If you have a lot of data connections this could increase the number of records. You may want to consider the capacity of your record collection device*.

Record Outgoing Calls Only

Valid entries	Usage
y	Enter **y** to record only outgoing calls. *This can save space if you are only concerned with charges for outbound calls.*
n	Enter **n** to record both outgoing and incoming calls.

Remove # From Called Number

Valid entries	Usage
y	Enter **y** to have the "#" (or "E") symbol removed from the **Dialed Number** field of the call detail record. *You might need to verify that your output device can accept this format.*
n	Enter **n** to have the trailing "#" (or "E") symbol appear in the **Dialed Number** field whenever inter-digit time out occurs or users dial # to indicate the end of dialing.

Secondary Output Endpoint

Appears when the **Secondary Output Format** field is administered.

Valid entries	Usage
eia	Use this if the secondary output device is connected to the eia port. This is not a valid option on DEFINITY R systems.
Extension number	This is the extension of the data module (if used) that links the secondary output device to the server running Avaya Communication Manager.
CDR1, CDR2	Use this value if the CDR device is connected over a TCP/IP link, and this link is defined as either CDR1 or CDR2 on the **IP Services** screen.

Secondary Output Format

Controls the format of the call records sent to the secondary output device.

CAUTION:

Only qualified (Avaya) service personnel should administer a secondary output device. This option may cause loss of data when the buffer contains large amounts of data.

Valid entries	Usage
customized **int-direct** **int-process** **lsu** **unformatted**	These are the only formats you can use for a secondary output device. The format must be compatible with your call accounting software. Verify this through your vendor or the accounting system documentation.

Suppress CDR for Ineffective Call Attempts

Ineffective call attempts are calls that are blocked because the user did not have sufficient calling privileges or because all outgoing trunks were busy. This includes the unavailable incoming or outgoing trunks due to trunk usage allocation for ISDN Call-by-Call Service Selection trunks, incoming calls rejected by Communication Manager due to NSF mismatch, and ISDN calls that did not complete at the far end, if a cause value was provided. These calls appear on the CDR record with a condition code "E."

Valid entries	Usage
y	Enter **y** to ignore ineffective call attempts. *Use this if you have limited storage space for CDR records and records often overrun the buffer.*
n	Enter **n** to report ineffective call attempts. *This can tell you if your users are often unable to place outgoing calls, or if a large number of incoming calls are not completed.* You can also use this if you need to have records of attempts to contact a client, and are using ISDN trunks. Using this option requires more space for records.

Use Enhanced Formats

Enhanced formats provide additional information about time in queue and ISDN call charges, where available. This affects the "expanded", "teleseer", "lsu", "printer", and "unformatted" output formats.

Valid entries	Usage
y/n	Enter **y** to enable the use of the Enhanced version of the specified primary output format. You cannot use Enhanced formats and ISDN formats at the same time.

Use ISDN Layouts

ISDN Layouts provide more accurate information about the inter-exchange carrier and ISDN network services used for a call. This affects "lsu" and "printer" output formats, as well as any format with ISDN layouts, such as "teleseer."

Valid entries	Usage
y/n	Enter **y** to enable the use of the ISDN version of the specified primary output format. You cannot use ISDN formats and Enhanced formats at the same time.

Field descriptions for page 2

Figure 286: CDR System Parameters screen

```
Change system-parameters cdr                                    Page 2 of 2
                              CDR SYSTEM PARAMETERS
      Data Item - Length          Data Item - Length          Data Item - Length
  1: time_____________ - 4_   17: ____________________ - __   33: ____________________ - __
  2: space____________ - 1_   18: ____________________ - __   34: ____________________ - __
  3: duration_________ - 4_   19: ____________________ - __   35: ____________________ - __
  4: return___________ - 1_   20: ____________________ - __   36: ____________________ - __
  5: line-feed________ - 1_   21: ____________________ - __   37: ____________________ - __
  6: ____________________ - __   22: ____________________ - __   38: ____________________ - __
  7: ____________________ - __   23: ____________________ - __   39: ____________________ - __
  8: ____________________ - __   24: ____________________ - __   40: ____________________ - __
  9: ____________________ - __   25: ____________________ - __   41: ____________________ - __
 10: ____________________ - __   26: ____________________ - __   42: ____________________ - __
 11: ____________________ - __   27: ____________________ - __   43: ____________________ - __
 12: ____________________ - __   28: ____________________ - __   44: ____________________ - __
 13: ____________________ - __   29: ____________________ - __   45: ____________________ - __
 14: ____________________ - __   30: ____________________ - __   46: ____________________ - __
 15: ____________________ - __   31: ____________________ - __   47: ____________________ - __
 16: ____________________ - __   32: ____________________ - __   48: ____________________ - __
                                    Record length = 11
```

This page appears only if the **Primary Record Format** field is **customized**.

Data Item

Enter the data items in the order they should appear on the customized record. Only use this screen if you have arranged with your vendor to customize your call accounting system to receive these records.

You must include at least one field in order to have a record. See the table below for valid entries. The last two data items in a the record must be **line-feed** and **return**, in that order.

For more information, see "Call Detail Recording" in *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205*.*

Data Item	Length	Data Item	Length
acct-code	15	ixc-code	4
attd-console	2	line-feed	1
auth-code	7	location-from	3
bandwidth	2	location-to	3

1 of 2

Data Item	Length	Data Item	Length
bcc	1	in-trk-code	4
calling-num	15	ma-uui	1
clg-pty-cat	2	node-num	2
clg-num/in-tac	10	location-to	3
code-dial	4	null	1
code-used	4	out-crt-id	3
cond-code	1	ppm	5
country-from	3	res-flag	1
country-to	3	return	1
dialed-num	23	sec-dur	5
duration	4	space	1
feat-flag	1	time	4
frl	1	timezone-from	3
in-crt-id	3	timezone-to	6
ins	3	tsc_ct	4
isdn-cc	11	tsc_flag	1
		vdn	5

2 of 2

Length

Enter the length of each data item, if different from the default.

Valid entries	Usage
The maximum record length depends on the call accounting system you use. Check with your vendor.	The date field should be six-digits to ensure proper output. Certain fields default to the required length.

Record Length

Display-only field indicating the accumulated total length of the customized record, updated each time the length of a data item changes.

Change Station Extension

This screen allows an administrator to change extensions on the switch from one extension to another all at once. When the screen is filled out and submitted, all administration that was associated with the current extension will now be associated with the new extension. Any administration references of the extension being changed, such as references used in a vector, coverage, etc., will now reference the new extension. Once the extension has been changed, all references to the previous extension will be removed from the switch.

If an extension is changed that is also administered on an adjunct (such as voice mail or an ASAI link), the extension on the adjunct must also be changed to ensure proper functionality.

Note:

A forwarded extension administered as a button will not be handled by the **change extension-station *xxxxxxx*** command. It is recommended that the administrator use the **list usage** command prior to changing any extensions.

Figure 287: Change Station Extension screen

```
change extension-station xxxxxxx                                Page 1 of 1

                     CHANGE STATION EXTENSION

           Station Name: xxxxxxxxxxxxxxxxxxxxxxxxxxxxxxx     Port: xxxxxx

                                FROM EXTENSION                TO EXTENSION
                                --------------                ------------
                      Station: xxx-xxxx                        xxx-xxxx
                Message Lamp: xxx-xxxx                         xxx-xxxx
       Emergency Location Ext.: xxx-xxxx                       xxx-xxxx
IP Parameter Emergency Location: xxx-xxxx                 See IP-Network Map Form
```

Note:

You cannot use the **change extension-station** command to change the extension of a station if that station is administered as the emergency location extension for another station. For example, if station A is administered as the emergency location extension for station B, then:

- You cannot change the extension of station A using the **change extension-station** command unless you first change station B to assign a different emergency location extension.
- You can change the extension of station B. If you do, the **Change Station Extension** screen displays station A's extension in the **Emergency Location Ext.** field under the **From Extension** header.

Emergency Location Extension

The Emergency Location Extension from the **Station** screen associated with the current extension is displayed under **From Extension**.

Valid entries for "To Extension"	Usage
0 to **9**	Type a new extension for the **Emergency Location Ext**. field that will appear on the **Station** screen, up to seven numbers that make up a valid extension number for your dial plan.

IP Parameter Emergency Location

The Emergency Location Extension from the **IP Address Mapping** screen associated with the current extension is displayed under **From Extension**.

Valid entries for "To Extension"	Usage
n/a	The words "**See Ip-network Map Form**" display. The administrator can only change this field on the **IP Address Mapping** screen (using the ip-network-map command).

Message Lamp

The Message Lamp Extension associated with the current extension is displayed under **From Extension**.

Valid entries for "To Extension"	Usage
0 to 9	Type a new extension for the **Message Lamp Ext.** field, up to seven numbers that make up a valid extension number for your dial plan.

Port

This field is read only, and displays the port of the existing extension.

Station

The current extension that is being changed (the extension that was typed in the **change extension-station *xxxxxxx*** command) is displayed under **From Extension**.

Valid entries for "To Extension"	Usage
0 to **9**	Type the new extension that you want the current extension changed to, up to seven numbers that make up a valid extension number for your dial plan.

Station Name

This field is read only, and displays the name of the existing extension (the extension that was typed in the **change extension-station *xxxxxxx*** command).

Class of Restriction

Use this screen to establish classes of restriction (COR). Classes of restriction control call origination and termination. Your system may use only one COR or as many as necessary to control calling privileges. You can assign up to 96 different CORs (0 to 95).

Consider the following to enhance your system security:

1. Assign a separate COR to incoming and outgoing trunk groups, then restrict calling between the two groups.
2. Limit the calling permissions as much as possible by setting appropriate Calling Party Restrictions and Facility Restriction Levels (FRLs).

Field descriptions for page 1

Figure 288: Class of Restriction screen (page 1)

```
change cor 10                                                  Page 1 of 4
                        CLASS OF RESTRICTION

               COR Number: 10
          COR Description: supervisor

                      FRL: 0                                        APLT? y
  Can Be Service Observed? n                  Calling Party Restriction: none
Can Be A Service Observer? y                   Called Party Restriction: none
        Time of Day Chart: 1              Forced Entry of Account Codes? n
         Priority Queuing? n                        Direct Agent Calling? y
     Restriction Override: none               Facility Access Trunk Test? n
     Restricted Call List? n                          Can Change Coverage? n
   Unrestricted Call List? __ __ __ __ __ __ __ __ __ __
           Access to MCT? y                    Fully Restricted Service? n
Group II Category For MFC: 7                   Hear VDN of Origin Annc.? n
         Send ANI for MFE? n_                   Add/Remove Agent Skills? y
            MF ANI Prefix: ______               Automatic Charge Display? n
Hear System Music on Hold? y      PASTE(Display PBX Data on telephone)? n
                             Can Be Picked Up By Directed Call Pickup? n
                                          Can Use Directed Call Pickup? n
                                          Group Controlled Restriction: inactive
```

Access to MCT?

This field refers to Malicious Call Trace.

Valid entries	Usage
y	Enter **y** to allow permissions to activate a request to trace a malicious call.
n	Entering **n** prohibits this user from requesting a malicious call trace, but does not prevent this extension from appearing in the MCT History report, should this extension be the subject of a malicious call trace.

Add/Remove Agent Skills

Valid entries	Usage
y/n	Enter **y** to allow users with this COR to add and remove skills.

APLT

Valid entries	Usage
y/n	Enter **n** to allow access to APLT trunk group Enhanced Private Switched Communications System (EPSCS) or Common Control Switched Arrangement (CCSA) off-net facilities. If fully restricted service is enabled, set this field to **n**.

Automatic Charge Display

Shows the cost of an active outgoing call using Periodic Pulse Metering (PPM) or ISDN Advice of Charge (AOC) on Digital Communications Protocol (DCP) or Avaya BRI stations. Not available in the U.S

.

Valid entries	Usage
y	Displays call charges during and at the end of the call.
n	Call charges can be seen if users press the **disp-chrg** button before the call drops.

Called Party Restriction

Valid entries	Usage
Inward	Blocks the calling party from receiving incoming exchange network calls, attendant originated calls, and attendant completed calls.
Manual	Blocks the called party from receiving all calls except for those originated or extended by the attendant.
Public	Blocks the called party from receiving public network calls. Attendant calls are allowed to go through to the called party as well as attendant-assisted calls if the **Restriction Override** field in the public restricted station's COR is **attd** or **all**.
Termination	Blocks the called party from receiving any calls at any time.
none	No called party restrictions.

Calling Party Restriction

This field determines the level of calling restriction associated with this COR.

Note:

To enhance system security, limit calling permissions as much as possible.

Valid entries	Usage
Origination	Blocks the calling party from originating a call from the facility at any time. The party can only receive calls. A telephone with this COR may initiate Remote Access calls, if the COR of the barrier code allows it.
Outward	Blocks the calling party from calling outside the private network. Users can dial other users on the same server running Avaya Communication Manager or within a private network. To enhance security, Avaya recommends that you use outward restrictions when practical.

1 of 2

Valid entries	Usage
All-toll	Blocks the calling party from making ARS and trunk access calls from a facility assigned the COR to certain toll areas as defined in the **Dialed String** field on the **Toll Analysis** screen. The **Dialed String** field must be marked as being associated with the system's Toll List. The call completes if the facility's COR also is associated with an Unrestricted Call List and whose **Dialed String** field also matches the dialed number.
Tac-toll	Blocks the calling party from making trunk access calls from the facility assigned the COR to certain toll areas as defined in the **Dialed String** field on the **Toll Analysis** screen. The **Dialed String** field must be marked as being associated with the system's Toll List. The call completes if the facility's COR also is associated with an Unrestricted Call List and whose **Dialed String** field also matches the dialed number. See Toll Analysis on page 1525 for additional information.
none	No calling party restrictions.

2 of 2

Can Be a Service Observer

If you want an observer to observe users, set the users' CORs to **y** on the observer's COR Service Observing Permission table.

SECURITY ALERT:

The use of Service Observing features may be subject to federal, state, or local laws, rules, or regulations; or require the consent of one or both of the parties to the conversation. Customers should familiarize themselves with and comply with all applicable laws, rules, and regulations before using these features.

Note:

You cannot enter **y** in the previous two fields unless **Service Observing (Basic)** is enabled on the **System-Parameters Customer-Options** screen.

Valid entries	Usage
y/n	Enter **y** if users with this COR can service observe other users.

Can Be Picked Up By Directed Call Pickup

Valid entries	Usage
y/n	Enter **y** to allow this Station's or EAS agent's calls to be picked up by using the Directed Call Pickup Up feature. Before you can set this field to **y**, you must set **Directed Call Pickup** on the **Feature-Related System Parameters** screen to **y**.

Can Be Service Observed

Note that this field allows or denies service observing for not only physical extensions, but also for logical agent IDs and VDNs. If you want an observer to observe users, set the users' CORs to **y** on the observer's COR Service Observing Permission table

Valid entries	Usage
y/n	Enter **y** if users with this COR can be service observed.

Can Change Coverage

Valid entries	Usage
y/n	Enter **y** to allow station users with this COR to select one of two previously administered coverage paths or to activate, change, or deactivate call forward all calls or call forward busy/don't answer from any on-site or off-site location.

Can Use Directed Call Pickup

Valid entries	Usage
y/n	Enter **y** to allow the station, attendant, or EAS agent to pick up calls using the Directed Call Pickup feature. Set **Directed Call Pickup** on the **Feature-Related System Parameters** screen to **y** to set this field to **y**.

Category For MFC ANI

The **Category for MFC ANI** field always controls categories for Russian signaling trunks. It also may control categories for R2-MFC signaling trunks, depending on what value is in the **Use COR for Calling Party Category** field on the **System-Parameters Multi-Frequency** screen.

The Calling Party Category digit administered in this field is included as part of the ANI information sent to the Central Office on request using R2-MFC signaling.

Valid entries	Usage
1 to **10**	Used in other than U.S.

COR Description

Valid entries	Usage
Up to 35 characters	Enter a description of the COR that indicates how you use it. If you make this as clear as possible (for example, Customer Service, Legal Department), it will be easier to remember which COR to assign when you add users.

COR Number

This is a display-only field when the screen is accessed via an administration command such as **`change`** or **`display`**.

Direct Agent Calling

Valid entries	Usage
y/n	If this is **y**, users may dial an ACD agent's extension directly, rather than anyone in the agent pool. If the system is in Night Service, the call routes to the Night Service extension. If the extension with this COR belongs to an agent, the agent may receive calls directly.

Facility Access Trunk Test

An associated feature button (**trk-ac-alm**) status lamp lights when a successful test attempt occurs. Pressing one of the alarm buttons (ten maximum) when its associated status lamp is lit turns off all lamps on all buttons whether the access is still in progress or has completed

.

Valid entries	Usage
y/n	Enter **y** to allow users with this COR to perform Facility Access Trunk Tests.

Forced Entry of Account Codes

FEAC must be enabled on the **System Parameters Customer-Options** screen and on the **CDR System Parameters** screen.

See Forcing Users to Enter Account Codes on page 597 for more information.

Note:

If a COR requiring entry of account codes is assigned a VDN, the route to commands executed by the associated vector will not be successful.

Valid entries	Usage
y/n	Enter **y** to indicate that an account code must be dialed when making outgoing trunk calls. If this is **y**, any telephone assigned the associated COR must dial an account code before making an outgoing call. If you set this to y for a COR assigned to a trunk group, users must dial account codes before calling out over that trunk group. This may be useful for trunks used in international calls, and those that are more expensive. If a call is being routed by ARS, account code checking is not done on the COR.

FRL

Valid entries	Usage
0 to **7**	Enter an originating FRL number. AAR and/or ARS features use this entry to determine call access to an outgoing trunk group. Outgoing call routing is determined by a comparison of the FRLs in the AAR/ ARS Routing Pattern and the FRL associated with the COR of the call originator (typically, a telephone user). An originating FRL of 0 has the least calling privileges. To enhance system security, assign the lowest possible FRL.

Fully Restricted Service

Note:

If this field is enabled, the **APLT** field must be **n**.

Valid entries	Usage
y/n	When **y** entered for a given COR, stations assigned that COR will not have access to the public network for either incoming or outgoing calls.

Group II Category For MFC

This field always controls categories for Russian signaling trunks. It can control categories for R2-MFC signaling trunks, depending on the value of the **Use COR for Calling Party Category** field on the **Multifrequency-Signaling-Related System Parameters** screen.

The Calling Party Category digit administered in this field is included as part of the ANI information sent to the Central Office on request using R2-MFC signaling

.

Valid entries	Usage
1 to **10**	Enter the value you want the server running Avaya Communication Manager to send as the Calling and/or Called Party Category for phones or trunks that use this COR.

Group Controlled Restriction

A display-only field that determines if the current COR is under controlled restriction. This field can help troubleshoot problems by first checking its value.

Valid entries	Usage
active	indicates the COR is controlled restricted
inactive	indicates the COR is not controlled restricted

Hear System Music on Hold

Valid entries	Usage
y/n	Enter **y** to allow the Music on Hold feature to be activated by a telephone.

Hear VDN of Origin Announcement

Valid entries	Usage
y/n	Enter **y** if users with this COR can receive VDN of Origin messages.

MF ANI Prefix

Defines the prefix to apply to an extension number when ANI is sent to the CO. This overrides any ANI prefix administered on the **Multifrequency Signaling** screen. This does not apply when ANI is tandemed through the Communication Manager server on tandem calls. This field also applies to the ANI for the server when the originating side is a trunk and there was no ANI.

Valid entries	Usage
1 to 7 digits or blank	If you want the entire number to display on the receiving end, enter all digits except the extension number.

Partitioned Group Number

This field appears only if **AAR/ARS Partitioning** is **y** and **Time of Day Routing** is **n** on the **System Parameters Customer-Options** screen.

Valid entries	Usage
1 to **8**	Enter the AAR/ARS partitioned group number associated with this COR.

PASTE (Display PBX Data on telephone)

Valid entries	Usage
y/n	Enter **y** to download all lists. Enter **n** to disallow the PASTE feature.

Priority Queuing

Valid entries	Usage
y	Enter **y** to allow the telephone user's calls to be placed ahead of non-priority calls in a hunt group queue
n	If you do not use Automatic Call Distribution (**ACD** is not enabled on the **System-Parameters Customer-Options** screen), this field must be **n**.

Restricted Call List

This list can be used whether the COR is toll restricted. The Restricted Call List (RCL) has priority over the Toll Analysis Unrestricted Call List (UCL). A call attempt from a facility assigned a COR (with **RCL** field set to **y**), whose dialed digit string is on the **Toll Analysis** screen and is marked as being associated with the RCL, will be denied.

Valid entries	Usage
y/n	Enter **y** to specify that this COR will have access to the system's Restricted Call List (see Toll Analysis on page 1525).

Restriction Override

Allows the specified users to bypass restriction on conference, transfer or call forwarding operations.

Valid entries	Usage
attendant	A telephone with a COR that is inward restricted cannot receive public network, attendant-originated, or attendant-extended calls. Enter **attendant** to give your attendants the ability to override this restriction.
all	Enter **all** if you want all of the users with this COR to override inward restrictions.
none	Enter **none** if you do not want any users of this COR to bypass the restrictions.

Send ANI for MFE

Only applicable for Spain. Valid for 2/6 signaling, but not 2/5 signaling. The following field appears only if Expert Agent Selection (EAS) is enabled on the **Feature-Related System-Parameters** screen

Valid entries	Usage
y	Enter **y** to enable Automatic Number Identification (ANI). When the value is **y**, Avaya Communication Manager sends the calling party's number to the public or IBERCOM network so that charges will be broken down by line.
n	If this value is **n**, charges are not itemized by line, and your company will receive a single bill for the total number of calls made (block charging).

Time of Day Chart

Appears only if **Time of Day** field is enabled on the **System Parameters Customer-Options** screen. See Setting up Time of Day Routing on page 281 for more information.

Valid entries	Usage
1 to **8**	Enter the AAR/ARS time-of-day-chart number associated with this COR.

Unrestricted Call List

Any entry on the **Toll Analysis** screen with an **X** in the **Toll List** column is restricted, meaning that the system blocks any attempt to complete a call containing the Dialed String. However, this field overrides that restriction.

For example, if the **Toll Analysis** screen shows a **Dialed String** entry of 538 and there is an **X** in the **Toll List** column, the 538 number is restricted. To override this restriction, in the **Toll Analysis** screen, enter **X** in the **5** column under the **Unrestricted Call List** heading. In the **Class of Restriction** screen, in this field, enter **5** to complete the restriction override.

Valid entries	Usage
1 to **10** or blank	Appears when **Calling Party Restriction** is **all-toll** or **tac-toll**. This field allows a user to complete a toll call with "restricted" dialed digits. This field is associated with the **Dialed String** field on the **Toll Analysis** screen. An Unrestricted Call List number is denoted on that screen.

Use COR for all Group II Responses

This field appears only if, on the **Multifrequency-Signaling-Related System Parameters** screen, the **Outgoing Call Type** field is **group-ii-mfc**.

Valid entries	Usage
y/n	Entering **y** allows the COR administered category to be used for both the calling party and called party categories.

Field descriptions for page 2

Figure 289: Class of Restriction screen (page 2)

```
change cor 10                                                  Page 2 of 4
                        CLASS OF RESTRICTION

                     MF Incoming Call Trace? n
                Brazil Collect Call Blocking? n
                     Block Transfer Display? n
Block Enhanced Conference/Transfer Displays? y
                     Remote Logout of Agent? n

                            Station Lock COR: 10
  Outgoing Trunk Disconnect Timer (minutes):
             Line Load Control:
      Maximum Precedence Level:        Preemptable?
           MLPP Service Domain:
```

Block Enhanced Conference/Transfer Display

Use this field to add display messages regarding conference and transfer features on digital phones.

Valid entries	Usage
y/n	Enter **y** to block all the enhanced conference/transfer display messages except "Transfer Completed"

Block Transfer Display

Valid entries	Usage
y/n	Enter **y** to prevent users of DCP, Hybrid, ISDN-BRI, or wireless display telephones from receiving a confirmation message when they transfer a call.

Brazil Collect Call Blocking

For Brazil only.

Valid entries	Usage
y/n	Enter **y** to permit all Brazilian trunks calls that terminate to a station to send back a double answer to the CO. This double answer tells the CO that this particular station cannot accept collect calls. The CO then tears down the call if it is a collect call. Set **Country** on the **Trunk Group** screen to **23** and set this field to **y**.

Line Load Control

Valid entries	Usage
1 to **4**	Enter the line load control level for this COR, where **1** has no restrictions, and **4** is most restrictive.

Maximum Precedence Level

Assign a maximum precedence level for extensions with this COR for use with the Multiple Level Precedence and Preemption feature.

Valid entries	Usage
fo	Flash Override
fl	Flash
im	Immediate
pr	Priority
ro	Routine (default)

MF Incoming Call Trace

Valid entries	Usage
y/n	Enter **y** to allow assignment of a Call Trace COR to a station. Avaya Communication Manager then generates an MFC backward signal (administered on the **System-Parameters Multifrequency-Signaling** screen) during call setup instead of the "free" signal. This triggers the central office to collect trace information before releasing the calling party, if the terminating station's COR has this feature set to **y**.

MLPP Service Domain

Valid entries	Usage
1 to **16777215**	Enter the service domain for users and trunks to which this particular COR is assigned.

Preemptable

Valid entries	Usage
y/n	An entry of **y** makes extensions with this COR preemptable for Multiple Level Precedence and Preemption calls.

Remote Logout of Agent

Use a feature access code to logout an idle ACD or EAS agent without being at the agent's telephone.

Valid entries	Usage
y/n	Enter **y** to allow remote logout of an idle ACD or EAS agent.

Station Lock COR

This field defaults to the current screen COR. Extensions that are assigned this COR can use Station Lock with the access code administered on the **FAC** screen

Valid entries	Usage
0 to **95**	This field defaults to current COR.

Outgoing Trunk Disconnect Timer (minutes)

This feature provides the capability to disconnect an outgoing trunk automatically after an administrable amount of time. This field defaults to blank (outgoing trunk calls are only disconnected when dropped by one or all parties), or you may enter a timer value in number of minutes to apply to outgoing trunk calls if the initiating party belongs to this COR

Valid entries	Usage
2 to **999**	Enter a value of as many as 3 characters in number of minutes. A warning tone is given to all parties on the trunk call 1 minute before the administered value (i.e., after 1 to 998 minutes have elapsed) and a second warning tone is heard 30 seconds later. The call is automatically disconnected 30 seconds after the second warning tone.

Field descriptions for page 3

Figure 290: Class of Restriction screen (page 3)

```
change cor 10                                                  Page 3 of 4
                        CLASS OF RESTRICTION
CALLING PERMISSION (Enter y to grant permission to call specified COR)
  0? n     12? n     24? n     36? n     48? n     60? n     72? n     84? n
  1? n     13? n     25? n     37? n     49? n     61? n     73? n     85? n
  2? n     14? n     26? n     38? n     50? n     62? n     74? n     86? n
  3? n     15? n     27? n     39? n     51? n     63? n     75? n     87? n
  4? n     16? n     28? n     40? n     52? n     64? n     76? n     88? n
  5? n     17? n     29? n     41? n     53? n     65? n     77? n     89? n
  6? n     18? n     30? n     42? n     54? n     66? n     78? n     90? n
  7? n     19? n     31? n     43? n     55? n     67? n     79? n     91? n
  8? n     20? n     32? n     44? n     56? n     68? n     80? n     92? n
  9? n     21? n     33? n     45? n     57? n     69? n     81? n     93? n
 10? n     22? n     34? n     46? n     58? n     70? n     82? n     94? n
 11? n     23? n     35? n     47? n     59? n     71? n     83? n     95? n
```

CALLING PERMISSION

Valid entries	Usage
y/n	A **y** means an originating facility assigned this COR can be used to call facilities assigned this COR. Enter **n** for each COR number (0 through 95) that cannot be called by the COR being implemented.

Field descriptions for page 4

Figure 291: Class of Restriction screen (page 4)

```
change cor 10                                                  Page 4 of 4
                      CLASS OF RESTRICTION
SERVICE OBSERVING PERMISSIONS
(Enter y to grant permission to service observe specified COR)
  0? n     12? n     24? n     36? n     48? n     60? n     72? n     84? n
  1? n     13? n     25? n     37? n     49? n     61? n     73? n     85? n
  2? n     14? n     26? n     38? n     50? n     62? n     74? n     86? n
  3? n     15? n     27? n     39? n     51? n     63? n     75? n     87? n
  4? n     16? n     28? n     40? n     52? n     64? n     76? n     88? n
  5? n     17? n     29? n     41? n     53? n     65? n     77? n     89? n
  6? n     18? n     30? n     42? n     54? n     66? n     78? n     90? n
  7? n     19? n     31? n     43? n     55? n     67? n     79? n     91? n
  8? n     20? n     32? n     44? n     56? n     68? n     80? n     92? n
  9? n     21? n     33? n     45? n     57? n     69? n     81? n     93? n
 10? n     22? n     34? n     46? n     58? n     70? n     82? n     94? n
 11? n     23? n     35? n     47? n     59? n     71? n     83? n     95? n
```

SERVICE OBSERVING PERMISSION

Valid entries	Usage
y/n	A **y** grants permission to observe specific CORs. Enter **n** for each COR number (0 through 95) that cannot be observed by the COR being implemented.

Class of Service

This screen administers access permissions for call processing features that require dial code or feature button access.

Note:

Class of Service (COS) does not apply to trunk groups except for the Remote Access feature.

A COS assignment defines whether or not a telephone user may access or use the following features and functions. Up to 16 different COS numbers may be administered (0 to 15).

Figure 292: Class of Service screen

```
change cos                                                      Page 1 of 1
                          CLASS OF SERVICE

                                  0 1 2 3 4 5 6 7 8 9 10 11 12 13 14 15

    Auto Callback                 n y y n y n y n y n  y  n  y  n  y  n
    Call Fwd-All Calls            n y n y y n n y y n  n  y  y  n  n  y
    Data Privacy                  n y n n n y y y y n  n  n  n  y  y  y
    Priority Calling              n y n n n n n n n y  y  y  y  y  y  y
    Console Permissions           n n n n n n n n n n  n  n  n  n  n  n
    Off-hook Alert                n n n n n n n n n n  n  n  n  n  n  n
    Client Room                   n n n n n n n n n n  n  n  n  n  n  n
    Restrict Call Fwd-Off Net     n y y y y y y y y y  y  y  y  y  y  y
    Call Forward Busy/DA          n n n n n n n n n n  n  n  n  n  n  n
    Personal Station Access (PSA) n n n n n n n n n n  n  n  n  n  n  n
    Extended Forwarding All       n n n n n n n n n n  n  n  n  n  n  n
    Extended Forwarding B/DA      n n n n n n n n n n  n  n  n  n  n  n
    Trk-to-Trk Restriction Override n n n n n n n n n n  n  n  n  n  n  n
    QSIG Call Offer Originations  n n n n n n n n n n  n  n  n  n  n  n
    Automatic Exclusion           n n n n n n n n n n  n  n  n  n  n  n
```

The screen lists the default values for each COS/feature combination. For a particular combination, **y** allows access to the feature and **n** denies access. Assign entries on the screen for each COS to be implemented. Default values are shown on the screen.

CAUTION:

Because many hunt groups are set up with COS 1, be careful when you assign restrictions to COS 1.

Automatic Callback

Allows this user to request Automatic Callback. See "Automatic Callback" in *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205, for more information.

Automatic Exclusion

Allows a user to activate automatically Exclusion when they go off hook on a station that has an assigned **Exclusion** button. If set to **n**, allows a user manual exclusion when they press the **Exclusion** button before dialing or during a call. Appears when, on the **Feature-Related System Parameters** screen, the **Automatic Exclusion by COS** field is **y**.

Call Forwarding All Calls

Allows this user to forward all calls to any extension. See "Call Forwarding" in *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205, for more information.

Call Forwarding Busy/DA

Allows this user to forward calls to any extension when the dialed extension is busy or does not answer. See "Call Forwarding" in *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205, for more information.

Client Room

Allows users to access Check-In, Check-Out, Room Change/Swap, and Maid status functions. In addition, Client Room is required at consoles or telephones that are to receive message-waiting notification. You can administer class of service for Client Room only when you have Hospitality Services and a Property Management System interface. See *GuestWorks and DEFINITY Systems Technician Handbook for Hospitality Installations* for more information.

Console Permissions

Console Permissions allow multiappearance telephone users to control the same features that the attendant controls. You might assign this permission to front-desk personnel in a hotel or motel, or to a call center supervisor. With console permission, a user can:

- Activate Automatic Wakeup for another extension
- Activate and deactivate controlled restrictions for another extension or group of extensions
- Activate and deactivate Do Not Disturb for another extension or group of extensions
- Activate Call Forwarding for another extension

- Add and remove agent skills
- Record integrated announcements

Data Privacy

Allows this user to enter a feature access code to protect a data call from interruption by any of the system's override or ringing features. See Data Privacy on page 519 for more information.

Extended Forwarding All

Allows a user to administer call forwarding (for all calls) from a remote location. You cannot change a COS to **y** if **Extended Cvg/Fwd Admin** on the **System Parameters Customer-Options** screen is **n**.

Extended Forwarding B/DA

Allows this user to administer call forwarding (when the dialed extension is busy or does not answer) from a remote location. You cannot change this COS to **y** if **Extended Cvg/Fwd Admin** on the **System Parameters Customer-Options** screen is **n**.

Off-Hook Alert

See "Emergency Access to the Attendant" in *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205, for more information. To enable this option, either the **Hospitality (Basic)** or **Emergency Access to Attendant** field must be enabled in your license file. When enabled, these fields display as **y** on the **System-Parameters Customer-Options** screen.

Personal Station Access (PSA)

Allows users to associate a telephone to their extension with their programmed services, using a feature access code. This field must be set to n for virtual phones. You cannot change this field to **y** if **Personal Station Access (PSA)** on the **System Parameters Customer-Options** screen is **n**. See "Personal Station Access" in *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205, for more information

Priority Calling

Allows user to dial a feature access code to originate a priority call. Such calls ring differently and override send all calls, if active. See "Priority Calling" in *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205, for more information

QSIG Call Offer Originations

Allows this user to invoke QSIG Call Offer services. See *Administration for Network Connectivity for Avaya Communication Manager,* 555-233-504, for more information.

Restrict Call Fwd-Off Net

This restricts users from forwarding calls to the public network. For security reasons, this should be enabled for all classes of service except the ones you use for very special circumstances. See "Call Forwarding Off-net" in *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205, for more information

Trk-to-Trk Restriction Override

Users with this COS override any system and/or COR-to-COR calling party restrictions that would otherwise prohibit the trunk-to-trunk transfer operation for users with this COS. See "Trunk-to-Trunk Transfer" in *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205, for more information

SECURITY ALERT:
Use this COS capability with caution. The ability to perform trunk-to-trunk transfers greatly increases the risk of toll fraud.

Code Calling IDs

On systems with chime paging, use this screen to assign a unique series of chimes (a *chime code*) to extensions. The chime code assigned to an extension plays over the speakers whenever that extension is paged. You may assign chime codes to up to 125 extensions.

Figure 293: Code Calling IDs screen

```
change paging code-calling-ids                              Page 1 of 2

                                  CODE CALLING IDs
     ID ASSIGNMENTS
               Id   Ext     Id   Ext     Id   Ext     Id   Ext     Id   Ext
               111: ____    141: ____    221: ____    251: ____    331: ____
               112: ____    142: ____    222: ____    252: ____    332: ____
               113: ____    143: ____    223: ____    253: ____    333: ____
               114: ____    144: ____    224: ____    254: ____    334: ____
               115: ____    145: ____    225: ____    255: ____    335: ____
               121: ____    151: ____    231: ____    311: ____    341: ____
               122: ____    152: ____    232: ____    312: ____    342: ____
               123: ____    153: ____    233: ____    313: ____    343: ____
               124: ____    154: ____    234: ____    314: ____    344: ____
               125: ____    155: ____    235: ____    315: ____    345: ____
               131: ____    211: ____    241: ____    321: ____    351: ____
               132: ____    212: ____    242: ____    322: ____    352: ____
               133: ____    213: ____    243: ____    323: ____    353: ____
               134: ____    214: ____    244: ____    324: ____    354: ____
               135: ____    215: ____    245: ____    325: ____    355: ____
```

Ext

This field assigns extensions to chime codes. Only one extension can be assigned to each chime code.

Valid entries	Usage
An extension number	Enter a physical extension, not a VDN, to assign that extension to a code. Otherwise, leave this field blank.

Related topics

See Setting up Chime Paging Over Loudspeakers on page 494 for instructions.

See "Loudpeaker Paging" in *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205, for more information for a description of the feature.

Command Permission Categories

Use this screen to administer a user's permissions associated with their login. When set to y, the permissions on this screen apply for the object that is not restricted. Use the second and third pages of the **Command Permission Categories** screen to restrict a user from any access to specified objects. To see pages 2 and 3, type **y** in the **Additional Restrictions** field.

Field descriptions for page 1

Figure 294: Command Permission Categories screen

```
change permissions angi3                                    Page 1 of 3
                          COMMAND PERMISSION CATEGORIES
                            Login Name: angi3
   COMMON COMMANDS
                 Display Admin. and Maint. Data? n
                            System Measurements? n
                 System Mgmt Data Transfer Only? n
   ADMINISTRATION COMMANDS
         Administer Stations? n             Administer Features? n
           Administer Trunks? n          Administer Permissions? n
     Additional Restrictions? n

   MAINTENANCE COMMANDS
           Maintain Stations? n       Maintain Switch Circuit Packs? n
             Maintain Trunks? n      Maintain Process Circuit Packs? n
            Maintain Systems? n               Maintain Enhanced DS1? n
```

Additional Restrictions

Use page 2 and 3 to add objects (up to 40) that this user cannot manipulate. If an object appears on the Additional Restrictions page, users cannot display, add, change, or do anything else with that object.

Valid entries	Usage
y/n	Enter **y** to create additional restrictions, and to have the second and third pages of this screen appear.

Administer Features

Use caution when assigning this permission to a user.

Valid entries	Usage
y/n	Enter **y** to allow a user to administer feature-related parameters, such as coverage paths, class of service, class of restriction, system parameters, authorization codes, and security.

Administer Permissions

This permission only applies to *super-user* logins.

Valid entries	Usage
y/n	Enter **y** to allow a user to administer logins and command permissions.

Administer Stations

Valid entries	Usage
y/n	Enter **y** to allow a user to add, change, duplicate, or remove stations, data modules and associated features, such as abbreviated dialing, vectors, and routing tables.

Administer Trunks

Give this permission only to users who are very familiar with these features.

Valid entries	Usage
y/n	Enter **y** to allow a user to administer AAR/ARS, trunk groups, remote access, and route patterns.

Display Admin. and Maint. Data

Valid entries	Usage
y/n	Enter **y** to allow a user to use the display, list, monitor, status, and schedule commands and also change their own passwords and schedule reports.

Login Name

This display-only field shows the login to which these permissions apply.

Maintain Enhanced DS1

You can only enter a value in this field if the **DS1 MSP** field is set to **y** on the **System-Parameters Customer-Options** screen.

Valid entries	Usage
y/n	Enter **y** to allow a user to perform enhanced DS1 maintenance.

Maintain Process Circuit Packs

You can only enter a value in this field if the **Processor and System MSP** field is set to **y** on the **System-Parameters Customer-Options** screen.

Valid entries	Usage
y/n	Enter **y** to allow a user to perform processor maintenance.

Maintain Stations

You can only enter a value in this field if the **Station** and **Trunk MSP** field is set to **y** on the **System-Parameters Customer-Options** screen.

Valid entries	Usage
y/n	Enter **y** to allow a user to perform station maintenance.

Maintain Switch Circuit Packs

You can only enter a value in this field if the **Station** and **Trunk MSP** field is set to **y** on the **System-Parameters Customer-Options** screen.

Valid entries	Usage
y/n	Enter **y** to allow a user to perform circuit pack maintenance.

Maintain Systems

You can only enter a value in this field if the **Processor and System MSP** field is set to **y** on the **System-Parameters Customer-Options** screen.

Valid entries	Usage
y/n	Enter **y** to allow a user to perform system maintenance.

Maintain Trunks

You can only enter a value in this field if the **Station** and **Trunk MSP** field is set to **y** on the **System-Parameters Customer-Options** screen.

Valid entries	Usage
y/n	Enter **y** to allow a user to perform trunk maintenance.

System Measurements

This field only appears for DEFINITY SI systems.

Valid entries	Usage
y/n	Enter **y** to allow a user to use the list measurements commands.

System Mgmt Data Transfer Only

Valid entries	Usage
y/n	Enter **y** to indicate the server running Avaya Communication Manager is accessible by Network Administration.

Field descriptions for page 2

Figure 295: Command Permission Categories—Restricted Object List screen

```
change permissions angi3                                    Page 2 of 3

                          COMMAND PERMISSION CATEGORIES
                             RESTRICTED OBJECT LIST
          _____________________________     _____________________________
          _____________________________     _____________________________
          _____________________________     _____________________________
          _____________________________     _____________________________
          _____________________________     _____________________________
          _____________________________     _____________________________
          _____________________________     _____________________________
          _____________________________     _____________________________
          _____________________________     _____________________________
          _____________________________     _____________________________
```

Pages 2 and 3 of this screen are identical, and allow you to specify up to 40 objects that this user cannot access. These pages do not appear unless you set both the **Administer Station** and **Additional Restrictions** fields to **y** on page 1 of this screen. If you want to limit a user's permissions beyond those on page 1, enter the objects in this list. For example, if you want a user to be able to add and change stations, but not Vector Directory Numbers (VDN), in the **Administer Stations** and **Additional Restrictions** fields, type **y**. Then on page 2, type **vdn** as a restricted object.

Configuration Set

This screen defines a number of call treatment options for EC500 cell telephone calls. The EC500 allows the use of up to 10 Configuration Sets, which are already defined in the system using default values.

Field descriptions for page 1

Figure 296: Configuration Set screen

```
change xmobile configuration-set 1                              Page   1 of 1

                              CONFIGURATION SET: 1

     Configuration Set Description:
               Calling Number Style:
CDR for Calls to EC500 Destination?
       Fast Connect on Origination?
      Post Connect Dialing Options?
```

Calling Number Style

Determines the format of the caller ID for calls from a local Avaya Communication Manager extension to an EC500 cell telephone.

Valid entries	Usage
network	Provides a display of only 10-digit numbers. For internal calls, the ISDN numbering tables are used to create the calling number and DCS calls use the ISDN calling number if provided. The externally provided calling number is used when available for externally originated calls.
pbx	Provides a display of less than 10-digits. Extensions sent as the calling number for all internally- and DCS network-originated calls.

CDR for Calls to EC500 Destination

Determines whether a call detail record is generated for any call to the cell telephone.

Note:

CDR reporting for EC500 calls relies on the **CDR Reports** field on the **Trunk Group** screen. If, on the **Trunk Group** screen, the **CDR Reports** field is **n**, no CDR is generated even if this field is **y**.

Valid entries	Usage
y	Treats calls to the XMOBILE station as trunk calls and generates a CDR.
n	Treats calls to the XMOBILE station as internal calls and does not generate a CDR.

Configuration Set Description

Describes the purpose of the configuration set.

Valid entries	Usage
Up to 20 alphanumeric characters or blank	For example, EC500 handsets.

Fast Connect on Origination

Determines whether some additional processing occurs on the server running Avaya Communication Manager prior to connecting a call.

Valid entries	Usage
y/n	Enter **y** to send CONNECT messages.

Post Connect Dialing Options

Determines whether additional capabilities, beyond standard ISDN dialing, are available for those incoming ISDN trunk calls that are mapped into XMOBILE stations. These options come into effect after the call has entered the active state (Communication Manager has sent a CONNECT message back to the network).

Valid entries	Usage
dtmf	Expect digits from either in-band or out-of-band, but not simultaneously. The server allocates a DTMF receiver whenever it needs to collect digits. This option normally would be used for EC500 XMOBILE station calls.
out-of-band	Expect all digits to be delivered by out-of-band signaling only. The server running Avaya Communication Manager collects digits that it needs from the out-of-band channel (no touch-tone receiver). In addition, any digits received when the server is not collecting digits are converted to DTMF and broadcast to all parties on the call. This option is in force for DECT XMOBILE station calls.
both	Expect all subsequent digits to be delivered by simultaneous in-band and out-of-band signaling. Out-of-band signaling consists of digits embedded in ISDN INFO messages while the in-band signaling consists of DTMF in the voice path. The server running Communication Manager collects all digits that it needs from the out-of-band channel. No touch tone receive is allocated in order to prevent collecting double digits. End-to-end signaling occurs transparently to the server via in-band transmission of DTMF. This option is in force for PHS XMOBILE station calls.

Console Parameters

This screen administers attendant console group parameters. This includes basic parameters for Centralized Attendant Service (CAS) and Inter-PBX Attendant Service (IAS). A list of the administered attendant consoles also displays on this screen.

Field descriptions for page 1

Figure 297: Console Parameters — Default Attendant Group screen

```
change console-parameters                                    Page   1 of x
                              CONSOLE PARAMETERS
         Attendant Group Name: OPERATOR
                         COS: 0                                    COR: 0
       Calls in Queue Warning: 5                    Attendant Lockout? y
        Ext Alert Port (TAAS):
                         CAS: none
                                                Night Service Act. Ext.:
                IAS (Branch)? n                 IAS Tie Trunk Group No.:
          IAS Att. Access Code:                   Alternate FRL Station:
               Backup Alerting? n         DID-LDN Only to LDN Night Ext? n
     Attendant Vectoring VDN:
```

AAR/ARS Access Code

Appears if the **CAS** field is **QSIG-branch**. An optional field that contains an AAR/ARS access code to route to the main PBX, if needed.

Valid entries	Usage
0 to **9**, *****, # blank	Enter up to 4 digits.

Alternate FRL Station

This is a display-only field indicating the extension of the alternate facility restriction level (FRL) activation station.

Attendant Group Name

Valid entries	Usage
1 to **27** alphanumeric characters	Enter a name for the attendant group.

Attendant Lockout

Attendant Lockout prevents an attendant from re-entering a multiple-party connection held on the console unless recalled by a telephone user.

Attendant Lockout provides privacy for parties on a multiple-party call held on the console. The held parties can hold a private conversation without interruption by the attendant.

Valid entries	Usage
y/n	Enter **y** to activate Privacy — Attendant Lockout. If **y** is entered, the attendant is prohibited from reentering a conference call that has been placed on hold unless recalled by a telephone user on the call.

Attendant Vectoring VDN

This field appears only if, on the **System-Parameters Customer-Options** screen, the **Attendant Vectoring** field is **y** and the **Tenant Partitioning** field is **n**.

Valid entries	Usage
Assigned VDN extension or blank	Enter an assigned Attendant VDN extension or blank.

Backup Alerting

Indicates whether or not system users can pick up alerting calls if the attendant queue has reached its warning state.

Calls In Queue Warning

Enter the number of incoming calls that can be in the attendant queue before the console's second Call Waiting lamp lights. The console's first Call Waiting lamp lights when any incoming calls are waiting to be answered. The second lamp lights when the number of calls waiting equals the value you entered in the **Calls in Queue Warning** field.

Valid entries	Usage
1 to **300**	Enter the number of incoming calls that can be in the attendant queue before the console's second Call Waiting lamp lights.

CAS

The CAS Main or Branch features must be enabled on the **System-Parameters Customer-Options** screen for either of these features to be functional here.

Valid entries	Usage
main	Enables CAS Main capability.
branch	Enables CAS Branch capability.
none	No Centralized Attendant Service is enabled.
QSIG-main	Can be used if, on the **System-Parameters Customer-Options** screen, the **Centralized Attendant** field is **y**. Indicates all attendants are located on the main PBX.
QSIG-branch	Can be used if, on the **System-Parameters Customer-Options** screen, the **Centralized Attendant** field is **y**. Indicates there are no local attendants and routes to the main PBX.

CAS Back-Up Ext.

This field handles attendant-seeking calls if the RLT trunk group to the CAS Main server is out of service or if CAS Back-Up is activated. This field must be explicitly defined as an extension in the dial plan. Neither a prefixed extension nor a VDN extension is allowed. Appears only when **branch** is entered in the **CAS** field.

Valid entries	Usage
An extension number for a station	
individual attendant console	
hunt group	
TEG	

COR

For more information about Class of Restriction (COR), see the *Feature Description and Implementation for Avaya Communication Manager* document

Valid entries	Usage
0 to **95**	Enter the class of restriction (COR) number that reflects the desired features for the attendant. You can override this COR, by assigning a different COR on the individual **Attendant** screen.

COS

For more information about Class of Service (COS), see the *Feature Description and Implementation for Avaya Communication Manager* document

Valid entries	Usage
1 to **15**	Enter a class of service (COS) number that reflects the desired features for all your attendant consoles. You can override this COS, by assigning a different COS on the individual **Attendant** screen.

DID-LDN Only to LDN Night Ext.

Valid entries	Usage
y	Enter **y** to allow only listed directory number (LDN) calls to go to the listed directory night service extension.
n	Enter **n** if you want all attendant seeking calls to route to the LDN night service extension.

Ext Alert Port (TAAS)

Enter the seven-digit port number assigned to the external alerting device. This supports the Night Service — Trunk Answer From Any Station feature.

Note:

Type an **x** in this field to indicate that there is no hardware associated with this port assignment. If an **x** is used here, you must also fill in the **Ext Alert (TAAS) Extension** field.

Ext Alert (TAAS) Extension

Appears only when an **x** is entered in the **Ext Alert Port (TAAS)** field. This extension is used by the Terminal Translation Feature (TTI) to assign a port to the Ext Alert Port from a station on the Ext Alert port during system installation or provisioning. Once a port is assigned (either via TTI or by changing the **Ext Alert Port** field from the G3-MA or other manager terminal) the extension is automatically removed and treated as unassigned.

IAS Att. Access Code

Enter the extension number of the attendant group at the main server running Avaya Communication Manager. This entry is required when IAS Branch is **y**. Does not appear if, on the **System-Parameters Customer-Options** screen, the **Centralized Attendant** field is **y**.

IAS (Branch)

Enables or disables Inter-PBX Attendant Service (IAS) Branch feature. Does not appear if, on the **System-Parameters Customer-Options** screen, the **Centralized Attendant** field is **y**.

Note:

CAS and IAS cannot both be active at the same time.

IAS Tie Trunk Group No.

Note:

Enter the number of the tie trunk group to the main for the IAS (Branch). This entry is required when IAS Branch is **y**. Does not appear if, on the **System-Parameters Customer-Options** screen, the **Centralized Attendant** field is **y**.

Valid entries	Usage
1 to 666 **1** to **2000**	For DEFINITY R, CSI, SI. For S8300 Media Server, S8700 or S8710 IP-Connect, S8700 or S8710 Multi-Connect.

Night Service Act. Ext.

This is a display-only field containing the extension of the current night service activation station, if any. Such a station is administered by assigning it a **night-serv** button.

QSIG CAS Number

Appears if the **CAS** field is **QSIG-branch**. Contains the complete number of the attendant group at the main server running Avaya Communication Manager, or a vector directory number (VDN) local to the branch server. This field cannot be left blank

Valid entries	Usage
0 to **9**	Enter up to 20 digits.

RLT Trunk Group No.

Appears only when **branch** is entered in the **CAS** field. Enter the trunk group number corresponding to the Release Link Trunk (RLT) trunk group to the main location when supporting CAS Branch service.

Field descriptions for page 2

Figure 298: Console Parameters — Default Attendant Group screen

```
change console-parameters                                      Page   2 of x
                              CONSOLE PARAMETERS

TIMING
  Time Reminder on Hold (sec): 10           Return Call Timeout (sec): 10
  Time in Queue Warning (sec):

  INCOMING CALL REMINDERS
      No Answer Timeout (sec): 20                      Alerting (sec): 40
                               Secondary Alert on Held Reminder Calls? y

ABBREVIATED DIALING
     List1: group    1        List2:                  List3:
            SAC Notification? n
                           COMMON SHARED EXTENSIONS
            Starting Extension:                 Count:
```

TIMING

Return Call Timeout (sec)

Valid entries	Usage
10 to **1024** or blank	Enter the time in seconds before a split away call (call extended and ringing a station or otherwise split away from the console) returns to the console. Be sure to allow five seconds for each ring at all points in a coverage path to ensure the entire path is completed before the call returns to the console.

Time In Queue Warning (sec)

Valid entries	Usage
9 to **999** or blank	Enter the number of seconds a call can remain in the attendant queue before activating an alert.

Time Reminder on Hold (sec)

Valid entries	Usage
10 to **1024**	Enter the number of seconds a call can remain on Hold.

INCOMING CALL REMINDERS

Alerting (sec)

Enter the number of seconds after which a held or unanswered call is disconnected from an attendant loop and routed to another attendant or night service

No Answer Timeout (sec)

Enter the number of seconds a call to the attendant can remain unanswered without invoking a more insistent sounding tone. Be sure to allow five seconds for each ring at all points in a coverage path to ensure the entire path is completed before the call returns to the console.

Valid entries	Usage
10 to **1024** or blank	

Secondary Alert on Held Reminder Calls?

Valid entries	Usage
y	Enter **y** to begin attendant alerting for Held Reminder Calls with secondary alerting.
n	Enter **n** to have held reminder calls alert the attendant the same as normal calls. Normal calls start with primary alerting and then switch to secondary alerting when the No Answer Timeout expires.

ABBREVIATED DIALING

List1, List2, List3

You can assign up to 3 abbreviated dialing lists to each attendant. However, you cannot assign a personal list to an attendant

.

Valid entries	Usage
enhanced	Allows the attendant to access the enhanced system abbreviated dialing list.
group	Allows the attendant to access the specified group abbreviated dialing list. You also must enter a group number.
system	Allows the attendant to access the system abbreviated dialing list.

SAC Notification

Valid entries	Usage
y/n	Enables or disables Enhanced Attendant Notification for Send All Calls.

COMMON SHARED EXTENSIONS

Count

Enter a number to indicate the number of consecutive extensions, beginning with the Start Extension to be used as common, shared extensions. For example, if you enter a starting extension of 4300 and a count of 3, the system provides three consecutive extension numbers (4300, 4301, and 4302) for parking calls.

The extensions should be assigned to the optional Attendant Selector Console in the 00 through 09 block (bottom row) in any hundreds group for easy identification by the attendant. The lamp associated with the number will identify "call parked" or "no call parked", instead of busy or idle status.

Valid entries	Usage
1 to **80** or blank	Enter a number to indicate the number of consecutive extensions, beginning with the Start Extension to be used as common, shared extensions.

Starting Extension

These extension numbers can be used by the attendant to park calls.

Field descriptions for page 3

Figure 299: Console Parameters — Default Attendant Group screen

```
change console-parameters                                    Page 3 of x
                                CONSOLE PARAMETERS
QUEUE PRIORITIES
            Emergency Access:1_
             Assistance Call:2_
                     CO Call:2_
            DID to Attendant:2_
                    Tie Call:2_
         Redirected DID Call:2_
             Redirected Call:2_
                 Return Call:2_
                 Serial Call:2_
 Individual Attendant Access:2_
             Interpositional:2_
   VIP Wakeup Reminder Call:2_
          Miscellaneous Call:2_

 Call-Type Ordering Within Priority Levels? n
```

Call-Type Ordering Within Priority Levels?

If you use call-type ordering, calls to the attendant are first grouped by the queue priority level, then by call type, and, finally, in the order received.

Valid entries	Usage
y	Enter **y** if you want to present calls by call type. You can assign a **type-disp** button on the **Attendant Console** screen so that the attendant can review the call type for the active call.
n	Enter **n** if you wish the calls to be queued in chronological order by queue priority level.

QUEUE PRIORITIES

Attendant Priority Queue allows attendants to answer calls by call category (for example, by trunk type). The Attendant Priority Queue handles incoming calls to an attendant when the call cannot be immediately terminated to an attendant. The calling party hears ringback until an attendant answers the call.

You may assign the same priority level to more than one call. Priority 1 is the highest priority and is the default for Emergency Access. Assign a priority level from **1** through **13** to each of the call types.

The attendant call categories are:

- Emergency Access — A call from a telephone user who dials the emergency access code (default is highest-priority level)
- Assistance Call— A call from a telephone user who dials the attendant-group access code, or from a telephone that has the Manual Originating Line Service feature activated
- CO Call — An incoming trunk call (CO/FX/WATS trunk) to an attendant group. This does not include trunk calls that return to the attendant group after a timeout or deferred attendant recall.
- DID to Attendant — An incoming DID trunk call to an attendant group. This does not include trunk calls that return to the attendant group after a timeout or deferred attendant recall.
- Tie Call — An incoming TIE trunk call (dial-repeating or direct types) to an attendant group. This does not include trunk calls that return to the attendant group after a timeout or deferred attendant recall.
- Redirected DID Call — A DID or ACD call that times out due to ring/no-answer, busy condition (if applicable), or Number Unobtainable and reroutes to the attendant group.

- Redirected Call — A call assigned to one attendant, but redirected to the attendant group because the attendant is now busy
- Return Call — A call returned to the attendant after it times out. If the attendant is now busy, the call redirects to the attendant group.
- Serial Call — A call from the Attendant Serial Call feature when an outside trunk call (designated as a serial call by an attendant) is extended to and completed at a telephone, and then the telephone user goes on-hook. If the attendant who extended the call is busy, the call redirects to the attendant group.
- Individual Attendant Access — A call from a telephone user, incoming trunk call, or a system feature to the Individual Attendant Access (IAA) extension of a specific attendant. If the attendant is busy, the call queues until the attendant is available.
- Interposition — A call from one attendant to the Individual Attendant Access (IAA) extension of another attendant
- VIP Wakeup Reminder Call — A VIP Wakeup reminder call.
- Miscellaneous Call — All other calls.

The call types, in descending order of priority, are:

- Type 1 call: outgoing public-network calls receive answer supervision when the Answer Supervision Timer of the trunk group expires, even if the trunk is actually still ringing. Also, incoming calls when answered by the attendant.
- Type 2 call: incoming external public-network calls before they receive answer supervision or before the Answer Supervision Timer of the trunk group expires
- Type 3 call: all other calls (internal calls, conference calls, and tie-trunk calls of any type)

Note that external public-network calls have priority over all other calls including conference calls. And, answered public-network calls have priority over those calls not yet answered.

Field descriptions for page 4

Figure 300: Console Parameters — Queue Priorities screen

```
change console-parameters                                    Page 4 of x
                              CONSOLE PARAMETERS

QUEUE PRIORITIES

      MLPP PRECEDENCE CALL
       Flash Override: 2
                Flash: 3
            Immediate: 4
             Priority: 5
```

QUEUE PRIORITIES

Flash Override

Valid entries	Usage
1 to **17**	Enter the queue priority for Flash Override precedence level calls.

Flash

Valid entries	Usage
1 to **17**	Enter the queue priority for Flash precedence level calls.

Immediate

Valid entries	Usage
1 to **17**	Enter the queue priority for Immediate precedence level calls.

Priority

Valid entries	Usage
1 to **17**	Enter the queue priority for Priority precedence level calls.

Field descriptions for page 5

Note:

If MLPP is not enabled, the MLPP Queues page does not appear, and the following page appears as page 4.

Figure 301: Console Parameters — Default Attendant Group screen

```
change console-parameters                                    Page 5 of x
                              CONSOLE PARAMETERS

  ASSIGNED MEMBERS ( Installed attendant consoles )
      Type        Grp  TN      Type        Grp  TN      Type        Grp  TN
  1:               1   1   11:              1   1   21:              1   1
  2:               1   1   12:              1   1   22:              1   1
  3:               1   1   13:              1   1   23:              1   1
  4:               1   1   14:              1   1   24:              1   1
  5:               1   1   15:              1   1   25:              1   1
  6:               1   1   16:              1   1   26:              1   1
  7:               1   1   17:              1   1   27:              1   1
  8:               1   1   18:              1   1   28:              1   1
  9:               1   1   19:              1   1
 10:               1   1   20:              1   1
```

ASSIGNED MEMBERS (Installed attendant consoles)

Grp

Display-only field that lists the Attendant Group number.

TN

Display-only field that lists the Tenant Partition number.

Type

Display-only field that shows all attendants in the group. You administer the individual attendant consoles on the **Attendant Console** screen.

Coverage Answer Group

This screen establishes Call Coverage Answer Groups.

An answer group contains up to eight members who act as a coverage point for another user. For example, if several secretaries are responsible for answering a department's redirected calls, all the secretaries could be assigned to an answer group. The answer group is assigned a group number, and that group number appears in the department's coverage path. All phones in an answer group ring (alert) simultaneously. Any member of the group can answer the call.

Each coverage answer group is identified by a number from 1 through the maximum number allowed by your system configuration (see *Hardware Guide for Avaya Communication Manager*, 555-245-207). The members of the group are identified by their extension number. Any telephone, including those administered without hardware (X-ported (but not attendants) can be assigned to a coverage answer group. Note that members whose extensions are X-ported will not be alerted).

Field descriptions for page 1

Figure 302: Coverage Answer Group screen

```
change coverage answer-group 3                                  Page 1 of 1
                        COVERAGE ANSWER GROUP

                     Group Number: 3___
                     Group Name: COVERAGE_GROUP_

GROUP MEMBER ASSIGNMENTS
   Ext      Name (first 26 characters)     Ext      Name (first 26 characters)
1: ________ __________________________ 5: ________ __________________________
2: ________ __________________________ 6: ________ __________________________
3: ________ __________________________ 7: ________ __________________________
4: ________ __________________________ 8: ________ __________________________
```

Ext

Enter the extension number (may not be a Vector Directory Number extension) for each member of this coverage answer group.

Valid entries	Usage
An assigned extension for a station.	

Group Name

Enter the group name you want to use to identify this group.

Tip:

Enter the extension numbers that are group members. This allows a list coverage answer group command to be used to list the phones that will be alerted. The list command can be used in conjunction with the list station, list coverage path, and list hunt group commands to determine stations involved in call coverage. This makes it possible to follow call coverage for any extension, allowing the administrator to easily track call coverage paths.

Valid entries	Usage
Up to 27 characters	For example, typing pool, room 12, secy, and so on.

Group Number

A display-only field when the screen is accessed using an administration command such as **add** or **change**.

Name

This display-only field indicates the name assigned when the member’s telephone is administered.

Coverage Path

This screen implements Call Coverage Paths. The screen provides the means to specify the call coverage criteria, the points in the coverage path used to redirect calls, and the number of times a principal's telephone rings before the call redirects to coverage.

Field descriptions for page 1

Figure 303: Coverage Path screen

```
change coverage path 2                                        Page  1 of  x
                                COVERAGE PATH

           Coverage Path Number: 2
                                          Hunt After Coverage: n
               Next Path Number: ___   Linkage: ___  ___

COVERAGE CRITERIA

  Station/Group Status   Inside Call   Outside Call
            Active?            n             n
              Busy?            y             y
     Don't Answer?             y             y Number of Rings:2
               All?            n             n
DND/SAC/Goto Cover?            y             y

COVERAGE POINTS

   Terminate to Coverage Pts. with Bridged Appearance? n

     Point1: __________ Point2: __________ Point3: __________
     Point4: __________ Point5: __________ Point6: __________
```

Coverage Path Number

A display-only field indicating the coverage path being administered.

Hunt After Coverage

Valid entries	Usage
y	Coverage treatment continues by searching for an available station in a hunt chain that begins with the hunt-to-station assigned on the **Station** screen of the last coverage point.
n	Coverage treatment is terminated; the call is left at the last available location (principal or coverage point).

Linkage

Display-only fields that show the (up to) two additional coverage paths in the coverage path chain.

Next Path Number

Enter the next coverage path in a coverage path chain. See "Call Coverage" in *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205, for more information If the coverage criteria of the current coverage path is not satisfied, the system steps down this chain until it finds a coverage path with redirection criteria that matches the call status. If the chain is exhausted before the system finds a match, the call does not redirect to coverage. No path number here indicates that this path is the only path for the principal.

Valid entries	Usage
1 to **9999** or blank	Enter the next coverage path in a coverage path chain.

COVERAGE CRITERIA

COVERAGE CRITERIA are the conditions that, when met, cause the call to redirect to coverage. Assign one of the following:

Valid entries	Usage
Active	Calls redirect if at least one call appearance is busy.
Busy	Calls redirect if all call appearances that accept incoming calls are busy.
Don’t Answer	Calls redirect when the specified number of rings has been exceeded.
All	Calls redirect immediately to coverage and overrides any other criteria with a **y** in this column.
DND/SAC/ Goto Cover	Must be assigned before a user can activate Do Not Disturb (Hospitality Services), Send All Calls (SAC), or Go to Cover features. Allows a calling user, when calling to another internal extension, to redirect a call immediately to coverage by pressing a **Go to Cover** button. Allows a principal temporarily to direct all incoming calls to coverage, regardless of the other assigned coverage criteria by pressing the **Send All Calls** (or **Do Not Disturb**) button. **Send All Calls** also allows covering users to temporarily remove their phones from the coverage path.

Number of Rings

Enter the number of rings.

Valid entries	Usage
1 to **99**	This is the number of rings a user’s telephone rings before the system redirects the call to the first point in the coverage path.

COVERAGE POINTS

Point1, Point2, Point3, Point4, Point5, Point6

The alternate destinations that comprise a coverage path. Coverage points must be assigned sequentially beginning with Point 1 (do not leave gaps). Each path can have up to six coverage points.

Valid entries	Usage
extension	Redirects the call to an internal extension or announcement. **Note:** If entering a Multi-Location Dial Plan shortened extension, note the following: When entering a Multi-Location Dial Plan shortened extension in a field designed for announcement extensions, certain administration end validations that are normally performed on announcement extensions are not done, and resultant warnings or submittal denials do not occur. The shortened extensions also do not appear in any display or list that shows announcement extensions. Extra care should be taken to administer the correct type of announcement for the application when assigning shortened extensions..
attd	Redirects the call to the attendant or attendant group. If the system has Centralized Attendant Service (CAS), the call goes to the CAS attendant.
h1 to **h999**	Redirects the call to the corresponding hunt-group. For example, enter **h32** if you want a coverage point routed to hunt group 32. (See Hunt Group on page 1077 for more information.)

1 of 2

Valid entries	Usage
c1 to **c750** **c1** to **c1000** (For S8300 Media Server, S8700 or S8710 IP-Connect, S8700 or S8710 Multi-Connect)	Redirects the call to the corresponding coverage answer group. For example, enter **c20** if you want a coverage point routed to call coverage answer group 20. (See Coverage Answer Group on page 857 for more information.)
r1 to **r999** **r1** to **r1000** (For S8300 Media Server, S8700 or S8710 IP-Connect, S8700 or S8710 Multi-Connect)	Redirects the call to the corresponding remote coverage point number. For example, enter **r27** if you want a coverage point routed to remote coverage point 27. (See Remote Call Coverage Table on page 1361 for more information.)
v + extension	Redirects the call to the corresponding VDN extension. For example, enter **v12345** if you want the last administered coverage point to be the VDN associated with extension 12345. Note that a Vector Directory Number may be used only as the last administered point in a coverage path.
	2 of 2

If calls redirect to an AUDIX in a DCS network, administer a unique **Hunt Group** screen. Assign the AUDIX extension in the **Group Extension** field. If the AUDIX is connected to the local node, set the **Message Center** field to **audix**; if the AUDIX is connected to another node, set the **Message Center** field to **rem-audix**.

If calls redirect to Message Center (a special Uniform Call Distribution hunt group), AUDIX, or to the attendant, do not list any subsequent coverage points. These calls will normally queue and never redirect to another coverage point. Calls to any hunt group will queue if possible. Calls redirect from a hunt group only if all hunt group members are busy and either the queue is full or there is no queue.

If the Coverage of Calls Redirected Off-Net feature is not enabled, a remote coverage point will function as the last point in the coverage path, because the system will no longer have control of the call once it has redirected off-net. However, if the Coverage of Calls Redirected Off-Net feature is enabled, a call redirected off-net can be monitored by the system and brought back for further call coverage processing.

Terminate to Coverage Pts. with Bridged Appearances

Valid entries	Usage
y	Allows a call to alert as both a bridged call and a redirected call.
n	The call skips the coverage point if it has already alerted as a bridged call.

Crisis Alert System Parameters

This screen allows you to define the system parameters associated with sending crisis alert messages.

Field descriptions

Figure 304: Crisis Alert System Parameters screen

```
change system-parameters crisis-alert
                          CRISIS ALERT SYSTEM PARAMETERS

ALERT STATION
    Every User Responds? n

ALERT PAGER
            Alert Pager? y
  Originating Extension: 7768
      Crisis Alert Code: 911
                Retries: 5
   Retry Interval (sec): 30
            Main Number: 303-555-0800

                      Pager Number          Pin Number
                      1: 3035559001         1: 7614567890
                      2: 123456789012345    2: ppp1234567890pp
                      3: 123456789012345    3: ppp1234567890pp

                 DTMF Duration - Tone (msec): 100   Pause (msec): 100
```

ALERT STATION

Every User Responds

Controls who needs to respond to a crisis alert.

Valid entries	Usage
y	If set to **y**, all users who have a crisis alert button are notified and must clear the alert for every emergency alert. Assign crisis alert buttons only to attendant consoles and stations that must be notified of an emergency call.
n	If set to **n**, all users are notified, but only one user needs to acknowledge an alert. This user may be the attendant or any other digital telephone with a crisis alert button. When the alert is acknowledged by one user, the alert is cleared at all stations except the one that acknowledged the alert.

ALERT PAGER

Alert Pager

Valid entries	Usage
y/n	Enter **y** to use Crisis Alert to a Digital Pager.

Crisis Alert Code

Displays when the **Alert Pager** field is **y**. This field requires an entry before submitting the screen.

Valid entries	Usage
1 through 3 digits	The numbers in this field are the first 3 digits in the crisis alert pager message. Avaya recommends you enter the numbers used to call the local emergency service or any digits used for an emergency situation (for example, 911).

DTMF Duration - Tone (msec)

The length of time the Dual-Tone Multi-Frequency (DTMF) tone is heard for each digit. Displays when the **Alert Pager** field is **y**.

Valid entries	Usage
20 to **2550**	Enter a number in increments of 10.

Main Number

The main telephone number to the location or a location code. This field is optional and does not require an entry. Displays when the **Alert Pager** field is **y**.

Valid entries	Usage
digits **0** to **9** **-** (dash)	Enter a number up to 15 digits to identify the location where the crisis alert call originated. It can be the main number to the location or a numerical identification. Any dashes are for display purposes only and not included in the message sent to the pager. This entry is the last group of digits displayed in the pager message.

Originating Extension

Used as the extension originating the call to send a crisis alert message to a pager. Displays when the **Alert Pager** field is **y**. This field requires an entry before submitting the screen.

Valid entries	Usage
1 to 7 digits	Requires a valid unassigned extension according to the dial plan.

Pager Number

Displays when the **Alert Pager** field is **y**. One of these fields must have a number or the screen cannot be submitted.

Valid entries	Usage
1 to 15 digits **-** (dash)	Any dashes are for display purposes only and not included in the message sent to the pager One of the pager number fields must have a number or the screen cannot be submitted.

Pause (msec)

The length of time between DTMF tones for each digit. Displays when the **Alert Pager** field is **y**.

Valid entries	Usage
20 to **2550**	Enter a number in increments of 10.

Pin Number

This field can be used for any combination of the pager pin number and pauses or left blank. Displays when the **Alert Pager** field is **y**.

Valid entries	Usage
digits **0** to **9** **p**(ause) #(pound) *****(star)	Enter a number up to 15 digits. A pause (about 2 seconds) is for timing of the message. For instance, after the pin number you may want to have a pause to allow time for the pager service to set up the correct pager message box. If the pager service requires you to submit a PIN number, enter it here.

Retries

Displays when the **Alert Pager** field is **y**.

Valid entries	Usage
0 to **10**	The number of times the system tries to send out the alert message in case of an unsuccessful attempt. This increases the chances that the pager receives a crisis alert message.

Retry Interval (sec)

Displays when the **Alert Pager** field is **y**. This field is not used unless the **Retries** field is **1** to **10**.

Valid entries	Usage
30 to **60**	The administrable time period (in seconds) between retries. If an attempt to call the pager fails, the retry call attempts after the retry interval period.

CTI Link

The cti-link commands are available only if, on the **System-Parameters Customer-Options** screen, either the **ASAI Link Core Capabilities and/or Computer Telephony Adjunct Links** field is **y**.

Field descriptions for page 1

Figure 305: CTL Link screen when Type field is ASAI or ADJLK

```
add cti-link next                                              Page 1 of 2
                                  CTI LINK
 CTI Link: 1
Extension: 40001
     Type: ASAI
     Port: 1C0501                                                 COR: 1
     Name: ASAI CTI Link 1









BRI OPTIONS
               XID? y     Fixed TEI? n
       MIM Support? n
                          CRV Length: 2
```

Figure 306: CTI Link screen when Type field is ASAI-IP or ADJ-IP

```
add cti-link next                                              Page 1 of 2
                                  CTI LINK
 CTI Link: 1
Extension: 40001
     Type: ASAI-IP
                                                                 COR: 1
     Name: ASAI CTI Link 1
```

CTI Link

A display-only field indicating the CTI link number.

Valid entries	Usage
1 to **8**	Avaya Communication Manager on a DEFINITY Server CSI DEFINITY G3i
1 to 16	S8300 Media Server Avaya Communication Manager on a DEFINITY Server R S8700 or S8710 Multi-Connect

Extension

This field displays the extension for this link.

Type

For each link that you want to add to your system, you must specify the CTI link type.

Valid entries	Usage
ADJLK	For ASAI links not used by co-resident DLG.
ADJ-IP	For ASAI adjunct links used by co-Resident DLG.
ASAI	For ASAI adjunct links not used by co-resident DLG.
ASAI-IP	For ASAI links used by co-Resident DLG.

Port

Appears when the **Type** field is **ASAI** or **ADJLK**. Enter 7 characters to specify a port, or an x.

Valid entries	Usage
01 to **44** (DEFINITY R)	First and second numbers are the cabinet number
01 to **03** (DEFINITY SI)	
A to **E**	Third character is the carrier
01 to **20**	Fourth and fifth characters are the slot number
01 to **32**	Sixth and seventh characters are the circuit number
x	Indicates that there is no hardware associated with the port assignment. Use for AWOH.

Name

Enter a name associated with this CTI link.

COR

Enter a Class of Restriction (COR) number to select the desired restriction.

BRI Options

XID

Appears when the **Type** field is **ASAI** or **ADJLK**. Used to identify Layer 2 XID testing capability.

MIM Support

Management Information Message Support. A display-only field that appears when the **Type** field is **ASAI** or **ADJLK**.

Fixed TEI

Appears when the **Type** field is **ASAI** or **ADJLK**. It indicates that the endpoint has a fixed Terminal Endpoint Identifier (TEI).

The TEI identifies a unique access point within a service. You must administer TEIs for fixed TEI terminals. However, for terminals with the automatic TEI capability, the system dynamically assigns the TEI.

Valid entries	Usage
y/n	Entering **y** displays the **TEI** field. For ASAI, enter **y**.

CRV Length

Appears when the **Type** field is **ASAI** or **ADJLK**. Enter **1** or **2** to indicate the length of CRV for each interface.

Field descriptions for page 2

Figure 307: Link screen when Type field is ASAI-IP or ADJ-IP

```
add cti-link next                                              Page 2 of 2
                                CTI LINK
FEATURE OPTIONS

    Event Minimization? n      Special Character for Restricted Number? n
```

Event Minimization

May be used when event reports normally would be sent on multiple associations, but the adjunct does not need to see more than one. Typically, these event reports are identical except for the association they are sent over (for example, call control, domain control, or active notification). Some applications discard duplicate events, so in this case, there is no point in sending them across the ASAI CTI link. When enabled, this option allows only a single such event to be sent. The selection of the association on which the event will be sent is based on association precedence as follows: active notification (if enabled), call control (if enabled), or domain control (if enabled). Use the **Station** screen to change this option. The new option settings take effect the next time the ASAI link is activated.

Valid entries	Usage
y/n	Enter **y** to control the behavior for that particular link.

Special Character for Restricted Number

Enables an ASAI CTI link to indicate the calling number restricted presentation within an event report. For further information, see *DEFINITY Enterprise Communications Server CallVisor ASAI Technical Reference*.

Valid entries	Usage
y/n	When set to **y** and a calling number received in a SETUP message has the presentation indicator set (octet 3a in the calling number), then an "*" is appended to the calling party number in the ASAI message.

Data modules

The following section provides descriptions of standard fields on **Data Module** screens. Some of the fields are used for specific data module types; others are used for all data modules. Unique fields and fields requiring special consideration are listed with the appropriate data module descriptions in this book.

Field descriptions for page 1

Figure 308: Data Module screen

```
change data-module 30                                        Page   1 of 2
                                DATA MODULE

   Data Extension: 30            Name: 27                           BCC:
             Type: data-line___   COS: 1
             Port: _______        COR: 1
              ITC: restricted__    TN: 1                  Connected to: dte


 ABBREVIATED DIALING
   List1:

 SPECIAL DIALING OPTION:



 ASSIGNED MEMBER (Station with a data extension button for this data module)

           Ext     Name
      1: 1002    27 character   station name
```

Baud Rate

Used with X.25 Data Modules. The maximum raw data transmission speed.

Valid entries	Usage
300 **1200** **2400** **4800** **9600** **19200**	
switched	You can enter this if the **Cable Type** field is **none** on the **PGATE** screen.

BCC

(Bearer Capability Class) A display-only field used with Data Line, Netcon, Processor Interface, Point-to-Point Protocol, Processor/Trunk (**pdm** selection), and System Port Data Modules. Appears when the **ISDN-PRI** or **ISDN-BRI Trunks** field is **y** on the **System-Parameters Customer-Options** screen. The value in this field corresponds to the speed setting of the data module. This field may be compared with the BCC value in an associated routing pattern when attempted calls utilizing the data module fail to complete. The BCC values must be the same.

See "Generalized Route Selection" in *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205, for a detailed description of Bearer Capability Classes (BCC) and their ability to provide specialized routing for various types of voice and data calls. The BCC value is used to determine compatibility when non-ISDN-PRI facilities are connected to ISDN facilities (ISDN-PRI Interworking).

Valid entries	Usage
1	Relates to 56-bkps
2, 3, 4	Relates to 64 kbps

Board

Used with Announcement Data Modules. Enter the five character announcement circuit pack number that identifies the physical circuit pack to which the announcement module is connected. You can enter **x** in this field to indicate that there is no hardware associated with this port assignment.

The five character announcement board number is comprised of:

Characters	Meaning	Value
1 to 2	Cabinet Number	**01** to **44** (DEFINITY R configurations) **1** to **64** (S8700 or S8710 IP-Connect)
		01 to **03** (DEFINITY SI configurations)
3	Carrier	**A** to **E**
4 to 5	Slot Number or X	**0** to **20**

Broadcast Address

Used with Ethernet data modules. See *Administration for Network Connectivity for Avaya Communication Manager* for more information. Does not appear for S8700 or S8710 IP-Connect.

Connected Data Module

Used with X.25 and Processor Interface (used with DEFINITY CSI and SI only) data modules. See *Administration for Network Connectivity for Avaya Communication Manager* for more information.

Connected to

Used with Data Line and Processor/Trunk (**pdm** selection) Data Module. This field shows to what the Asynchronous Data Unit (ADU) is connected.

Valid entries	Usage
dte	Data Terminal Equipment. Used with Data Line and Processor/Trunk Data Modules.
isn	Information Systems Network. Used with Data Line and Processor/Trunk Data Modules.

COS

Does not appear for **ethernet** or **x.25**. Enter the desired class of service.

Valid entries	Usage
0 to **15**	Select the allowed features.

COR

Does not appear for **ethernet**. Enter the desired class of restriction.

Valid entries	Usage
0 to **95**	Select the allowed restriction.

Data Extension

A display-only field indicating the extension assigned to the data module.

Destination Number

Used with X.25 and Processor Interface data modules. See *Administration for Network Connectivity for Avaya Communication Manager* for more information.

DTE/DCE

Used with X.25 Data Modules. Specifies how the above endpoint type acts.

Valid entries	Usage
dte	Data Terminal Equipment
dce	Data Communications Equipment

Enable Link

Used with X.25, Point-to-Point, and Processor Interface data modules. See *Administration for Network Connectivity for Avaya Communication Manager* for more information. This field is in different locations on the screen for different data module types.

Endpoint Type

Used with X.25 Data Modules. An endpoint type is a type of packet switched data endpoint that uses X.25 call control procedures. The X.25 Endpoint connects to external ports on the PGATE board and to the TDM bus via a DS1 trunk. Ports connected to the adjunct endpoint can be either DTEs or DCEs. The type of endpoint (DTE or DCE) is administrable on the "data-mod" screen.

Valid entries	Usage
adjunct	Mandatory entry.

Error Logging

Used with X.25 Data Modules.

Valid entries	Usage
y/n	Enter **y** to record X.25 protocol errors in the hardware error log. See the *Maintenance for Avaya Communication Manager and DEFINITY Server* manual for your model for more information about error logs.

Establish Connection

Used with X.25, Point-to-Point, and Processor Interface (used with DEFINITY CSI and SI only) data modules. See *Administration for Network Connectivity for Avaya Communication Manager* for more information.

Highest PVC Logical Channel

A display-only field used with X.25 Data Modules. Appears when the **Permanent Virtual Circuit** field is **y**. Specifies how the above endpoint type acts.

Valid entries	Usage
1 to **64**	Indicates that 1 to 64 Logical Channels are allowed on this port.

IP Address Negotiation

Used with Point-to-Point data modules. Does not appear for S8700 or S8710 IP-Connect. See *Administration for Network Connectivity for Avaya Communication Manager* for more information.

ITC

(Information Transfer Capability) Used with 7500, Announcement, data-line, Netcon, Processor/ Trunk (**pdm** selection), Processor Interface, and System Port Data Modules. Appears only when, on the **Trunk Group** screen, the **Comm Type** field is **56k-data** or **64k-data**. Indicates type of transmission facilities to be used for ISDN calls originated from this endpoint. Does not display for voice-only or BRI stations.

Valid entries	Usage
restricted	Either restricted or unrestricted transmission facilities are used to complete the call. A restricted facility is a transmission facility that enforces 1's density digital transmission (that is, a sequence of 8 digital zeros are converted to a sequence of 7 zeros and a digital 1).
unrestricted	Only unrestricted transmission facilities are used to complete the call. An unrestricted facility is a transmission facility that does not enforce 1's density digital transmission (that is, digital information is sent exactly as is).

Link

Used with X.25, Ethernet, Point-to-Point, and Processor Interface (used with DEFINITYCSI and SI only) data modules. See *Administration for Network Connectivity for Avaya Communication Manager* for more information. This field is in different locations on the screen for different data module types.

Valid entries	Usage
1 to **99**	Enter a communication interface link number.

Maintenance Extension

Used with Netcon and Processor Interface Data Modules.

Valid entries	Usage
Enter the extension number required to perform maintenance functions on the standby netcon physical channel in a duplicated system.	The standby remote loop around tests fails if this field is not administered.

MM Complex Voice Ext

Used with 7500 and World Class BRI Data Modules. Does not appear on S8700 or S8710 IP-Connect. This field contains the number of the associated telephone in the multimedia complex. This field appears only after you set the **Multimedia** field to **y**. This field is left blank until you enter the data module extension in **MM Complex Data Ext** on the **Station** screen.

Valid entries	Usage
Valid values conform to your dial plan	Once you complete the field on the **Station** screen, these two extensions are associated as two parts of a one-number complex, which is the extension of the telephone.

Multimedia

Used with the 7500 and World Class BRI Data Modules. Appears only if, on the **System-Parameters Customer-Options** screen, the **MM** field is **y**.

Valid entries	Usage
y/n	Enter **y** to make this data module part of a multimedia complex.

Name

Valid entries	Usage
Up to 27 alphanumeric characters	Enter the name of the user associated with the data module. The name is optional and can be blank.

Network uses 1’s for Broadcast Addresses

Used with Ethernet data modules. See *Administration for Network Connectivity for Avaya Communication Manager* for more information.

Node Name

Used with Ethernet (not on S8700 or S8710 IP-Connect) and Point-to-Point data modules. See *Administration for Network Connectivity for Avaya Communication Manager* for more information.

PDATA Port

Used with System Port Data Modules. Enter a seven-digit alphanumeric port location to which the data module is connected. Used to relate the physical PDATA port to which the mode 3 portion of the system port is connected. This entry must be assigned to a port on a PDATA Line Board.

Valid entries	Usage
01 to **22**	First and second characters are the cabinet number
01 to **64**	First and second characters are the cabinet number (S8700 or S8710 IP-Connect)
A to **E**	Third character is the carrier
01 to **20**	Fourth and fifth characters are the slot number in the carrier
01 to **12**	Sixth and seventh characters are the circuit number

Permanent Virtual Circuit

A display-only field used with X.25 Data Modules.

Valid entries	Usage
y/n	A **y** indicates that the logical channels for PVC are allowed for this port.

Physical Channel

Used with Netcon and Processor Interface Data Modules. The Physical Channel number is referred to on associated system forms as the Interface Link number.

Valid entries	Usage
01 to **08**	For Processor Interface Data Modules, enter the 2-digit circuit number of the Processor Interface port. A multi-carrier cabinet system supports the use of two Processor Interface circuit packs, the first circuit pack (mounted in Control Carrier A) supports physical channels or links 01 through 04; the second (mounted in Control Carrier A) supports physical channels or links 05 through 08. A single-carrier cabinet system supports one Processor Interface circuit pack and physical channels or links 01 through 04 only.
01 to **04**	For DEFINITY CSI configurations. For Netcon Data Modules, enter a netcon data channel.
01 to **08**	For DEFINITY SI configurations. For Netcon Data Modules, enter a netcon data channel.

Port

Used with 7500, Data Line, Ethernet, Processor/Trunk, PPP, System Port, X.25, and World Class BRI Data Modules. Specifies a port location to which the data module is connected.

Characters	Meaning	Value
1-2	Cabinet Number	**01** to **44** (DEFINITY R configurations) **01** to **03** (DEFINITY SI configurations) **01** to **64** (S8700 or S8710 IP-Connect)
3	Carrier	A to E
4-5	Slot Number	0 to 20
6-7	Circuit Number	01 to 04 (x.25 circuit pack) 01 to 31 (DEFINITY SI, S8700 or S8710 IP-Connect (tdm, pdm) configurations) 01 to 16 (ppp for S8700 or S8710 IP-Connect) 01 to 08 (system-port for S8700 or S8710 IP-Connect) 17/33 (ethernet on S8700 or S8710 IP-Connect)

Note:

You can enter **x** in the **Port** field to indicate that there is no hardware associated with the port assignment (also known as Administration Without Hardware (AWOH). These stations are referred to as "phantom stations." If this data module is designated as a secondary data module (Secondary data module set to **y**) An **x** cannot be entered into this field. The port of a primary data module cannot be changed to **x** if a secondary data module is administered.

Remote Loop-Around Test

Used with Processor/Trunk and X.25 Data Modules. Appears when the **Type** field is **pdm**, **tdm**, or **x.25**.

Valid entries	Usage
y/n	For Processor/Trunk Data Modules, enter **y** if the data module supports a loop-back test at the EIA interface. In general, Avaya equipment supports this test but it is not required by Level 2 Digital Communications Protocol. Enter **n** to abort a request for this test.
y/n	For X.25 Data Modules, enter **y** to allow remote loop-around tests on this port. See the *Maintenance for Avaya Communication Manager and DEFINITY Server* manual for your model for more information about remote loop-around tests. Enter **n** to abort a request for this test.

Secondary data module

Used with Processor/Trunk Data Modules. Appears only when the **Type** field is **pdm**. The primary data module must be administered before the secondary data module may be added. If the **Port** field is **x**, the **Secondary Data Module** field cannot be **y**.

Valid entries	Usage
y	This PDM is the secondary data module used for Dual I-channel AUDIX networking.
n	This is the primary PDM, or if this data module is not used for AUDIX networking.

Subnet Mask

Used with Point-to-Point data modules (for S8700 or S8710 IP-Connect). See *Administration for Network Connectivity for Avaya Communication Manager*, 555-233-504, for more information.

Switched Virtual Circuit

A display-only field used with X.25 Data Modules.

Valid entries	Usage
y/n	A **y** indicates that the Switched Virtual Circuit is not allowed for this port.

TN

Valid entries	Usage
1 through **100**	Enter the Tenant Partition number.

Type

Enter the type of data module

Valid entries	Usage
7500	Assigns a 7500 Data Module. The 7500 data module supports automatic TEI, B-channel, maintenance and management messaging, and SPID initialization capabilities. BRI endpoints, both voice and/or data, are assigned to either the ISDN-BRI - 4-wire S/T-NT Interface circuit pack or the ISDN-BRI - 2-wire U circuit pack. Each can support up to 12 ports. Since BRI provides multipoint capability, more than one ISDN endpoint (voice or data) can be administered on one port. For BRI, multipoint administration allows for telephones having SPID initialization capabilities, and can only be allowed if no endpoint administered on the same port is a fixed tie endpoint and no station on the same port has B-channel data capability. Currently, multipoint is restricted to 2 endpoints per port.
announcement	Assigns an announcement data module. The announcement data module is built-in to the integrated announcement circuit pack and is administered using the **Announcement Data Module** screen. This data module (in conjunction with an administered Netcon Data Module in DEFINITY SI configurations) allows the system to save and restore the recorded announcements file between the announcement circuit pack and the system memory.

1 of 4

Valid entries	Usage
data-line	Assigns a Data Line Data Module. The **Data Line Data Module (DLDM)** screen assigns ports on the Data Line circuit pack (DLC) that allows EIA 232C devices to connect to the system. The DLC, with a companion Asynchronous Data Unit (ADU), provides a less expensive data interface to the system than other asynchronous DCP data modules. The DLC supports asynchronous transmissions at speeds of Low and 300, 1200, 2400, 4800, 9600, and 19200 bps over 2-pair (full-duplex) lines. These lines can have different lengths, depending on the transmission speed and wire gauge. The DLC has 8 ports. The connection from the port to the EIA device is *direct*, meaning that no multiplexing is involved. A single port of the DLC is equivalent in functionality to a data module and a digital line port. The DLC appears as a data module to the Digital Terminal Equipment (DTE) and as a digital line port to the server running Avaya Communication Manager. The DLC connects the following EIA 232C equipment to the system: • Printers • Non-Intelligent Data Terminals • Intelligent Terminals, Personal Computers (PCs) • Host Computers • Information Systems Network (ISN), RS-232C Local Area Networks (LANs), or other data switches.
ethernet	Assigns an Ethernet data module. The **Ethernet Data Module** screen assigns the 10BaseT port on the Control-LAN (C-Lan) circuit pack. This port provides a TCP/IP connection to network hub or LAN. See *Administration for Network Connectivity for Avaya Communication Manager* for more information on Ethernet data modules.
netcon	Assigns a Netcon Data Module. Netcon data modules are the Processor Data Modules (PDMs) that are integrated into the system's network control ports that provide asynchronous circuit-switched interfaces to the maintenance and administration terminals, Hospitality journal printers, and CDR digital output. They are characterized by their special locations, that is, special port identifications. **Note:** The Netcon data module is only applicable to DEFINITY Server SI configurations. For DEFINITY Server R and later configurations, use the **`system-port`** command.

Valid entries	Usage
pdm	Assigns a DCE interface for Processor/Trunk Data Modules. These screens assign Modular Processor Data Modules (MPDMs) and Modular Trunk Data Modules (MTDMs). One screen is required for assigning MPDMs (700D), 7400B, 7400D or 8400B Data Module, and another screen for MTDMs (700B, 700C, 700E, 7400A). One screen must be completed for each MPDM, 7400B, 7400D, 8400B or MTDM. The MPDM, 7400B, or 8400B Data Module provides a Data Communications Equipment (DCE) interface for connection to equipment such as data terminals, CDR output devices, on-premises administration terminal, Message Server, Property Management System (PMS), AUDIX, and host computers. It also provides a Digital Communications Protocol (DCP) interface to the digital switch. (DCE is the equipment on the network side of a communications link that provides all the functions required to make the binary serial data from the source or transmitter compatible with the communications channel.) The MTDM provides an Electronic Industries Association (EIA) Data Terminal Equipment (DTE) interface for connection to off-premises private line trunk facilities or a switched telecommunications network and a DCP interface for connection to the digital switch. (DTE is the equipment comprising the endpoints in a connection over a data circuit. For example, in a connection between a data terminal and a host computer, the terminal, the host, and their associated modems or data modules make up the DTE.) The MTDM or 7400A Data Module also can serve as part of a conversion resource for Combined Modem Pooling.
ppp	Assigns a Point-to-Point Protocol data module. The **PPP Data Module** screen assigns a synchronous TCP/IP port on the C-Lan. These ports are tailored to provide TCP/IP connections for use over telephone lines. See *Administration for Network Connectivity for Avaya Communication Manager*, 555-233-504, for more information on Point-to-Point data modules.

Valid entries	Usage
procr-intf	Assigns a Processor Interface Data Module. The Processor Interface data modules are the Processor Data Modules (PDMs) that are integrated into the system's synchronous/asynchronous Processor Interface circuit pack ports. **Note:** The Processor Interface data module is applicable only to DEFINITY SI configurations. They are used to provide the following interfaces: • 3B/Call Management System (maximum of 1) • 3B/Message Server (maximum of 1) • Distributed Communications System (maximum of 8) • AUDIX (maximum of 1) **Note:** Not all maximums can be achieved at the same time Connections for these interfaces are achieved via a digital line port and MPDM combination, and/or in the case of DCS or ISDN-PRI, via a DS1 interface. One direct EIA connection is available (labeled as Processor Interface on the back of the Control Cabinet) for simplex operation. When used, the physical channel assignment must be "01". Use of the EIA connection eliminates the need for one digital line port/MPDM combination. See *Administration for Network Connectivity for Avaya Communication Manager* for more information on Processor Interface data modules.
system-port	Assigns a System Port Data Module.
tdm	Assigns a DTE interface for Processor/Trunk Data Modules. See the **pdm** entry above.
wcbri	Assigns a World Class BRI Data Module.
x.25	Assigns an X.25 Data Module in DEFINITY R configurations for communications to Adjuncts and other nodes in a DCS network. See *Administration for Network Connectivity for Avaya Communication Manager* for more information on X.25 data modules.
	4 of 4

DESTINATION

CHAP

Appears when the **Type** field is **ppp**. Used with Point-to-Point data modules. See *Administration for Network Connectivity for Avaya Communication Manager*, 555-233-504, for more information.

CHAP Secret

Appears when the **CHAP** field is **y**. Used with Point-to-Point data modules. See *Administration for Network Connectivity for Avaya Communication Manager*, 555-233-504, for more information.

Valid entries	Usage
character string	Enter 1 to 30 characters; first character cannot be @.

Digits

Appears when the **Type** field is **ppp**. Used with Point-to-Point data modules. See *Administration for Network Connectivity for Avaya Communication Manager*, 555-233-504, for more information.

Valid entries	Usage
Extension or blank	Enter the extension of the destination or blank.

Node Name

Appears when the **Type** field is **ppp**. Used with Point-to-Point data modules. See *Administration for Network Connectivity for Avaya Communication Manager*, 555-233-504 for more information.

ABBREVIATED DIALING

List1

Used with 7500, Data Line, Netcon, Processor/Trunk, Processor Interface, and World Class BRI Data Modules. Supports Data Hot Line. This field can be left blank.

Valid entries	Usage
e	Enhanced
g	Group. You also must enter a group list number.
p	Personal. You also must enter a personal list number.
s	System.

SPECIAL DIALING OPTION

Used with 7500, Data Line, Netcon, Processor/Trunk, Processor Interface, and World Class BRI Data Modules. Identifies the type of dialing for calls when this data module originates calls.

Valid entries	Usage
hot-line	
default	
blank	For regular (normal) keyboard dialing.

HOT LINE DESTINATION

Abbreviated Dialing Dial Code

Used with 7500, Data Line, Netcon, Processor/Trunk, Processor Interface, and World Class BRI Data Modules. Appears only when the **Special Dialing Option** field is **hot-line**. Entry in this field supports Data Hot Line

Valid entries	Usage
0 to **999**	This number is associated with the AD List. When the user goes off-hook on a Data Hot Line call, the system dials the AD number.

DEFAULT DIALING

Abbreviated Dialing Dial Code

Used with 7500, Data Line, Netcon, Processor/Trunk, Processor Interface, and World Class BRI Data Modules. Only appears when the **Special Dialing Option** field is **default**. When the user goes off-hook and enters a carriage return following the DIAL prompt, the system dials the AD number. The data call originator can also perform data-terminal dialing by specifying a dial string that may or may not contain alphanumeric names.

Valid entries	Usage
0 to **999**	Enter a list number associated with the abbreviated dialing list.

CIRCUIT SWITCHED DATA ATTRIBUTES

Used with 7500 and World Class BRI Data Modules.

Note:

These fields represent defaults needed for modem pooling conversion resource insertion when the endpoint does not support data query capability and administered connections. These fields have no significance for data modules providing data query [all Avaya -supported ISDN-BRI data modules (7500 and ADM)]. For Avaya ISDN-BRI or World Class ISDN-BRI data modules, use the default settings.

Default Duplex

Used with 7500 and World Class BRI Data Modules. Used to identify the duplex mode.

Valid entries	Usage
full	Allows simultaneous two-way transmission.
half	Allows only one transmission direction at a time.

Default Mode

Used with 7500 and World Class BRI Data Modules. Used to identify the data mode.

Valid entries	Usage
sync	Synchronous
async	Asynchronous

Default Speed

Used with 7500 and World Class BRI Data Modules. Used to identify the data rate.

Valid entries	Usage
1200 **2400** **4800** **19200**	
56000 **64000**	Can be entered when the **Default Mode** field is **sync**.

ASSIGNED MEMBER

Ext and Name

Used with Data Line, Announcement, Netcon, Processor/Trunk, Processor Interface, and System Port Data Modules. Displays the extension number and name of the user (previously administered) with associated **Data Extension** buttons who shares the module.

DATA MODULE CAPABILITIES

Default Data Applications

Used with 7500 and World Class BRI Data Modules. Used to identify mode to be Used with originating data calls when the mode is not specified with the calling parameters. This mode is also used for terminating trunk calls that do not have bearer capability specified or administered connections. See "Generalized Route Selection" in *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205, for additional information.

Valid entries	Usage
M0	Mode 0. Use this setting for a WCBRI endpoint used as an administered connection.
M1	Mode 1
M2_A	Mode 2 asynchronous
M2_S	Mode 2 synchronous
M3/2	Mode 3/2 adaptable

Default ITC

Used with 7500 and World Class BRI Data Modules.

Valid entries	Usage
restricted	For a WCBRI endpoint used as an administered connection.
unrestricted	

MM Complex Voice Ext:

Used with 7500 and World Class BRI Data Modules. This display-only field contains the number of the associated telephone in the multimedia complex. It only appears when the **Multimedia** field is **y**. This field is left blank until you enter the data module extension in **MM Complex Data Ext f**ield on the **Station** screen. Once you complete the field on the **Station** screen, these two extensions are associated as two parts of a one-number complex, which is the extension of the telephone. Valid values conform to your dial plan.

Field descriptions for page 2

This version of page 2 appears when the **Type** field is **data-line**.

Figure 309: Data Line Data Module screen— if KYBD Dialing is y

```
change data-module 30                DATA MODULE                Page 2 of 2
        CAPABILITIES
                        KYBD Dialing? y                    Configuration? n
                            Busy Out? n
        SPEEDS
              Low? y           1200? y           4800? y          19200? y
              300? y           2400? y           9600? y     Autoadjust? n
        OPTIONS
                     Permit Mismatch? n                     Dial Echoing? y
                Disconnect Sequence: two-breaks              Answer Text? y
                              Parity: even          Connected Indication? y
```

Figure 310: Data Line Data Module screen — if KYBD Dialing is n

```
change data-module 30                  DATA MODULE                Page 2 of 2
        CAPABILITIES
                        KYBD Dialing? n
                            Busy Out? n
        SPEEDS
              Low? y           1200? y           4800? y          19200? y
              300? y           2400? y           9600? y
        OPTIONS
                     Permit Mismatch? n
```

See DLC Option Settings on page 898 for additional information when assigning entries for the remaining fields on the screen.

CAPABILITIES

Busy Out

This option should be enabled for DTEs that are members of a hunt group and to allow "busy out" when DTE turns power off so that calls do not terminate on that DTE.

Valid entries	Usage
y/n	Enter **y** to place the DLC port in a busied-out state once the DTE control lead to the DLC is dropped.

Configuration

Appears when the **KYBD Dialing** field is **y**. This option normally is enabled for "originate/receive" DTE such as non-intelligent terminals and disabled for intelligent devices such as computers. The **KYBD Dialing** field must be **y** with this option.

Valid entries	Usage
y/n	Enter **y** to allow the viewing and changing of options from the DTE.

KYBD Dialing

This option must be enabled to allow data endpoints to originate calls via the EIA 232C interface and obtain ASCII feedback text. When enabled, the user gets the dial prompt. This option normally is enabled for "originate/receive" DTE that has a need to set up data calls. If this option is disabled, originations cannot be done at the DTE and text feedback does not occur at the DTE during call setup/take down. Data call answering is still allowed but without text feedback.

Note:

ADU-type hunt groups connecting the system to terminal servers on a host computer should have these hunt group extensions assigned as "no" keyboard dialing.

Valid entries	Usage
y/n	Enter **y** to allow keyboard dialing. This enables the data endpoint to receive and transmit text during call origination or termination. Low must be **n**.

SPEEDS

Enter **y** to select operating speeds as follows:

Valid entries	Usage
Low	Enter **y** to instruct the DLC to operate at a low speed from 0 to 1800 bits per second (bps). Enter **n** if the **KYBD Dialing** field is **y**.
300, 1200, 2400, 4800, 9600, or 19200	Enter **y** beside the desired operating speed. Enter **n** if the speed is not desired. The DLC can be any one of these speeds. The speed is matched for the duration of the call, from call setup to call takedown. When multiple speeds are selected (select three or more, do not select just two speeds) and autoadjust is disabled, the DTE's speed must be the highest selected speed. This is required because all feedback text is delivered to the DTE at the highest selected speed.
Autoadjust	Appears when the **KYBD Dialing** field is **y**. Enter **y** which tells the DLC port to automatically adjust to the operating speed and parity of the DTE it is connected to. Enter **n** if this option is not desired. Autoadjust can be selected with any of the speeds selected in the previous step. Autoadjust allows the DLC port to determine the speed and parity of the DTE and then match itself to this speed. Autoadjust only applies to calls originated by the user through Keyboard Dialing.

Answer Text

Appears when the **KYBD Dialing** field is **y**. This option enables text feedback that is normally delivered to the DTE when a call is answered or disconnected. The Answer Text option applies to DLC-generated text as well as text received from the system. If this option is disabled, the system still generates the text, but the DLC prevents it from being sent to the device.

This applies to the following messages:

- INCOMING CALL
- ANSWERED
- DISCONNECTED
- DISCONNECTED OTHER END

This option usually is disabled when the answering DTE is a computer or an intelligent device.

Valid entries	Usage
y/n	Enter **y** to allow text messages to be delivered to the DTE when a call is being answered.

Answer Text

Appears when the **KYBD Dialing** field is **y**. This option enables text feedback that is normally delivered to the DTE when a call is answered or disconnected. The Answer Text option applies to DLC-generated text as well as text received from the system. If this option is disabled, the system still generates the text, but the DLC prevents it from being sent to the device.

This applies to the following messages:

- INCOMING CALL
- ANSWERED
- DISCONNECTED
- DISCONNECTED OTHER END

This option usually is disabled when the answering DTE is a computer or an intelligent device.

Valid entries	Usage
y/n	Enter **y** to allow text messages to be delivered to the DTE when a call is being answered.

OPTIONS

Answer Text

Appears when the **KYBD Dialing** field is **y**. This option enables text feedback that is normally delivered to the DTE when a call is answered or disconnected. The Answer Text option applies to DLC-generated text as well as text received from the system. If this option is disabled, the system still generates the text, but the DLC prevents it from being sent to the device.

This applies to the following messages:

- INCOMING CALL
- ANSWERED
- DISCONNECTED
- DISCONNECTED OTHER END

This option usually is disabled when the answering DTE is a computer or an intelligent device.

Valid entries	Usage
y/n	Enter **y** to allow text messages to be delivered to the DTE when a call is being answered.

Connected Indication

Appears when the **KYBD Dialing** field is **y**. This option generates a "CONNECTED" message to the DTE when the connection has been established. If the **KYBD Dialing** field is **n**, the connected indication is provided by the DLC activating its EIA 232C control lead.

Valid entries	Usage
y/n	Enter **y** to select this option.

Dial Echoing

Appears when the **KYBD Dialing** field is **y**.

Valid entries	Usage
y/n	Enter **y** to echo characters back to the DTE. Dial echoing should be disabled when keyboard dialing is done by an intelligent device.

Disconnect Sequence

Appears when the **KYBD Dialing** field is **y**. Selects the sequence for a disconnect.

Valid entries	Usage
long-break	A long-break is greater than 2 seconds.
two-breaks	Two-breaks is within 1 second.

Parity

Appears when the **KYBD Dialing** field is **y**. Select the desired type of parity. The DLC generates the parities when call setup text is sent to the DTE. The DLC does not check the parity when receiving dialing characters. Parity has nothing to do with the far end; it is used by the DLC to terminal communications during call setup. Set to match the connected DTE.

Valid entries	Usage
even	Set to match the connected DTE.
odd	
mark	
space	

Permit Mismatch

This option allows the EIA interface to operate at a rate different than that agreed to in the data module handshake. (The data module handshake is always the highest compatible rate as determined by the reported speed option of each data module.) Permit Mismatch eliminates the need to change the DTE/DLC speed every time a call is placed to/from an endpoint operating at a different speed. When this option is enabled, the DLC reports the highest optioned speed and all the lower speeds (or the previously selected autoadjust speed) during the handshake process.

Valid entries	Usage
y/n	Enter **y** to instruct the DLC to operate at the highest selected speed, which is a higher rate than the far-end data module.

CAUTION:

Caution must be used when using this option to send information from a DTE/DCE that is transmitting data at higher rates than that of the far end. Sustained usage of this type transmission results in loss of data. Whenever this option is enabled, the DTE must match the highest speed selected for the associated DLC port.

This option is intended to be used by a DTE device operating locally at a higher baud rate than that of its far-end connection but transmitting relatively low amounts of data (for example, a user typing at a terminal). Also, this option may be selected whether Keyboard Dialing is selected.

Note:

The Low speed setting is not reported as an available speed when the **Permit Mismatch** field is **y**.

DLC Option Settings

The following provides additional information on the option settings for DLCs when used with the following types of devices:

- Printers
- Non-intelligent terminals
- Data terminals and personal computers
- Host computers
- Information Systems Network (ISN)

Printers

A DLC port with a companion ADU, when attached to a printer, usually terminates a data call. Therefore, in this connection, the printer is the endpoint device. The originating device may be attached to a DCP mode 2 data module (such as the MPDM) or the DLC. A Z3A ADU extends the range of the EIA 232C connection.

When a receive-only printer (or any printer that does not generate the Transmit Data and DTR leads) is used, the ADU must be powered from a small plug-mounted transformer (2012D, or equivalent) connected to pins 7 and 8 of the modular jack. (See **ADU User Manual** for details.)

An ADU cannot be used if the printer has hardware flow control using the Clear To Send (CTS) lead. An ADU can be used, however, if the printer is using software flow control.

A printer connected to a DLC is usually assigned as a line. Table 9 lists the option settings for printer connections.

Table 9: DLDM screen settings for printer connection *1 of 2*

Field on screen	Option	Comments
Speed	Highest speed at which the Printer operates	Subject to distance limitations; Autoadjust not used
KYBD Dialing	no	
Busy Out	yes	If printer is member of Hunt Group
Permit Mismatch	yes	No, if printer is low speed
Parity	-	Don’t care
Dial Echoing	-	Don’t care

1 of 2

Table 9: DLDM screen settings for printer connection *2 of 2*

Field on screen	Option	Comments
Disconnect Sequence	-	Don’t care
Answer Text	-	Don’t care
Connected Indication	-	Don’t care
Configuration	no	
		2 of 2

Non-intelligent terminals

A non-intelligent terminal connected to the DLC usually is assigned as a line. Table 10 lists the option settings for non-intelligent terminals.

Table 10: DLDM screen settings for connection to non-intelligent terminals

Field On screen	Option	Comments
Speed	All speeds at which the terminal can operate; autoadjust	Subject to distance limitations; Autoadjust when the **KYBD Dialing** field is **y** and the Terminal can generate an ASCII "return"
KYBD Dialing	yes	
Busy Out	no	Yes, if terminal is member of a hunt group
Permit Mismatch	yes	-
Parity	Same as DTE	
Dial Echoing	yes	Only if the **KYBD Dialing** field is **y**
Disconnect Sequence	2	Depends on terminal
Answer Text	yes	
Connected Indication	-	Don’t care
Configuration	yes	

Data terminals and personal computers

An intelligent data terminal or a personal computer (PC) attached to a DLC can either originate or terminate a data call. A single ADU at the site of the originating device extends the distance signals can travel to the switch (the model ADU depends on the terminal connector). An analog telephone can be attached to this arrangement whenever an ADU uses the standard building wiring. Table 11 lists the option settings used for data terminal and personal computer connections.

Table 11: DLDM screen settings for connection to data terminal or personal computer

Field on screen	Option	Comments
Speed	All speeds at which the Data Terminal or PC can operate	Subject to distance limitations; Autoadjust not used
KYBD Dialing	yes	
Busy Out	no	Yes, if device is accessed through a hunt group
Permit Mismatch	yes	No, if device does not support XON/XOFF flow control
Parity	Same as DTE	
Dial Echoing	no	These devices can dial in the ASCII stream without human intervention
Disconnect Sequence	Long <BREAK>	-
Answer Text	no	These devices may not want to see any text
Connected Indication	-	Don’t care
Configuration	yes	

Host computers

A host computer may originate and terminate a data call. For this application, the number of DLCs required depends on the number of ports needed. An MADU can be used (instead of 8 ADUs) to complete the connection. Table 12 lists option settings for a port that has a terminating connection to a host computer or an originating connection from a host computer.

Note:

If the **KYBD Dialing** field is **n**, the rest of the option settings are irrelevant.

Table 12: DLDM screen settings for terminating connection to host computer

Field on screen	Option	Comments
Speed	All speeds at which the computer can operate	Subject to distance limitations; Autoadjust not used
KYBD Dialing	no	
Busy Out	-	Don’t care
Permit Mismatch	-	Don’t care
Parity	-	Don’t care
Dial Echoing	-	Don’t care
Disconnect Sequence	-	Don’t care
Answer Text	-	Don’t care
Connected Indication	-	Don’t care
Configuration	-	Don’t care

This version of page 2 does not appear for S8700 or S8710 IP-Connect.

Figure 311: X.25 Adjunct Data Module screen

```
change data-module 30                                          Page 2 of 2
                                  DATA MODULE
   LAYER 2 PARAMETERS
            Number of Outstanding Frames (w): 4
                  Retry Attempt Counter (N2): 2
                              Frame Size (N1): 135
    Retransmission (T1) Timer (1/10 seconds): 10
              Idle (T4) Timer (1/10 seconds): 30
   LAYER 3 PARAMETERS
     Number of Outstanding Packets: 2
     Restart (T20) Timer (seconds): 8
       Reset (T22) Timer (seconds): 10
```

LAYER 2 PARAMETERS

Frame Size (N1)

Specifies the number of bytes in a frame.

Valid entries	Usage
135 **263**	If the value is **135**, there can be up to 1080 bits within a frame. This value is suitable for all adjuncts and for DCS.

Idle (T4) Timer (1/10 seconds)

The T4 timer is a system parameter that represents the time a DTE allows without frames being exchanged on the data link.

Valid entries	Usage
0 to **250**	Enter the time a DTE allows without frames being exchanged on the data link.

Number of Outstanding Frames (w)

Specifies layer 2 window size.

Valid entries	Usage
1 to **7**	If you enter **2**, up to 2 frames can be sent without confirmation.

Retransmission (T1) Timer (1/10 seconds)

The T1 timer is started at the beginning or the end of the transmission of a frame. At the end of this timer, retransmission of a frame is initiated according to the procedures for link set-up and disconnection or information transfer.

Valid entries	Usage
0 to **250**	Enter the time, in 1/10 seconds, at the end of which retransmission of a frame is initiated

Retry Attempt Counter (N2)

Valid entries	Usage
0 to **7**	Specify the number of times to send one frame when this frame is not confirmed for a period of time.

LAYER 3 PARAMETERS

Number of Outstanding Packets

Valid entries	Usage
2 to **7**	Specify the number of packets that can be sent without confirmation.

Reset (T22) Timer (seconds)

The T22 timer is a DTE time-limit started when DTE issues a reset indication and terminated when the reset request is received or confirmed.

Valid entries	Usage
0 to **500**	

Restart (T20) Timer (seconds)

The T20 timer is a DTE time-limit started when DTE issues a restart indication and terminated when the restart request is received or confirmed.

Valid entries	Usage
0 to **500**	

Figure 312: 7500 and World Class BRI Data Module screen

```
change data-module 30                                          Page 2 of 2
                                  DATA MODULE
  BRI LINK/MAINTENANCE PARAMETERS
                   XID? y      Fixed TEI? n      TEI: ___
          MIM Support? y     Endpt Init? y     SPID: 300_____   MIM Mtce/Mgt?
y
```

BRI LINK/MAINTENANCE PARAMETERS

Country Protocol

Used with World Class BRI Data Modules. Enter the protocol that corresponds to your supported initialization and codesets. The Country Protocol must match any previously-administered endpoint on the same port. The following table lists the valid protocol entries.

Country/Area	Protocol
Australia	2
ETSI (Europe)	etsi
Japan	3
Singapore	6
United States (Bellcore National ISDN)	1

Endpt ID

Used with World Class BRI Data Modules. Appears only if the **Endpt Init** field is **y**. This field provides for multipoint configuration conformance to the Bellcore Terminal Initialization procedures. In these procedures, a multipoint configuration requires that the last 2 digits of the Service Profile Identifier (SPID) be between **00** and **63** and be binary unique for each endpoint. This field, combined with the SPID, gives the effective SPID administered into the terminal. Bellcore ISDN-1 requires that the SPID programmed into the endpoint contain at least 9 digits. (For example, if the **SPID** field is **1234**, and the **Endpt ID** field is set to **01**, then the SPID administered on the terminal is 000123401. The three leading zeros are necessary to create a 9-digit SPID.)

Valid entries	Usage
00 to **62**	Enter a 2-digit number. Each **Endpt ID** field must have a unique value for each endpoint on the same port.

Endpt Init

Used with 7500 and World Class BRI Data Modules. Endpoint initialization is a procedure, required for multipoint operation, by which User Service Order Profile (USOP) is associated with an endpoint on the ISDN-BRI. This association is made via the Service Profile Identifier (SPID), administered into the system and entered into the ISDN-BRI terminal. For a ISDN-BRI terminal to become operational in a multipoint configuration, both the administered SPID and the SPID programmed into the ISDN-BRI terminal must be the same. This means that the SPID of the new or re-used terminals must be programmed to match the administered SPID value.

Valid entries	Usage
y/n	Indicates the terminal's endpoint initialization capability.

Fixed TEI

Used with 7500 and World Class BRI Data Modules. Used to indicate whether the endpoint has Fixed Terminal Equipment Identifier (TEI) capability. TEI identifies a unique access point within a service. For Fixed TEI stations, the TEI must be administered. Terminals with automatic TEI capability, the associated TEI is assigned by the system.

Valid entries	Usage
y/n	Enter **y** to indicate the endpoint has Fixed Terminal Equipment Identifier (TEI) capability.

MIM Mtce/Mgt

Used with 7500 Data Modules.

Valid entries	Usage
y/n	Entering **y** indicates the terminal supports MIM Maintenance and Management capabilities, other than endpoint initialization.

MIM Support

Used with 7500 Data Modules. Management Information Message Support. Used to support two types of capabilities: MIM endpoint initialization capability (SPID support), and other Maintenance/Management capability.

Valid entries	Usage
y/n	

SPID

Used with 7500 and World Class BRI Data Modules. Appears only if the **Endpt Init** field is **y**. The Service Profile Identifier (SPID) is a variable parameter of up to 10 digits. The SPID must be different for all terminals on the ISDN-BRI and from the Service SPID. The SPID should always be assigned. If the SPID is not assigned for the first ISDN-BRI on a port, any other ISDN-BRI assignment to that port is blocked.

Valid entries	Usage
0 to **9999999999**	

TEI

Used with 7500 and World Class BRI Data Modules. Appears only if the **Fixed TEI** field is **y**.

Valid entries	Usage
0 to **63**	Enter a 1 to 2-digit number.

XID

(Exchange identification) Used with 7500 and World Class BRI Data Modules. Used to identify layer 2 XID testing capability.

Valid entries	Usage
y/n	Avaya recommends setting to **n**.

Date and Time

Use this screen to set the system date and time, to select the daylight savings plan number, if any, and to show whether the current time is standard time or daylight savings. Settings on this screen affect the internal clock and timestamp of the server running Avaya Communication Manager. You should update the date and time for a leap year or a system restart after a power failure. The correct date and time assure that CDR records are correct. CDR does not work until the date and time have been entered.

For additional information, see the *Avaya Communication Manager Contact Center Guide to ACD Contact Centers*.

Field descriptions for page 1

Figure 313: Date and Time screen

```
set time
                          DATE AND TIME
DATE
      Day of the Week: _______          Month: _________
     Day of the Month: __                Year: _____

TIME
     Hour: __ Minute: __      Second: __                    Type:
_________________

            Daylight Savings Rule: _

WARNING: Changing the date or time may impact BCMS, CDR, SCHEDULED EVENTS,
         and MEASUREMENTS
```

Day of the Month

Valid entries	Usage
1 to **31**	Enter the current day of the month. The system clock uses this as the current date.

Day of the Week

Valid entries	Usage
Sunday through **Saturday**	Enter the current day of the week. The system clock uses this as the current day.

Daylight Savings Rule

This field displays which daylight savings rule is in use for your system.

Valid entries	Usage
0 to **15**	Enter the appropriate rule number. The system clock uses this as the current daylight savings rule. These rules are defined on the **Daylight Savings Rules** screen.

Hour

The system uses a 24-hour clock. For example, 14:00 is the same as 2:00 p.m.

Valid entries	Usage
0 to **23**	Enter the current hour to be used by the system clock.

Minute

Valid entries	Usage
0 to **59**	Enter the current minute. The system clock uses this as the current minute.

Month

Valid entries	Usage
January through **December**	Enter the current month. The system clock uses this as the current month.

Second

This display-only field shows the seconds and cannot be modified. It resets to zero when you save the information on this screen.

Type

Valid entries	Usage
daylight-savings	Enter **daylight-savings** to indicate daylight savings time is in effect.
standard	Enter **standard** to indicate standard time is in effect.

Year

Valid entries	Usage
1990 to **2099**	Enter the current year. The system clock uses this as the current year.

Related topics

To update the date and time for the change to or from daylight savings time, use the **Daylight Saving Rule** screen. See Establishing Daylight Savings Rules on page 31 for instructions on how to set up daylight savings rules.

Daylight Savings Rules

Use this screen to enter up to 15 customized daylight savings rules. You can specify the day, month, date, time, and increment each daylight savings rule goes into effect and the day, month, date, and time it stops. Rule 0 makes no adjustment to the system clock for daylight savings and cannot be modified. Telephone displays are affected by these settings.

Field descriptions for page 1

Figure 314: Daylight Savings Rules screen

```
change daylight-savings-rules                                  Page 1 of 2
DAYLIGHT SAVINGS RULES
Rule      Change Day                                Month        Date
Time    Increment

0:   No Daylight Savings

1:   Start: first _________ on or after _________ __ at __:__   _________
     Stop:  first _________ on or after _________ __ at __:__
2:   Start: first _________ on or after _________ __ at __:__
     Stop:  first _________ on or after _________ __ at __:__
3:   Start: first _________ on or after _________ __ at __:__
     Stop:  first _________ on or after _________ __ at __:__
4:   Start: first _________ on or after _________ __ at __:__
     Stop:  first _________ on or after _________ __ at __:__
5:   Start: first _________ on or after _________ __ at __:__
     Stop:  first _________ on or after _________ __ at __:__
6:   Start: first _________ on or after _________ __ at __:__
     Stop:  first _________ on or after _________ __ at __:__
7:   Start: first _________ on or after _________ __ at __:__
     Stop:  first _________ on or after _________ __ at __:__
```

Change day (Start)

Valid entries	Usage
Sunday through **Saturday** or **Day**	Enter the day of the week you want the clock to move ahead to begin daylight savings. If you enter **Day** in this field, the clock will change on the exact date entered in the next two fields.

Change day (Stop)

Valid entries	Usage
Sunday through **Saturday** or **Day**	Enter the day of the week you want the clock to move back to return to standard time. If you enter **Day** in this field, the clock will change on the exact date entered in the next two fields.

Date (Start)

Valid entries	Usage
0 to **31**	Enter the day of the month you want the clock to move ahead to begin daylight savings.

Date (Stop)

Valid entries	Usage
0 to **31**	Enter the date you want the clock to move back to return to standard time.

Increment (Start)

Valid entries	Usage
0 to **23**	Enter the number of hours you want the clock to move ahead for daylight savings and to move back to return to standard time.
0 to **9**	Enter the number of minutes you want the clock to move ahead for daylight savings and to move back to return to standard time.

Month (Start)

Valid entries	Usage
January through **December**	Enter the month you want the clock to move ahead to begin daylight savings.

Month (Stop)

Valid entries	Usage
January through **December**	Enter the month you want the clock to move back to return to standard time.

Rule

This display-only field indicates the daylight savings rule number.

Time (Start)

The system uses a 24-hour clock. For example, 14:00 is the same as 2:00 p.m.

Valid entries	Usage
0 to **23**	Enter the hour you want the clock to move ahead to begin daylight savings.
0 to 59	Enter the minute you want the clock to move ahead to begin daylight savings.

Time (Stop)

The system uses a 24-hour clock. For example, 14:00 is the same as 2:00 p.m.

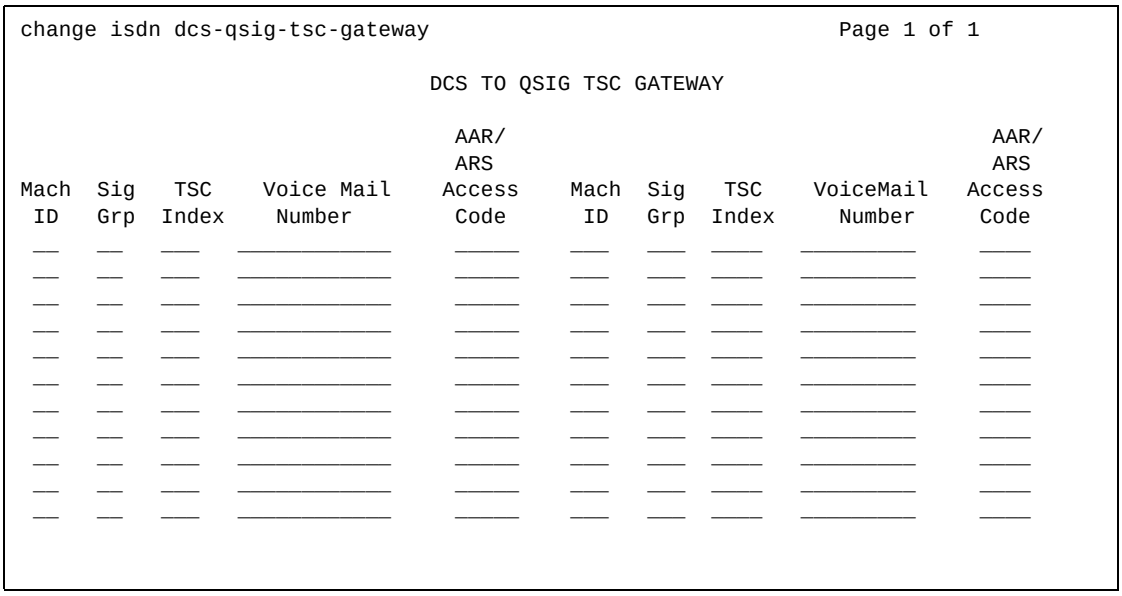
Valid entries	Usage
0 to **23**	Enter the hour you want the clock to move back to return to standard time.
0 to **59**	Enter the minute you want the clock to move back to return to standard time.

DCS to QSIG TSC Gateway screen

Use the **DCS to QSIG TSC Gateway** screen to determine when and how to convert messages from an administered AUDIX NCA-TSC to a QSIG NCA-TSC. This screen maps the AUDIX NCA-TSC to the appropriate machine ID index to find the QSIG subscriber entry in the **QSIG MWI-Prefix** screen. It also assigns the voice mail number to be used when a DCS served-user node interrogates a QSIG message center.

This screen only appears if the Interworking with **DCS** field is enabled on the **System-Parameters Customer-Options** screen.

Figure 315: DCS to QSIG TSC Gateway screen



```
change isdn dcs-qsig-tsc-gateway                                Page 1 of 1

                                DCS TO QSIG TSC GATEWAY

                                  AAR/                                    AAR/
                                  ARS                                     ARS
Mach  Sig   TSC    Voice Mail    Access    Mach  Sig   TSC    VoiceMail   Access
 ID   Grp  Index    Number        Code      ID   Grp  Index    Number     Code
 __   __   ___   ____________    _____    ___   ___  ____   _________    ____
 __   __   ___   ____________    _____    ___   ___  ____   _________    ____
 __   __   ___   ____________    _____    ___   ___  ____   _________    ____
 __   __   ___   ____________    _____    ___   ___  ____   _________    ____
 __   __   ___   ____________    _____    ___   ___  ____   _________    ____
 __   __   ___   ____________    _____    ___   ___  ____   _________    ____
 __   __   ___   ____________    _____    ___   ___  ____   _________    ____
 __   __   ___   ____________    _____    ___   ___  ____   _________    ____
 __   __   ___   ____________    _____    ___   ___  ____   _________    ____
 __   __   ___   ____________    _____    ___   ___  ____   _________    ____
```

AAR/ARS Access Code

This field can be left blank.

Valid entries	Usage
0 to **9**, *****, #	Enter up to 4-digit access code.

Mach ID

You can enter up to 20 machine IDs.

Valid entries	Usage
1 to **20**	Enter a unique machine ID. The system does not allow you to specify an ID that you already entered on the **Processor Channel** screen.

Sig Grp

You must complete the **Signaling Group** field for each machine ID.

Valid entries	Usage
1 to **110**	Enter the assigned signaling group number between 1 and 110 for DEFINITY SI and CSI.
1 to **416**	Enter the assigned signaling group number between 1 and 416 for DEFINITY R
1 to **650**	Enter the assigned signaling group number between 1 and 650 for S8300 Media Server, S8700 or S8710 IP-Connect, S8700 or S8710 Multi-Connect.

TSC Index

You must complete the **TSC Index** field for each machine ID.

Valid entries	Usage
1 to **64**	Enter the assigned signaling group number for **qsig-mwi** application type on the **Signaling Group** screen.

Voice Mail Number

This field can be left blank.

Valid entries	Usage
0 to **9**	Enter the complete Voice Mail Dial Up number up to 15 digits.

Dial Plan Analysis Table

The Dial Plan Analysis Table is the system's guide to translating the digits dialed by users. This screen enables you to determine the beginning digits and total length for each type of call that Avaya Communication Manager needs to interpret. The **Dial Plan Analysis Table** and the **Dial Plan Parameters** screen work together to define your system's dial plan.

Figure 316: Dial Plan Analysis Table screen

```
display dialplan analysis                                Page 1 of 12
                          DIAL PLAN ANALYSIS TABLE
                                                          Percent Full: 7

      Dialed  Total  Call        Dialed  Total  Call        Dialed  Total  Call
      String  Length Type        String  Length Type        String  Length Type
        00      2    attd
        1       3    dac
        2       4    ext
        3       4    ext
        3       1    aar
        4       1    ars
        4       5    ext
        5       5    ext
        5       7    ext
        6       5    ext
        7210    7    ext
        8       7    ext
        9       1    fac
        *       3    fac
        #       3    fac
```

Call Type

Valid entries	Usage
aar	Automatic Alternate Routing — Used to route calls within your company over your own private network. In order to use this code in your dial plan , the ARS/AAR Dialing without FAC feature must be enabled on the **System Parameters Customer Options** screen. (Contact your Avaya technical support representative to discuss the ARS/AAR Dialing Without FAC feature before enabling it.) When dialing digits of Call Type aar, as soon as the dialed digits have reached the administered length, the digits are treated as if an AAR feature access code (FAC) was dialed. Control is transferred and the digits are routed according to the AAR Analysis and Digit Conversion forms. In the example shown on the Dial Plan Analysis Table on page 916, extensions of 3xxx cannot be dialed directly. Whenever a user dials the first digit of 3, the system immediately interprets the dialed string as an AAR string and transfers control to AAR. Extensions of 3xxx can only be accessed using AAR Digit Conversion. That is, you must dial a longer AAR number from which AAR Digit Conversion deletes leading digits to form a number of the form 3xxx.
ars	Automatic Route Selection — Used to route calls that go outside your company over public networks. ARS is also used to route calls to remote company locations if you do not have a private network. In order to use this code in your dial plan , the ARS/AAR Dialing without FAC feature must be enabled on the **System Parameters Customer Options** screen. (Contact your Avaya technical support representative to discuss the ARS/AAR Dialing Without FAC feature before enabling it.) When dialing digits of Call Type ars, as soon as the dialed digits have reached the administered length, the digits are treated as if an ARS feature access code (FAC) was dialed. Control is transferred and the digits are routed according to the ARS Analysis and Digit Conversion forms. In the example shown on the Dial Plan Analysis Table on page 916, extensions of 4xxxx cannot be dialed directly. Whenever a user dials the first digit of 4, the system immediately interprets the dialed string as an ARS string and transfers control to ARS. Extensions of 4xxxx can only be accessed using ARS Digit Conversion. That is, you must dial a longer ARS number from which ARS Digit Conversion deletes leading digits to form a number of the form 4xxxx.

1 of 3

Valid entries	Usage
attd	Attendant — Defines how users call an attendant. Attendant access numbers can start with any number from 0 to 9 and contain 1 or 2 digits. If a telephone's COR restricts the user from originating calls, this user cannot access the attendant using this code. Beginning with the November 2003 release of Communication Manager (2.0), you can also administer the attendant access code by entering an appropriate **fac** or **dac** entry on the **Dial Plan Analysis** screen, and then entering the actual access code on the **Feature Access Codes** screen. Location-specific attendant access codes can be administered on the **Locations** screen.
dac	Dial access code — Allows you to use trunk access codes (TAC) and feature access codes (FAC) in the same range. Dial access codes can start with any number from 0 to 9, * or # and can contain up to 4 digits. If an extension entry and a DAC entry have the same Dialed String, the extension entry can be longer than the DAC entry only if all of the trunk groups covered by that DAC entry have Dial Access on the **Trunk Group** screen set to "no." You can use the DAC to activate or deactivate a Communication Manager feature or to seize a trunk from a trunk group, or both. In the first case, the DAC functions as a FAC, in the second as a TAC. For example, you can define the group 300 to 399 for dial access codes, and allow both FAC and TAC in that range. You can use 4-digit DACs for ordinary trunk access, but they do not work for attendant control of trunk groups, trunk-ID buttons, or DCS, and only the last 3 digits of the codes can be recorded in CDR records. See also the description below for **fac**.
ext	Primary extension — Defines extension ranges that can be used on your system. Extension can have a first digit of 0 through 9 and can be 1 to 7 digits in length. Extension cannot have the same first digit as a 1-digit ARS or AAR feature access code (FAC). When a dial plan has mixed station numbering, extensions of various lengths (all with the same first digit) are mapped on the Dial Plan Analysis table. The system then employs an inter-digit time-out to ensure that all dialed digits are collected.

2 of 3

Valid entries	Usage
fac	Feature access code only — A FAC can be any number from 1 to 9 and contain up to 4 digits. You can use * or #, but only as a first digit. Avaya recommends that a FAC have the longest total length for a given dialed string when using mixed numbering. Otherwise, problems may occur when, for example, 3-digit FACs and 4-digit extensions begin with the same first digit and the FAC is an abbreviated dialing list access code. However, if the entry in the dial plan that defines the FAC is used to define the AAR or ARS access code, then it *must* have the longest total length in the dial plan.
pext	Prefixed extension — Is made up of a prefix (first digit) that can be a **0** to **9** (* and # not allowed) and an extension number of up to 5 digits in length. The maximum length of a prefix and extension combination is 6 digits. You cannot administer a dial access code with the same first digit as a prefixed extension. The purpose of the prefix is to identify the call type as an extension. After digit collection, the prefix digit is removed from the string of dialed digits. The remaining digits (extension number) are then processed. A prefixed extension allows the use of extensions numbers with any dialed string (the extension length must be specified on the table). The "prefixed extension" cannot have the same dialed string as the ARS or AAR facility access code (FAC).

3 of 3

Dialed String

The dialed string contains the digits that Avaya Communication Manager will analyze to determine how to process the call. This field allows you to enter up to four digits, so you can allocate blocks of 1000 numbers even when using a 7-digit dial plan

Valid entries	Usage
0 to **9**, ***** and #	Enter any combination of 1 to 4 digits. the following restrictions apply: • The digits * and # may only be used as first digits, and only for the Call Types **fac** and **dac**. • For Call Type **attd**, if the Total Length is 2, the Dialed String must be 2 digits long. • Two Dial Plan entries can use the same Dialed String only if the Dialed String is 1 digit long. Longer Dialed Strings must all be unique. • A new entry cannot be administered if it causes an existing extension, feature access code, or trunk access code to become inaccessible.

Percent Full

Displays the percentage (0 to 100) of the system's memory resources that have been allocated for the dial plan that are currently being used.

Total Length

Valid entries	Usage
1 to **2** for **attd** **1** to **4** for **dac** **1** to **4** for **fac** **1** to **7** for **ext** **2** to **6** for **pext**	Enter the number of digits for this call type. The allowed length varies by call type. This must be greater than or equal to the number of digits in the Dialed String.

Dial Plan Parameters

The **Dial Plan Parameters** screen works with the **Dial Plan Analysis Table** to define your system's dial plan.

It also controls the appearance of 6- and 7-digit extensions on station displays. 6- and 7-digit extensions may be hard to read as a block. Avaya Communication Manager allows you to select a single system-wide format for all 6-digit extensions and a format for all 7-digit extensions.

Figure 317: Dial Plan Parameters screen

```
change dialplan parameters                                      Page 1 of 1
                            DIAL PLAN PARAMETERS

                 Local Node Number:  2
                   ETA Node Number:
               ETA Routing Pattern:
        UDP Extension Search Order: local-extensions-first
  6-Digit Extension Display Format: xx.xx.xx
  7-Digit Extension Display Format: xxx-xxxx
      AAR/ARS Internal Call Prefix:
AAR/ARS Internal Call Total Length:
```

6-Digit Extension Display Format

Determines how the system punctuates 6-digit extensions.

Valid entries	Usage
xx.xx.xx	This field can contain all 'x' characters (no punctuation) or you can use a combination of 6 'x' characters and 0 to 2 hyphens (-), spaces, or periods (.) to depict how 6-digit extensions will display. There must be 6 'x' characters and no more than 2 punctuation marks. You must specify a format or accept the default. You cannot leave this field blank.

7-Digit Extension Display Format

Determines how the system punctuates 7-digit extensions.

Valid entries	Usage
xxx-xxxx	This field can contain all ‘x’ characters (no punctuation) or you can use a combination of 7 ‘x’ characters and 0 to 1 hyphen (-), space, or period (.) to depict how 7-digit extensions will display. There must be 7 ‘x’ characters and no more than 1 punctuation mark. You must specify a format or accept the default. You cannot leave this field blank.

AAR/ARS Internal Call Prefix

The digits entered in this field are concatenated with the calling or called extension. Appears only if, on the **System-Parameters Customer-Options** screen, the **ARS/AAR Dialing Without FAC** field is **y**. (Contact your Avaya technical support representative to discuss the ARS/AAR Dialing Without FAC feature before enabling it.)

Valid entries	Usage
0 to **9** up to 8 digits**,** or blank	Enter a string from 1 to 8 digits long (not including * or #).

AAR/ARS Internal Call Total Length

The total length of the internal call digit string. Appears only if, on the **System-Parameters Customer-Options** screen, the **ARS/AAR Dialing Without FAC** field is **y**. (Contact your Avaya technical support representative to discuss the ARS/AAR Dialing Without FAC feature before enabling it.)

Valid entries	Usage
6 to **10** or blank	Enter the total length of the internal call digit string, which includes the Internal Call Prefix and the calling or called extension.

Note:

If either the **AAR/ARS Internal Call Prefix** or the **AAR/ARS Internal Call Total Length** field is non-blank and valid, the other must also be non-blank and valid. In addition, the longest extension length on the **Dial Plan Analysis** screen, plus the length of the **ARS/AAR Internal Call Prefix**, must equal or be greater than, the **ARS/AAR Internal Call Total Length** value.

ETA Node Number

Enter the number of the destination server for Extended Trunk Access (ETA) calls. ETA calls are unrecognized numbers you can send to another switch for analysis and routing. Such numbers can be Facility Access Codes, Trunk Access Codes, or extensions that are not in the UDP table.

Valid entries	Usage
1 to **999** or blank	Enter the number of a destination server.

ETA Routing Pattern

Enter the number of the routing pattern to reach the destination server.

Valid entries	Usage
1 to **640** or blank	Avaya DEFINITY Server R, CSI, or SI.
1 to **999** or blank	Avaya S8300 Media Server, S8700 or S8710 IP-Connect, S8700 or S8710 Multi-Connect.

Local Node Number

Enter a number to identify a specific node in a server network. This entry must match the DCS switch node number and the CDR node number if they are specified.

Valid entries	Usage
1 to **63**	Enter the number of a specific node in a network.
blank	The field may be left blank if automatic restoration, DCS, and CDR are not used.

UDP Extension Search Order

Appears if, on the **System-Parameters Customer-Options** screen, the **Uniform Dialing Plan** field is **y**. Specifies the first table to search to match a dialed extension. If the dialed extension is not found in the specified place, it then is searched for in the "other" place.

Valid entries	Usage
local-extensions-first	Searches the local server first to match a dialed extension; if not found, then uses the UDP tables to route the call.
udp-table-first	Searches the UDP tables for an off-switch conversion; if not found, then searches the local server for the dialed extension.

Digit Absorption

This screen implements up to 5 digit absorption lists. The screen may be required for each CO and FX trunk group connected to a step-by-step CO. Each outgoing digit string from the server running Communication Manager to the step-by-step CO is treated according to entries in the "Absorption Treatment Assignment" section of the screen.

Note:

If the **Digits** field on the **Trunk Group** screen is blank, you cannot administer Digit Absorption.

Field descriptions for page 1

Figure 318: Digit Absorption screen

```
Change digit absorption                                        Page 1 of 1
                              DIGIT ABSORPTION

                            List Number: __
  ABSORPTION TREATMENT INFORMATION (All selections must be from same group)
                 Choice      Meaning
    Group I.       A       Digit not absorbed.
                   B       Digit absorbed repeatedly.
                   C       Digit absorbed once with no further absorption.
    Group II.      A       Digit not absorbed.
                   D       Digit absorbed only if it is the first digit.
                   E       Digit absorbed only if it is the second digit and
                           the first digit was already absorbed.
                   F       Digit asorbed only if it is the first or second digit.

  ABSORPTION TREATMENT ASSIGNMENT (select treatment (A-F) for each digit below)
          0: A          2: A          4: A          6: A          8: A
          1: A          3: A          5: A          7: A          9: A
```

Absorption Treatment Assignment

Valid entries	Usage
A to **F**	Enter a desired treatment letter. All choices for the digits 0 through 9 must be taken from the same group (Group I or Group II).

Absorption Treatment Information

This is a display-only section. It shows how Digit Absorption treats each digit, 0 through 9, depending on the assignment of A through C for Group I, and A, D, E, and F for Group II. Enter the assignment on the next section on the screen.

List Number

A display-only field indicating the Digit Absorption List number (**0** to **4**). The list number is referenced from a field entry on the associated trunk group.

DLG Administration

This screen appears if, on the **Station** screen, the **Service Type** field is **DLG**. This screen includes all clients that are allowed to connect to the virtual DEFINITY LAN Gateway (DLG).

Field descriptions for page 1

The following screen appears if the **Service Type** field is **DLG** and is administered. This screen includes all clients that are allowed to connect to the co-resident DEFINITY LAN Gateway (DLG). This page is always the last page on the **IP Services** screen.

Figure 319: DLG Administration screen

```
                          DLG Administration                    Page X of Y

 CTI_-Link    Enabled    Client Name         Client Link    Client Status

     1          y        client1                 1            in use
     3          y        client2                 1            idle
     4          y        client1                 2            idle
```

Client Link

The client name on the CTI server side identifies the logical connection on the client side (IP address and host name) for multiple links and for the corresponding client (CTI link number/ client name) that are connected to you.

Valid entries	Usage
1 to **X** or blank	Enter the number between 1 and the number of CTI links allowed on the system (for instance, 8 or 16).

Client Name

Enter the node name of the client adjunct requesting service for the specified CTI link.

Client Status

A display-only field indicating **idle** (currently there is no active connection) or **in use** (currently there is an active connection for this client link).

CTI Link

A display-only field indicating the link number.

Enabled

Controls whether the co-resident DLG will allow connections for the corresponding client.

Valid entries	Usage
y	Enter **y** to access DLG.
n	Will refuse connections for that Client Name/ Client Link pair.

DS1 Circuit Pack

Use this screen to administer all DS1 circuit packs.

Field descriptions for page 1

Figure 320: DS1 Circuit Pack screen

```
 add ds1 xxxxxx                                          Page 1 of 2
                              DS1 CIRCUIT PACK

              Location: _____                          Name: ________________
              Bit Rate: _____                    Line Coding: ____
    Line Compensation: _                         Framing Mode: ___
       Signaling Mode: ___________                  D-Channel: __
              Connect: _________                    Interface: ___________
        Interconnect: ___                       Peer Protocol: ________
                                              Country Protocol: ____
                                              Protocol Version: _
Interface Companding: _____                              CRC? _
            Idle Code: _________
                                   DCP/Analog Bearer Capability: ________
                                              T303  Timer(sec):

    MMI Cabling Board: _____  MMI Interface: ESM

 MAINTENANCE PARAMETERS

       Slip Detection? _                  Near-end CSU Type: ________
                                            Block Progress Indicator? n
```

Figure 321: DS1 Circuit Pack screen for Croatia and South Africa

```
add ds1 xxxxxx                                          Page 1 of 2
                              DS1 CIRCUIT PACK

            Location: ______                          Name: _____________
            Bit Rate: ______                   Line Coding: ____

      Signaling Mode: __________

        Interconnect: ____                Country Protocol: ______

 Interface Companding: ____
           Idle Code: ____

              Received Digital Metering Pulse Minimum (ms):
              Received Digital Metering Pulse Maximum (ms):
                    Received Digital Metering Pulse Value:
      Slip Detection: __                  Near-end CSU Type: ______
                                                Block Progress Indicator? n
```

The following screen is valid *only* for the TN2242.

Figure 322: DS1 Circuit Pack screen for Channel Associated Signaling

```
add ds1 xxxxxx                                          Page 1 of 2
                              DS1 CIRCUIT PACK

            Location: 01A13                           Name: ________________
            Bit Rate: 2.048                    Line Coding: cmi

       Signaling Mode: CAS
         Interconnect: pbx
                                           Country Protocol: 3

 Interface Companding: mulaw
           Idle Code: 11111111



MAINTENANCE PARAMETERS

       Slip Detection? n
```

Bit Rate

Use this field to select the maximum transmission rate for DS1 circuit packs that support either T-1 or E-1 service. For circuit packs that only support one of these services, the field is a display-only field.

Note:

Once an **add ds1** operation is complete (that is, the **DS1** screen has been submitted) you can't change the **Bit Rate** field with a **change ds1** command. Instead, execute a **remove ds1** command. Then use the **add ds1** command to administer the circuit pack again. You'll have to re-enter all the information for the circuit pack.

TN464C (and later release) circuit packs have an option switch that must be set to match the entry in the **Bit Rate** field.

Valid entries	Usage
1.544	Use for T-1 service.
2.048	Use for E-1 service.

Channel Numbering

The ETSI and ISO QSIG specifications require that B-channels on an E1 be encoded as 1 to 30 in the Channel ID IE. Prior to the existence of this field, Avaya Communication Manager only used this scheme for Country Protocols 2a (Australia) and 13a (Germany 1TR6). This field appears when the **Signaling Mode** field is **isdn-pri**, the **Bit Rate** field is **2.048**, the **Connect** field is **pbx**, and the **Interface** field is **peer-master** or **peer-slave**.

Valid entries	Usage
timeslot	
sequential	If Avaya Communication Manager is connected via QSIG trunks to a switch/server supporting the ETSI QSIG or ISO QSIG specifications, this field must be **sequential**. When the **Signaling Mode** field is **isdn-pri** and the **Bit Rate** field is **2.048**, but the **Channel Numbering** field does not display because of the setting of other fields, it is set internally to **sequential** for 2a (Australia) and 13a (Germany).

Connect

In order to control communications at layers 2 and 3 of the ISDN-PRI protocol, use this field to specify what is on the far end of this DS1 link. This field only appears when the **Signaling Mode** field is **isdn-pri**.

Valid entries	Usage
pbx	Enter **pbx** if this DS1 link is connected to another switch in a private network. If **pbx** is entered, the **Interface** field appears.
line-side	Enter **line-side** when Communication Manager is acting as the network side of an ISDN-PRI interface. Use **line-side** to connect to Roll About Video equipment.
network	Enter **network** when the DS1 link connects Communication Manager to a central office or any other public network switch.
host	Enter **host** when the DS1 link connects Communication Manager to a computer.

Country Protocol

The entry in this field must match the country protocol used by the far-end server. For connections to a public network, your network service provider can tell you which country protocol they are using.

This field appears if the **Signaling Mode** field is **CAS** or **isdn-pri**. For the Japanese 2Mbit trunk circuit pack, this is a display-only field if the **Signaling Mode** field is **CAS**.

Valid entries	Usage
1 to **25**	Enter the country protocol used by the central office at which this link terminates.
etsi	Enter **etsi** if your network service provider uses the protocol of the European Telecommunications Standards Institute (ETSI). Enter **etsi** only if the **Signaling Mode** field is **isdn-pri**.

CRC

This field indicates whether a cyclic redundancy check (CRC) will be performed on transmissions that the DS1 circuit pack receives. This field does not display for all circuit packs.

Valid entries	Usage
y	Enter **y** when the **Signaling Mode** field is **CAS** and the DS1 link is providing E-1 service.
n	Enter **n** for all other applications.

D-Channel

The Japanese 2Mbit trunk circuit pack, when administered to support ISDN-PRI signaling, allows you to assign the D-channel to any channel from 1 to 31 in an E-1 facility. You cannot submit the screen if this field is blank. Using the **change ds1** command, you can change this field if the D-channel is not used in a signaling group. This field appears only when the **Location** field indicates the circuit pack is a Japanese 2Mbit trunk circuit pack and the **Signaling Mode** field is **isdn-pri**.

Valid entries	Usage
1 to **31**	Enter the number of the channel that will be used as the D-channel.

DCP/ANALOG Bearer Capability

This field appears when the **Signaling Mode** field is **isdn-pri**. This field sets the information transfer capability in a bearer capability IE of a setup message to **speech** or **3.1kHz**.

Valid entries	Usage
3.1kHz	Provides 3.1kHz audio encoding in the information transfer capability.
speech	Provides speech encoding in the information transfer capability.

DMI-BOS

The DMI/BOS protocol is used for high-speed digital communications between a host computer and Avaya Communication Manager. With this 24-channel protocol, channels 1 to 23 of the DS1 link carry data and channel 24 carries control signaling. DMI/BOS has greater capacity than a robbed-bit 24-channel facility. This field appears only when the **Signaling Mode** field is **common-chan**.

Valid entries	Usage
y	Enter **y** to activate the Digital Multiplexed Interface-Bit Oriented Signaling (DMI-BOS) format.
n	Enter **n** to use an Avaya proprietary format.

Framing Mode

Use this field to select either superframe (sf or d4) or extended superframe (esf) for T1 service on the DS1 link. The framing mode you use must match the mode used on the other end of the link, so work with your network services provider to determine the appropriate entry for this field.

This field only appears if the **Bit Rate** field is **1.544** (that is, if you're using T-1 service). If you're using E-1 service, Avaya Communication Manager automatically selects CEPT1 framing.

Tip:

Avaya recommends using ESF when your service provider supports it, especially if you may someday upgrade the facility to ISDN. The ESF format provides enhanced performance measurements and uses a sophisticated error-checking method to ensure data integrity.

Valid entries	Usage
d4	Enter **d4** to use the basic DS1 superframe (sf). Avaya recommends this mode only for voice traffic.
esf	Enter **esf** to use the Extended Superframe format. Avaya recommends this mode for digital data traffic. If you enter **esf** for a TN464F, TN767E, or a later suffix DS1 circuit pack, a second page of the **DS1 Circuit Pack** screen becomes available to administer ESF Data Link options.

Idle Code

CAUTION:

Customers: The entry in the **Country Protocol** field sets the default idle code. Do not change the default without assistance from Avaya or your network services provider.

For some circuit packs, this is a display-only field.

Valid entries	Usage
any 8-digit string of 0's and 1's	This entry sets the signal sent out over idle DS0 channels. The string must be compatible with the protocol used by the far-end switch/server.

Interconnect

For E-1 service using channel-associated signaling, the entry in this field tells Avaya Communication Manager whether the DS1 circuit pack is using a public or private network protocol. The entry in this field must agree with the entry in the **Group Type** field on the **Trunk Group** screen. This field appears only when the **Signaling Mode** field is **CAS**.

Valid entries	Usage
pbx	If **pbx** is selected, the board operates as a tie trunk circuit pack.
CO	If **CO** is selected, the board operates as a CO or DID circuit pack.

Interface

This field only appears when the **Connect** field is **pbx**; that is, when this DS1 link is providing an ISDN-PRI connection in a private network. The **Interface** field controls how your server negotiates glare with the far-end switch. The servers at either end of the DS1 link must have complementary settings in this field: if not, the D-channel won't even come up. For example, if the media server at one end of the link is administered as **network**, the other end must be administered as **user**.

Valid entries	Usage
Use the following 2 values for private network applications in the U.S.	
network	Enter **network** if your server overrides the other end when glare occurs. If you are connecting your server to a host computer, set this field to **network**.
user	Enter **user** if your server releases the contested circuit and looks for another when glare occurs. If you are connecting your server to a public network, set this field to **user**.
Use the following values for private networks (including QSIG networks) outside the U.S. Entering either of these values causes the **Peer Protocol and Side** fields to appear.	
peer-master	Enter **peer-master** if your switch overrides the other end when glare occurs.
peer-slave	Enter **peer-slave** if your switch releases the contested circuit and looks for another when glare occurs.

Interface Companding

The entry in this field must match the companding method used by the far-end switch. This field does not appear for all DS1 circuit packs.

Valid entries	Usage
alaw	Enter **alaw** for E-1 service.
mulaw	Enter **mulaw** for T-1 service.

Interworking Message

This field determines what message the server sends when an incoming ISDN trunk call interworks (is routed over a non-ISDN trunk group).

Valid entries	Usage
PROGress	Normally select this value. PROGress asks the public network to cut through the B-channel and let the caller hear tones such as ringback or busy tone provided over the non-ISDN trunk.
ALERTing	ALERTing causes the public network in many countries to play ringback tone to the caller. Select this value only if the DS1 is connected to the public network, and it is determined that callers hear silence (rather than ringback or busy tone) when a call incoming over the DS1 interworks to a non-ISDN trunk.

ITN-C7 Long Timers

This field controls the T302 and T303 timers. It only appears if the **Signaling Mode** field is **isdn-pri**.

Valid entries	Usage
y	Use if you want to increase the length of the long timers.
n	Leave n if you want to use the default long timers.

Line Coding

This field selects the type of line coding used on this facility. The setting in this field must match the setting on the far-end of the link, or you must have an intervening CSU to convert the line coding protocols. Voice calls will work even if line coding does not match, but a single data call will bring down the DS1 facility. For the TTC 2Mb CMI Trunk circuit pack, this is a display-only field showing **cmi** (coded mark inversion).

The following information is for reference. Talk with your network service provider or your Avaya technical support representative to find the appropriate protocol for your application.

CAUTION:

If you change this field, you must busy out the DS1 circuit pack. You must also change the following screens: **Route-Pattern**, **Access Endpoint**, **PRI Endpoint**, **Signaling-Group**, *and* **Trunk-Group**.

Note:

When the DS1 circuit pack is used for ISDN service, the ISDN D-channel data is inverted when **ami-basic** or **ami-zcs** is entered and not inverted when **b8zs** or **hdb3** is entered.

Valid entries	Usage
b8zs (bipolar eight zero substitution)	Enter **b8zs** for T-1 facilities that support voice and/or data traffic. Enter **b8zs** if you need a 64K clear channel.
ami-zcs (alternate mark inversion - zero code suppression)	Enter **ami-zcs** only for T-1 facilities that carry voice traffic: Avaya does not recommend this for digital-data applications. If you anticipate upgrading this facility to ISDN, use b8zs line coding if possible.
ami-basic (alternate mark inversion-basic)	Enter **ami-basic** for unrestricted E-1 facilities.
hdb3 (high density bipolar 3)	Enter **hdb3** for restricted E-1 facilities.
cmi (coded mark inversion)	Used in Japan, **cmi** is the only type of line coding you can use with the Japanese 2Mbit trunk circuit pack. This field becomes a display-only field when you are administering the Japanese 2Mbit trunk circuit pack.

Line Compensation

The appropriate entry in this field varies with the type of cable used, so work with your network service provider to determine the correct setting in your situation. The following table shows the appropriate entries for different lengths of 22-gauge ABAM cable terminated on a DSX-1 cross-connect.

Valid entries	Usage
1	Length: 000 to 133 (ft), 000 to 40.5 (m)
2	Length: 133 to 266 (ft), 40.5 to 81.0 (m)
3	Length: 266 to 399 (ft), 81.0 to 122 (m)
4	Length: 399 to 533 (ft), 122 to 163 (m)
5	Length: 533 to 655 (ft), 163 to 200 (m)

The following table shows the appropriate entries for different lengths of 22-gauge ABAM cable directly connecting to DS1 interfaces.

Valid entries	Usage
1	Length: 0000 to 0266(ft), 000 to 081(m)
2	Length: 0266 to 0532(ft), 081 to 162(m)
3	Length: 0532 to 0798(ft), 162 to 243(m)
4	Length: 0798 to 1066(ft), 243 to 325(m)
5	Length: 1066 to 1310(ft), 325 to 400(m)

Location

This display-only field shows the port address specified in the **add** command when the circuit pack was first administered.

MMI Cabling Board

This field appears only if the **MMCH** field is **y** on the **System-Parameters Customer-Options** screen.

Valid entries	Usage
slot address (cabinet, carrier, slot)	Enter the slot location (cabinet, carrier, slot) of the multimedia interface circuit pack that is connected to the Expansion Services Module (ESM).

MMI Interface

This display-only field appears if the **MMCH** field is **y** on the **System-Parameters Customer-Options** screen and there is a value in the **MMI Cabling Board** field.

Name

Use this field to assign a significant, descriptive name to the DS1 link. Avaya recommends putting the vendor's circuit ID for the link in this field, because that information helps you troubleshoot problems with the link, but you could also use this field to indicate the function or the destination of this DS1 facility. In that case, put the DS1 link circuit ID in the **Name** field of the trunk group associated with this link.

Valid entries	Usage
1 to 15 characters	Enter a name for the DS1 link.

Peer Protocol

This allows you to administer the peer level protocol that will operate in a private network. This field appears if the **Interface** field is **peer-master** or **peer-slave**. To enter **Q-SIG**, the **Basic Call Setup** field on the **System-Parameters Customer-Options** screen must be **y**

Valid entries	Usage
Q-SIG	This implements QSIG Network Basic Call.
TTC	For private networking. Requires a Digital Trunk (Japan 2 MB TTC) (TN2242) circuit pack.

Protocol Version

In countries whose public networks allow multiple layer-3 signaling protocols for ISDN-PRI service, this field selects the protocol that matches your network service provider's protocol. See Public network signaling administration for ISDN-PRI Layer 3 on page 940 to see which countries support which protocols.

This field appears only when:

- The **Signaling Mode** field is **isdn-pri** and the **Connect** field is **network.**
- The **Signaling Mode** field is **isdn-pri,** the **Connect** field is **pbx**, and the **Interface** field is **user** or **network**.

Valid entries	Usage
a, b, c, d	The entry in this field must match the protocol used by your network service provider, so work with your vendor to determine the appropriate entry.

WARNING:
The AT&T Switched Network Protocol does not support restricted displays of connected numbers. Therefore, if you administer the 1a country-protocol/ protocol-version combination on the **DS1** screen, you cannot set the **Send Connected Number** field to r (restricted) on the **ISDN-PRI Trunk Group** screen, as this causes display problems.

Public network signaling administration for ISDN-PRI Layer 3

The table below shows Avaya Communication Manager public-network access connections for ISDN-PRI Layer 3.

Admin value	Country	Protocol supported	B-channel mtce msg
1-a	United States, Canada	AT&T TR 41449/ 41459 (tested with AT&T network, Canadian network, and MCI network)	Service
1-b	United States	Bellcore TR 1268; NIUF.302; ANSI T1.607	Restart
1-c	United States	NORTEL DMS-250 BCS36/IEC01	Service
1-d	United States	Telecordia SR-4287	Service
2-a	Australia	AUSTEL TS014.1; Telecom Australia TPH 1856 National ISDN protocol	Restart
2-b	Australia	ETSI ISDN protocol	Restart
3	Japan	NTT INS-NET	Restart
4	Italy	ETS 300 102	Restart
5	Netherlands	ETS 300 102	Restart
6	Singapore	ETS 300 102	Restart
7	Mexico	ETS 300 102	Restart
8	Belgium	ETS 300 102	Restart
9	Saudi Arabia	ETS 300 102	Restart
10-a	United Kingdom	ETS 300 102 (for connection to DASS II/DPNSS through external converter)	Restart

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Admin value	Country	Protocol supported	B-channel mtce msg
10-b	United Kingdom, Ireland	ETS 300 102 (Mercury); British Telecom ISDN 30; Telecom Eireann SWD 109	None
11	Spain	Telefonica ISDN Specification	Restart
12-a	France	VN4 (French National PRI)	None
12-b	France	ETS 300 102 modified according to P10-20, called Euronumeris	None
13-a	Germany	FTZ 1 TR 6 (German National PRI)	None
13-b	Germany	ETS 300 102	Restart
14	Czech Republic, Slovakia	ETS 300 102	Restart
15	Russia (CIS)	ETS 300 102	Restart
16	Argentina	ETS 300 102	Restart
17	Greece	ETS 300 102	Restart
18	China	ETS 300 102	Restart
19	Hong Kong	ETS 300 102	Restart
20	Thailand	ETS 300 102	Restart
21	Macedonia	ETS 300 102	Restart
22	Poland	ETS 300 102	Restart
23	Brazil	ETS 300 102	Restart
24	Nordic	ETS 300 102	Restart
25	South Africa	ETS 300 102	Restart
ETSI-a	Europe, New Zealand, etc.	ETS 300 102	Restart
ETSI-b		ETS 300 102	None

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Received Digital Metering Pulse Maximum (ms)

This field appears only when the **Signal Mode** field is **cas** (Channel Associated Signaling), the **Interconnect** field is **co** or **pbx**, and the **Country Protocol** field is administered for a protocol that uses periodic pulse metering (PPM) as defined in Incoming digital PPM signaling default per Country Protocol code on page 943. The default value depends on the **Country Protocol f**ield's entry.

Valid entries	Usage
20 to **1000** ms in increments of 10ms.	Work with your network services provider to determine the appropriate entry. The entry must be greater than the **Received Digital Metering Pulse Minimum** field.

Received Digital Metering Pulse Minimum (ms)

This field appears only when the **Signal Mode** field is **cas** (Channel Associated Signaling), the **Interconnect** field is **co** or **pbx**, and the **Country Protocol** field is administered for a protocol that uses periodic pulse metering (PPM) as defined in Incoming digital PPM signaling default per Country Protocol code on page 943. The default value depends on the **Country Protocol** field's entry.

Valid entries	Usage
20 to **1000** ms in increments of 10ms.	Work with your network services provider to determine the appropriate entry. The entry must be less than the **Received Digital Metering Pulse Maximum** field.

Received Digital Metering Pulse Value

This field appears when the **Signal Mode** field is **cas** (Channel Associated Signaling), the **Country Protocol** field is **21**, and the **Interconnect** field is **co** or **pbx**.

Valid entries	Usage
0, **1**	Work with your network services provider to determine the appropriate entry.

Table 13: Incoming digital PPM signaling default per Country Protocol code *1 of 2*

Code	Country	PPM Min (ms)	PPM Max (ms)	PPM Value
0	null	NA	NA	NA
1	U.S.	NA	NA	NA
2	Australia	80	180	0
3	Japan	NA	NA	NA
4	Italy	120	150	1
5	Netherlands	90	160	0
6	Singapore	NA	NA	NA
7	Mexico	20	180	1
8	Belgium	20	180	1
9	Saudi Arabia	NA	NA	NA
10	UK	NA	NA	NA
11	Spain	20	220	0
12	France	NA	NA	NA
13	Germany	NA	NA	NA
14	Czech Republic	20	420	1
15	Russia CIS	NA	NA	NA
16	Argentina	10	180	1
17	Greece	100	180	1
18	China	NA	NA	NA

1 of 2

Table 13: Incoming digital PPM signaling default per Country Protocol code *2 of 2*

Code	Country	PPM Min (ms)	PPM Max (ms)	PPM Value
19	Hong Kong	NA	NA	NA
20	Thailand	20	180	1
21	Macedonia Croatia	120 20	180 80	1 1
22	Poland	100	150	0
23	Brazil	NA	NA	NA
24	Nordic	NA	NA	NA
25	South Africa	160	240	0, 1

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Side

This field controls how your server running Communication Manager resolves glare at layer 3 over an ISDN-PRI link in QSIG private networks. It appears if the **Interface** field is **peer-master** or **peer-slave**.

The default value of the field changes depending upon which value the **Interface** field contains.

CAUTION:

It is critical that administration on this server correctly pairs with the administration of the far-end switch/server. If the far-end is administered as the b side, this field should be set to a regardless of whether the layer 2 designation is peer-master or peer-slave, and vice versa.

Valid entries	Usage
a	Enter **a** if the **Interface** field is **peer-master** (this server overrides the far-end when glare occurs).
b	Enter **b** if the **Interface** field is **peer-slave** (this server releases the contested circuit and looks for another when glare occurs).

Signaling Mode

This field selects the signaling method used for the DS1 link. This mode must match the method used on the other end of the link, so work with your network services provider to determine the appropriate entry for this field.

Valid entries	Usage
CAS (Channel Associated Signaling)	Enter **CAS** for out-of band signaling with E-1 service. This setting yields 30 64-kbps B-channels for voice or data transmission. Channel 0 is used for framing while channel 16 carries signaling.
robbed-bit	Enter **robbed-bit** for in-band signaling with T-1 service. This setting yields 24 56-kbps B-channels for voice transmission.
isdn-pri	Enter **isdn-pri** for either T-1 or E-1 ISDN service. This setting supports both Facility Associated Signaling and Non-Facility Associated Signaling.
isdn-ext	Enter **isdn-ext** for either T-1 or E-1 ISDN service. This setting supports only Non-Facility Associated Signaling.
common-chan	Enter **common-chan**, for out-of-band signaling with T-1 service. This setting yields 23 64-kbps B-channels for voice or data transmission. Channel 24 is used for signaling.

T303 Timer (sec)

Use this field to enter the number of seconds the system waits for a response from the far end before invoking Look Ahead Routing. Appears when the **Group Type** field is **isdn-pri**.

Valid entries	Usage
2 to **10**	Enter a number of seconds between 2 and 10. Default is 4. If Long Timers is enabled, defaults to a read-only value of 13.

MAINTENANCE PARAMETERS

Alarm When PRI Endpoint Detached

Use this field for DS1 circuit packs connected to Roll-About Video equipment. This field appears only when the **Connect** field is **line-side**.

Valid entries	Usage
y/n	Enter **y** if you want the server running Avaya Communication Manager to generate an alarm when the DS1 board detects a loss of signal (for example, if the video equipment is disconnected).

EC Configuration

Appears when Echo Cancellation is **y** on the **DS1 Circuit Pack** screen.

Valid entries	Usage
1 to **15**	Four default echo cancellation configurations are available: Enter 1, 5 to 15: provides the most rapid adaptation in detecting and correcting echo at the beginning of a call. **Note:** Plan 1 introduces 6 db of loss for additional cancellation. Enter 2: slightly slower adaptation to echo, use if speech is clipped when both parties talk at the same time. Enter 3: slightly slower adaptation to echo, may result in a 2 or 3 second fade on strong echo Enter 4: in cases of extreme echo, excessive clipping or breakup of speech. May result in slight echo or background noise.

EC Direction

Direction of echo cancellation. Appears when Echo Cancellation is **y** on the **DS1 Circuit Pack** screen.

Valid entries	Usage
inward/outward	Indicate the direction of the echo that is being cancelled.

Echo Cancellation

Appears when **DS1 Echo Cancellation** is **y** on the **System-Parameters Customer-Options** screen and circuit packs support echo cancellation.

Valid entries	Usage
y/n	Enter **y** to allow echo cancellation.

Near-end CSU Type

This field appears only when the DS1 circuit pack is a TN767D or TN464E or later suffix model, the **Bit Rate** field is **1.544** and the **Country Protocol** field is **1** (U.S.). This field does not display for all circuit packs.

Valid entries	Usage
other	Enter **other** if no channel service unit is attached to the DS1 facility or if the CSU is an external unit. No options are available on page 2 for administering an external CSU.
integrated	Enter **integrated** if a 120A CSU module is attached to the DS1 board. This integrated channel service unit (ICSU) can accept software-administrable option downlinks (that is, it can respond to test codes from technician's equipment and report its status). When you enter **integrated**, fields for administering options on the ICSU appear on page 2 of the **DS1 Circuit Pack** screen.

Slip Detection

Slips — synchronization errors — slow digital transmissions and can result in data loss. The server maintains a slip-count record for each DS1 interface to detect errors and evaluate their severity (the type of alarm). If as many as 50 percent of those spans administered for slip detection are experiencing slips (with respect to the primary), then a decision is made to switch to the secondary.

CAUTION:

Always enter **y** for DS1 circuit packs that serve as primary or secondary synchronization references.

Valid entries	Usage
y	Enter **y** to allow maintenance software to measure the slip-rate of this circuit pack and determine whether it's excessive. Typically, enter **y** for DS1 spans used for data applications and for spans used as synchronization references. This excludes all T1-spans connecting channel banks, unless the channel bank is externally timed. This entry enables switching between the primary and secondary synchronization references and an internal high-accuracy clock.
n	Enter **n** for DMI-BOS links or when testing is not required. Normally, enter **n** for DS1 spans that are used exclusively for voice and that do not serve as the primary or secondary synchronization source.

Block Progress Indicator

This field allows you to block sending of the progress indicator in the SETUP message. It appears when the **Country Protocol** field is set to 1 and the **Protocol Version** field is set to b.

Valid entries	Usage
y	Enter **y**(es) to prevent the progress indicator from being sent in the SETUP message.
n	Enter **n**(o) to allow the progress indicator to be sent.

Field descriptions for page 2

Figure 323: DS1 Circuit Pack screen

```
add ds1 xxxxxx
                                DS1 CIRCUIT PACK

ESF DATA LINK OPTIONS

                        Network Management Protocol:
  Send ANSI-T1.403 One-Second Performance Reports?
                                Far-end CSU Address:


INTEGRATED CSU OPTIONS
                                        Transmit LBO:
                                        Receive ALBO:
                                        Upon DTE LOS:

CPE LOOPBACK JACK OPTIONS
                     Supply CPE Loopback Jack Power?
```

CAUTION:

Customers: Do not change fields on this page without assistance from Avaya or your network service provider.

Page 2 does not appear for all DS1 circuit packs. For those circuit packs that support it, this page appears only when the **Framing Mode** field is **esf** or the **Near-end CSU Type** field is **integrated**.

ESF DATA LINK OPTIONS

Far-end CSU Address

This field, which, appears only if the **Framing Mode** field is **esf**.

Valid entries	Usage
a **b**	Enter **b**. This field administers the transmit direction address used for the **ESF data link** command with both integrated and external channel service units (CSU).

Network Management Protocol

This field appears only if the **Framing Mode** field is **esf**.

Valid entries	Usage
tabs	The entry in this field, used only with circuit packs that have an integrated channel service unit (CSU), allows the data link to be remotely monitored.

Send ANSI-T1.403 One-Second Performance Reports

This field selects whether your DS1 circuit pack will send error reports to the far-end server/switch. These reports are useful for network management, and are sent at 1-second intervals when enabled. This field appears only if the **Framing Mode** field is **esf**. It is used only with circuit packs that have an integrated channel service unit (CSU).

Valid entries	Usage
y/n	Enter **n**. Consult your Avaya technical support representative if you think you may want to use these reports.

INTEGRATED CSU OPTIONS

Receive ALBO (Receive Automatic Line Build-Out)

This field increases the strength of incoming signals by a fixed amount to compensate for line losses.

Valid entries	Usage
26db	To set this field correctly, you should measure the signal loss on this specific facility. However, you may enter **26db** for most applications. **36db** is occasionally appropriate, mainly on campus networks that don't conform to public telephone network standards.
36db	

Transmit LBO (Transmit Line Build-Out)

This field reduces the outgoing signal strength by a fixed amount. The appropriate level of loss depends on the distance between your Communication Manager server (measured by cable length from the smart jack) and the nearest repeater. Where another server/switch is at the end of the circuit, as in campus environments, use the cable length between the 2 switches to select the appropriate setting from the table below. This field appears if the **Near-end CSU Type** field is **integrated**.

Valid entries	Usage
0db	For distances of 2,001 to 3,000 feet
-7.5db	For distances of 1,001 to 2,000
-15db	For distances of 0 to 1,000 feet
-22.5db	For mid-span repeaters

Upon DTE LOS

DTE stands for "Data Terminal Equipment." This field tells Communication Manager what to do if the outgoing signal from the DS1 circuit pack (the data terminal equipment) to the network is lost.

Valid entries	Usage
loopback	Enter **loopback** to return the network signal to the network. This prevents any alarms from being generated at the far-end
ais	Enter **ais** (Alarm Indicator Signal) to send an unframed all-ones signal (the AIS or Blue Alarm) to the far-end server/switch. This option alerts your network service provider to the problem immediately and aids in troubleshooting.

CPE LOOPBACK JACK OPTIONS

Supply CPE Loopback Jack Power

If a CPE (Customer Premise Equipment) Loopback Jack is installed, the DS1 board should supply power to the equipment during loopback testing.

Valid entries	Usage
y/n	Enter **y** if a CPE Loopback Jack is installed. If not, you must enter **n**.

Related topics

See Setting up Digital Trunks on page 438 for instructions.

See "DS1 Trunk Service" in *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205, for more information.

Extended Pickup Group

This screen allows grouping of pickup groups into extended pickup groups. This allows users to answer calls outside their immediate group. The maximum number of groups that can be added to an extended pickup group is 25.

Field descriptions

Figure 324: Extended Pickup Group screen

```
change extended-pickup-group 1                              Page   1 of   2
                              EXTENDED PICKUP GROUP

                            Extended Group Number: 56

Pickup   Pickup Group                          Pickup   Pickup Group
Number     Number                              Number     Number

 0:      _____                                 13:      _____
 1:      _____                                 14:      _____
 2:      _____                                 15:      _____
 3:      _____                                 16:      _____
 4:      _____                                 17:      _____
 5:      _____                                 18:      _____
 6:      _____                                 19:      _____
 7:      _____                                 20:      _____
 8:      _____                                 21:      _____
 9:      _____                                 22:      _____
10:      _____                                 23:      _____
11:      _____                                 24:      _____
12:      _____                                 25:      _____
```

Extended Group Number

This display-only field shows the number associated with a collection of pickup groups. The extended group is a collection of pickup groups that can answer calls from other pickup groups in the same extended group.

Pickup Group Number

This field determines which call pickup groups can answer calls in the extended pickup group.

Valid entries	Usage
1 to **800** or blank (DEFINITY CSI, SI) **1** to **5000** or blank (DEFINITY R, S8300 Media Server, S8700 or S8710 IP-Connect, S8700 or S8710 Multi-Connect)	Enter the pickup group numbers for each of the pickup groups that you want to belong to this extended group.

Pickup Number

This display-only field shows the pickup number assigned to the pickup group number. This is the number users dial after the feature access code (FAC) to pick up calls in their extended pickup group.

Extensions Administered to have an MCT-Control Button

This screen lists the extensions that can take control of a Malicious Call Trace (MCT) request. In order to give a user the ability to take control of such requests, you need to add their extension to this list and assign them a mct-control feature button.

Field descriptions for page 1

Figure 325: Malicious Call Trace control extensions screen

```
 change mct-group-extensions
                                                              Page 1 of 1
Extensions Administered to have an MCT-Control Button:
  1: _________ 19: _________ 37: _________ 55: _________ 73: _________  91: _________
  2: _________ 20: _________ 38: _________ 56: _________ 74: _________  92: _________
  3: _________ 21: _________ 39: _________ 57: _________ 75: _________  93: _________
  4: _________ 22: _________ 40: _________ 58: _________ 76: _________  94: _________
  5: _________ 23: _________ 41: _________ 59: _________ 77: _________  95: _________
  6: _________ 24: _________ 42: _________ 60: _________ 78: _________  96: _________
  7: _________ 25: _________ 43: _________ 61: _________ 79: _________  97: _________
  8: _________ 26: _________ 44: _________ 62: _________ 80: _________  98: _________
  9: _________ 27: _________ 45: _________ 63: _________ 81: _________  99: _________
 10: _________ 28: _________ 46: _________ 64: _________ 82: _________ 100: _________
 11: _________ 29: _________ 47: _________ 65: _________ 83: _________
 12: _________ 30: _________ 48: _________ 66: _________ 84: _________
 13: _________ 31: _________ 49: _________ 67: _________ 85: _________
 14: _________ 32: _________ 50: _________ 68: _________ 86: _________
 15: _________ 33: _________ 51: _________ 69: _________ 87: _________
 16: _________ 34: _________ 52: _________ 70: _________ 88: _________
 17: _________ 35: _________ 53: _________ 71: _________ 89: _________
 18: _________ 36: _________ 54: _________ 72: _________ 90: _________
```

1 to 100

Enter the extension for a telephone or attendant console that you want to have an **MCT-Control** button. Note that you must also assign the **mct-control** button on the extension's **Station** or **Attendant Console** screen.

Feature Access Code

This screen assigns feature access codes (FACs) that, when dialed, activate or cancel the system features. Each field on this screen has the same valid values, which must conform to feature access codes or dial access codes as defined by your dial plan.

Valid entries	Usage
1 to 4 digit number, * #	***** and # may be used as first digit. However, analog rotary dial phones cannot use the "*" and "#" symbols.

Field descriptions for page 1

Figure 326: Feature Access Code (FAC) screen

```
change feature-access-codes                                  Page 1 of x
                              FEATURE ACCESS CODE (FAC)
            Abbreviated Dialing List1 Access Code: ____
            Abbreviated Dialing List2 Access Code: ____
            Abbreviated Dialing List3 Access Code: ____
   Abbreviated Dial - Prgm Group List Access Code: ____
                        Announcement Access Code: ____
                         Answer Back Access Code: ____
                           Attendant Access Code: ____
       Auto Alternate Routing (AAR) Access Code: ____
        Auto Route Selection (ARS) Access Code1: ____  Access Code 2: ____
                  Automatic Callback Activation: ____   Deactivation: ____
   Call Forwarding Activation Busy/DA: ____   All: ____   Deactivation: ____
                          Call Park Access Code: ____
                        Call Pickup Access Code: ____
   CAS Remote Hold/Answer Hold-Unhold Access Code: ____
                  CDR Account Code Access Code: ____
                        Change COR Access Code: ____
                   Change Coverage Access Code: ____
                     Contact Closure Open Code: ____     Close Code: ____
                    Contact Closure Pulse Code: ____
```

Abbreviated Dialing List1 Access Code

Used to access AD list 1.

Abbreviated Dialing List2 Access Code

Used to access AD list 2.

Abbreviated Dialing List3 Access Code

Used to access AD list 3.

Abbreviated Dial - Prgm Group List Access Code

Used to enter a group list from a telephone. The user's extension must be entered on the **Abbreviated Dial Group List** screen in order to program the group list.

Announcement Access Code

Used to record announcements.

Answer Back Access Code

Used to retrieve parked calls.

Attendant Access Code

This field only appears and is valid if an attd entry does not exist on the **Dial Plan Analysis** screen. You cannot have an entry in both the **Dial Plan Analysis** screen and the **Feature Access Code (FAC)** screen. While the **Dial Plan Analysis** screen allows administration of only one attd code that connects to one attendant, this field on the **Feature Access Code** screen allows you to administer more than one attendant access code in a single distributed network. Attendant access numbers can start with any number from 0 to 9 and contain 1 or 2 digits.

Auto Alternate Routing (AAR) Access Code

Used to access AAR.

Auto Route Selection (ARS) Access Code1

Used to access ARS. You can have one ARS access code for local and one for long distance, and route accordingly.

(ARS) Access Code 2

Also used to access ARS.

Automatic Callback Activation/Deactivation

Used to activate/cancel Automatic Callback.

Call Forwarding Activation Busy/DA

Used to forward calls to an administered number if the user is busy or does not answer.

(Call Forwarding Activation) All

Used to forward calls to an administered number for all calls.

(Call Forwarding) Deactivation

Used to deactivate call forwarding.

Call Park Access Code

Used to park an active call, which can then be retrieved from a different station using the answer back access code. Do not administer to have the same first digit as another feature access code that is longer in length.

Call Pickup Access Code

Used to answer a call directed to a pickup group.

CAS Remote Hold/Answer Hold-Unhold Access Code

Used by a Centralized Attendant Service (CAS) attendant to place calls on hold and answer calls held at a remote server running Communication Manager.

CDR Account Code Access Code

Used prior to entering an account code for CDR purposes.

Change COR Access Code

Used to allow users to change their class of restriction (COR) from a telephone. This field can only be used if the **Change COR by FAC** field is enabled on the **System-Parameters Customer-Options** screen.

Change Coverage Access Code

Used to change a coverage path from a telephone or remote station.

Contact Closure Close Code

Used to close a contact closure relay. Must be administered if the **Contact Closure Open Code** field is administered.

Contact Closure Open Code

Used to open a contact closure relay. Must be administered if the **Contact Closure Close Code** field is administered.

Contact Closure Pulse Code

Used to pulse a contact closure relay.

Field descriptions for page 2

Figure 327: Feature Access Code (FAC) screen

```
change feature-access-codes                                      Page  2  of x
                            FEATURE ACCESS CODE (FAC)
                  Data Origination Access Code:
                      Data Privacy Access Code:
              Directed Call Pickup Access Code:
    Emergency Access to Attendant Access Code:
                    Enhanced EC500 Activation:        Deactivation:
    Extended Call Fwd Activate Busy D/A_  All:        Deactivation:
       Extended Group Call Pickup Access Code:
               Facility Test Calls Access Code:
                           Flash Access Code:
            Group Control Restrict Activation:        Deactivation:
                  Hunt Group Busy Activation:         Deactivation:
                             ISDN Access Code:
                Last Number Dialed Access Code:
   Leave Word Calling Message Retrieval Lock:
  Leave Word Calling Message Retrieval Unlock:
             Leave Word Calling Send A Message: #66
           Leave Word Calling Cancel A Message: *66
               Malicious Call Trace Activation:       Deactivation:
         Meet-me Conference Access Code Change:
```

Data Origination Access Code

Used to originate a data call from a voice station.

Data Privacy Access Code

Used to isolate a data call from call waiting or other interruptions.

Directed Call Pickup Access Code

Used to establish directed call pickup.

Emergency Access To Attendant Access Code

Used to gain access to the attendant in an emergency situation. Such calls alert as emergency calls. This field cannot be used if the **Emergency Access to Attendant** field is not enabled on the **System-Parameters Customer-Options** screen.

Enhanced EC500 Activation

Used to activate extension to cellular where calls to an office number are extended to a cell telephone.

Extended Call Fwd Activate Busy D/A

Used to activate call forwarding from a telephone or remote location.

Extended Call Fwd Activate All

Used to activate call forwarding from a telephone or remote location.

Extended Call Fwd Deactivation

Used to deactivate call forwarding from a telephone or remote location.

Note:

An extension must have Station Security Codes administered to use the following FACs:

- Extended Call Forward All Activate
- Extended Call Forward Busy/Don’t Answer Activate
- Extended Call Forward Deactivate
- Change Coverage

Extended Group Call Pickup Access Code

The feature access code (FAC) users enter when a call directed to another pickup group is to be answered. Users must enter a valid "Pickup Number" following the Extended Group Call Pickup Access Code to complete the operation.

Facility Test Calls Access Code

Used to place activate a facility test call.

SECURITY ALERT:
To ensure the security of your system, leave Facility Test Calls Access Code blank except when actually testing trunks.

Flash Access Code

Used to generate trunk flash. This code ensures that the flash signal is interpreted by the central office switch, rather than by Avaya Communication Manager.

Group Control Restrict Activation / Deactivation

Used to change the restriction level for all users with a given class of restriction. Requires console permissions.

Hunt Group Busy Activation/Deactivation

Hunt Group members can dial these codes to place themselves in a busy state, and to become available again.

ISDN Access Code

Used to place an ISDN call without using ARS, AAR, or UDP.

Last Number Dialed Access Code

Used to redial the last number dialed from this station.

Leave Word Calling Message Retrieval Lock

Used to lock the display module on telephones. The lock function activates at a telephone by dialing this system-wide lock access code. This prevents unauthorized users from displaying, canceling, or deleting messages associated with the telephone. The **Lock Messages** field on the **Station** screen also must be enabled.

Leave Word Calling Message Retrieval Unlock

Used to unlock a telephone's display module. The lock function is canceled at the telephone by dialing this unlock FAC followed by the SCC.

Leave Word Calling Send A Message

Used to send a leave word calling message.

Leave Word Calling Cancel A Message

Used to cancel a leave word calling message.

Malicious Call Trace Activation

Used to activate a trace request on a malicious call.

Meet-me Conference Access Code Change

Allows the controlling user of a Meet-me Conference VDN to change the access code.

Field descriptions for page 3

Figure 328: Feature Access Code (FAC) screen

```
change feature-access-codes                                    Page 3 of x
                          FEATURE ACCESS CODE (FAC)

PASTE  (Display PBX data on Phone) Access Code:
  Personal Station Access (PSA) Associate Code:        Dissociate Code:
       Per Call CPN Blocking Code Access Code:
     Per Call CPN Unblocking Code Access Code:
                            Posted Messages:

                Priority Calling Access Code:
                         Program Access Code:
      Refresh Terminal Parameters Access Code:
             Remote Send All Calls Activation:       Deactivation:
             Self Station Display Access Code:
                    Send All Calls Activation:       Deactivation:
        Station Firmware Download Access Code:
                      Station Lock Activation:       Deactivation:
     Station Security Code Change Access Code:
             Station User Admin of FBI Assign:                 Remove:
  Station User Button Ring Control Access Code:
                        Telephone Activation: #*
            Terminal Dial-up Test Access Code:
```

PASTE (Display PBX data on telephone) Access Code

Allows users to view call center data on display phones. PASTE is used in conjunction with Avaya IP Agent.

Personal Station Access (PSA) Associate Code

Used to associate a telephone with the telephone features assigned to a user’s extension.

Personal Station Access (PSA) Dissociate Code

Used to remove the association between a physical telephone and an extension number. You cannot provide the code until Personal Station Access (PSA) on the **System Parameters Customer-Options** screen is **y**.

Per Call CPN Blocking Code Access Code

If CPN blocking is off for a trunk group, users can turn it on for a call by using this code. When they dial this code, the calling party number is not sent to the public network.

Per Call CPN Unblocking Code Access Code

If CPN blocking is on for a trunk group, users can turn it off for a call by using this code. When they dial this code, the calling party number is sent to the public network.

Posted Messages

Only appears if the **Posted Messages** field is set to **y** on the **System-Parameters Customer-Options** screen. Used to access the Posted Messages feature. See "Posted Messages" in *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205, for more information

Program Access Code

Used to program abbreviated dial buttons on an individual telephone.

Refresh Terminal Parameters Access Code

Used to update terminal parameters on an individual telephone when system settings have changed.

Remote Send All Calls Activation/Deactivation

Used to activate or deactivate the Send All Calls feature. Requires console permissions.

Self Station Display Activation

The **Self Station** field is not active. If set to a valid FAC, a digital station displays its primary extension number when the FAC is entered.

Send All Calls Activation/Deactivation

Used to activate or deactivate sending all calls to coverage with minimal or no alerting at the station.

Station Firmware Download Access Code

This field specifies the feature access code used for 2420/2410 DCP station firmware downloads.

Station Lock Activation/Deactivation

Used to activate or deactivate Station Lock.

Station Security Code Change Access Code

Enter the code the user must dial to change their Station Security Code. The SCC must be administered before the user can change it using this FAC. That is, a user cannot change a blank SCC.

Station User Admin of FBI Assign

Use to activate or deactivate Facility Busy Indicators.

Station User Button Ring Control Access Code

Used to control the ring behavior for each line appearance and bridged appearance from the station. Allows users to have their phones ring either silently or audibly.

Terminal Dial-Up Test Access Code

Used to perform tests on digital telephones to make sure that the telephone and the buttons are communicating properly with the server running Avaya Communication Manager. To use a Terminal Dial-up Test FAC on a telephone with bridged appearances, add a bridged-appearance of the principal telephone. See your Maintenance documentation for information about the Digital Terminal Remote Looparound Test.

Field descriptions for page 4

Figure 329: Feature Access Code (FAC) screen

```
change feature-access-codes                                    Page 4 of x
                            FEATURE ACCESS CODE (FAC)
Terminal Translation Initialization Merge Code:       Separation Code:
            Transfer to Voice Mail Access Code:
          Trunk Answer Any Station Access Code:
              User Control Restrict Activation:       Deactivation:
  Voice Coverage Message Retrieval Access Code:
 Voice Principal Message Retrieval Access Code: *80
           Whisper Page Activation Access Code:
```

Terminal Translation Initialization Merge Code

Enter the digits that must be dialed to install (merge) a station without losing any of its previous feature settings. The Terminal Translation Initialization Separation Code must have been used, or an **X** administered in the **Port** field of the **Station** screen, when the telephone was removed from its former location in order for the Terminal Translation Initialization Merge Code to be effective. (If you try to use this and it doesn't work, check the **Station** screen for this extension. If there is still a port assigned, type **X** in the **Port** field, then try the TTI merge again.)

Terminal Translation Initialization Separation Code

Enter the digits that must be dialed to remove (separate) a station from a location without losing any of its feature settings.

Transfer to Voice Mail Access Code

Enter the digits that must be dialed to allow coverage to transfer the caller to the original call recipient's AUDIX mail where the caller can leave a message. Do not administer this code to have the same first digit as another feature access code that is longer in length.

Trunk Answer Any Station Access Code

Enter the access code that station users must dial to answer calls alerting on night bells.

User Control Restrict Activation/Deactivation

Used to change the restriction level for a specific extension. Requires console permissions.

Voice Coverage Message Retrieval Access Code

Allows users to retrieve voice messages for another user (for whom they are a coverage point) via a digital display module.

Voice Principal Message Retrieval Access Code

Allows users to retrieve their own voice messages for another user via a digital display module.

Whisper Page Activation Access Code

Allows users to place a page to another user's telephone, when active on a call. The paged user, and not the other parties on the call, hears the page.

Field descriptions for page 5

The feature access codes on this page pertain only to ACD call centers.

Figure 330: Feature Access Code (FAC) screen

```
change feature-access-codes                                  Page 5 of x
                        FEATURE ACCESS CODE (FAC)

                  Automatic Call Distribution Features

                   After Call Work Access Code: ___
                             Assist Access Code: ___
                            Auto-In Access Code: ___
                           Aux Work Access Code: ___
                              Login Access Code: ___
                             Logout Access Code: ___
                          Manual-In Access Code: ___
    Service Observing Listen Only Access Code: ___
    Service Observing Listen/Talk Access Code: ___
                  Add Agent Skill Access Code: ___
               Remove Agent Skill Access Code: ___
          Remote Logout of Agent Access Code: ___
```

Add Agent Skill Access Code

Enter the digits an agent must dial to be able to add a skill to their current skill set.

After Call Work Access Code

Enter the code the agent must dial when the agent will be performing work-related ACD activities.

Assist Access Code

Enter the digit the agent must dial to request assistance from the split supervisor.

Auto-In Access Code

Enter the code the agent must dial to become automatically available to receive another ACD call each time a call is released.

Aux Work Access Code

Enter the code the agent must dial when the agent will be performing non-ACD activities.

Login Access Code

Enter the code the agent must dial to gain access to the ACD functions. This is a system-wide code for all ACD agents.

Logout Access Code

Enter the logout code the agent must enter to exit ACD. This is a system-wide logout code for all ACD agents.

Manual-In Access Code

Enter the code the agent must dial to receive a single, new ACD call upon the completion of an ACD call.

Note:

The following two fields appear only if **Service Observing (Remote/By FAC)** on the **System Parameters Customer-Options** screen is **y**.

Remove Agent Skill Access Code

Enter the digits an agent must dial to be able to remove a skill from their current skill set.

Note:

The next field is available only if **Vectoring (Basic)** and **Vectoring (Prompting)** have been enabled on the **System-Parameters Customer-Options** screen.

Remote Logout of Agent Access Code

Enter the digits you need to dial to remotely logout an idle ACD or EAS agent.

Service Observing Listen Only Access Code

Enter the code that must be dialed to allow a station with Service Observing permission (COR) to listen to other agent ACD calls without being heard on the ACD call.

Service Observing Listen/Talk Access Code

Enter the code that must be dialed to allow a station with Service Observing permission (COR) to both listen and be heard on an ACD call.

Note:

The following two fields appear only if **Expert Agent Selection (EAS) Enabled** is optioned on the **Feature-Related System-Parameters** screen.

Field descriptions for page 6

The feature access codes on this page pertain only to Call Vectoring/ Prompting features.

Figure 331: Feature Access Code (FAC) screen

```
change feature-access-codes                                    Page 6 of x
                            FEATURE ACCESS CODE (FAC)

                        Call Vectoring/Prompting Features

  Converse Data Return Code:
```

Converse Data Return Code

Enter a 1 to 4 digit number (# may be used as the first digit). If there is data to be returned to the switch, the Converse Data Return Code is outpulsed before the data to be passed is outpulsed.

Field descriptions for page 7

The feature access codes on this page pertain only to Hospitality features.

Figure 332: Feature Access Code (FAC) screen

```
change feature-access-codes                                  Page 7 of x
                            FEATURE ACCESS CODE (FAC)

                              Hospitality Features

                        Automatic Wakeup Call Access Code: *11___
              Housekeeping Status (Client Room) Access Code: _______
              Housekeeping Status (Client Room) Access Code: _______
              Housekeeping Status (Client Room) Access Code: _______
              Housekeeping Status (Client Room) Access Code: _______
              Housekeeping Status (Client Room) Access Code: _______
              Housekeeping Status (Client Room) Access Code: _______
                  Housekeeping Status (Station) Access Code: _______
                  Housekeeping Status (Station) Access Code: _______
                  Housekeeping Status (Station) Access Code: _______
                  Housekeeping Status (Station) Access Code: _______
                     Verify Wakeup Announcement Access Code: _______
                           Voice Do Not Disturb Access Code: _______
```

Automatic Wakeup Call Access Code

Enter the access code the user must dial to schedule or cancel a wakeup call.

Housekeeping Status (Client Room) Access Code

Enter the access code the housekeeper dials from the client's room to provide room status. These codes are transmitted to the Property Management System (PMS) for processing. You can assign a definition to the six codes on the **Hospitality** screen.

Housekeeping Status (Station) Access Code

Enter the access code the housekeeper must dial to provide room status. This access code must be dialed from designated telephones. There are four codes.

Verify Wakeup Announcement Access Code

Enter the access code the user can dial to verify a wakeup announcement.

Voice Do Not Disturb Access Code

Enter the access code the user must dial to enter or cancel a do not disturb request without using a display - through the use of voice prompting.

Field descriptions for Multimedia Features page

The feature access codes on this page pertain only to Multimedia Call Handling (MMCH).

Figure 333: Feature Access Code (FAC) screen

```
change feature-access-codes                                    Page 8 of x
                              FEATURE ACCESS CODE (FAC)
                                 Multimedia Features

                      Basic Mode Activation:
                  Enhanced Mode Activation:
              Multimedia Call Access Code:
      Multimedia Data Conference Activation:  Deactivation:
       Multimedia Multi-Address Access Code:
           Multimedia Parameter Access Code:
```

Basic Mode Activation

If you enter this FAC when your system is an Enhanced multimedia complex, it will revert to a Basic multimedia complex. If you enter this FAC when your system is a Basic mode station it will do nothing.

Enhanced Mode Activation

If you enter this FAC when your system is a Basic multimedia complex, it will become an Enhanced multimedia complex. If you enter this FAC when your system is an Enhanced mode station it will do nothing.

Multimedia Call Access Code

If you enter this FAC from any voice station, it indicates to Avaya Communication Manager that you are making an Enhanced mode multimedia call. If you originate a multimedia call with the multimedia call access code, it will originate a call according to the Default Multimedia Parameters selected on the **Feature Related System Parameters** screen.

Multimedia Data Conference Activation

If you enter this FAC from any voice station that is participating in a multimedia call, it will alert Avaya Communication Manager that you want to enable data collaboration with the other parties on the call. If you enter this FAC a second time, it will give denial treatment (since a collaboration session is already active). This FAC only applies to voice stations on servers equipped with ESM adjuncts.

Multimedia Data Conference Deactivation

If you enter this FAC from the telephone that enabled data collaboration on a multimedia mode call, it will deactivate the data session and revert to a voice and video call. If a user enters this FAC while participating in a data-collaboration multimedia call that the user did not initiate, the system responds with denial treatment.

Multimedia Multi-Address Access Code

The multimedia multi-address access code is similar to the multimedia call access code. It allows origination of a multimedia call from a voice station. It is used when the destination being dialed requires a different address for each of the 2 B-channels. For example, ISDN-BRI provided by a Central Office is provisioned with separate listed directory numbers for each B-channel. In order to make a 2B multimedia call to such a device, two sets of addresses must be entered.

Originating a multimedia call with the multimedia multi-address access code will originate a call according to the Default Multimedia Parameters selected on the **Feature-Related System-Parameters** screen.

Multimedia Parameter Access Code

This FAC can be entered by any voice station to indicate to Avaya Communication Manager that you want to initiate a multimedia mode call with a specific bearer capability. This FAC would be followed by a 1 or 2 to indicate the following parameter selections respectively: 2x64 (unrestricted initial system default), 2x56 (restricted).

Field descriptions for MLPP features page

The feature access codes on this page pertain only to Multiple Precedence and Preemption (MLPP) calls.

Figure 334: Feature Access Code (FAC) screen

```
change feature-access-codes                                  Page   8 of   x
                              FEATURE ACCESS CODE (FAC)
                                 MLPP Features

             Precedence Calling Access Code: 8


    WNDP PRECEDENCE ACCESS CODES:
                 Flash Override Access Code: 90
                           Flash Access Code: 91
                       Immediate Access Code: 92
                        Priority Access Code: 93
                         Routine Access Code: 94
```

Precedence Calling Access Code

Enter a Feature Access Code that conforms to your dial plan, to be used to access the Multiple Level Precedence and Preemption feature.

WNDP PRECEDENCE ACCESS CODES

The system uses a feature access code to determine the precedence level for a call when the Worlwide Numbering and Dial Plan (WNDP) feature is active. Different feature access codes will be assigned for each PRECEDENCE level. When the WNDP feature is not active, the user will dial the PRECEDENCE CALLING feature access code followed by a digit indicating the precedence level of the call.

Flash Override Access Code

Enter a FAC to correspond with the Flash Override preemption level.

Flash Access Code

Enter a FAC to correspond with the Flash preemption level.

Immediate Access Code

.Enter a FAC to correspond with the Immediate preemption level.

Priority Access Code

.Enter a FAC to correspond with the Priority preemption level.

Routine Access Code

.Enter a FAC to correspond with the Routine preemption level.

Feature-Related System Parameters

This screen implements system parameters associated with various system features.

Note:

This screen used to contain Call Coverage and Call Forwarding parameters. These fields have been moved to a separate screen, which you can access with the command **`change system-parameters coverage-forwarding`**.

Field descriptions for page 1

Figure 335: Feature-Related System Parameters screen

```
change system-parameters features                                    page 1 of x
                         FEATURE-RELATED SYSTEM PARAMETERS
                            Self Station Display Enabled? n
                                 Trunk-to-Trunk Transfer? none
 Automatic Callback - No Answer Timeout Interval (rings): 4_
                     Call Park Timeout Interval (minutes): 10
     Off-Premises Tone Detect Timeout Interval (seconds): 20_
                               AAR/ARS Dial Tone Required? y
                                       Music/Tone On Hold: music    Port: _______
           Music (or Silence) On Transferred Trunk Calls: all
                     DID/Tie/ISDN/SIP Intercept Treatment: attd

 Internal Auto-Answer of Attd-Extended/Transferred Calls? y
                 Automatic Circuit Assurance (ACA) Enabled? y
                                       ACA Referral Calls: local
                                 ACA Referral Destination: ________
            ACA Short Holding Time Originating Extension: ______
             ACA Long Holding Time Originating Extension: ______
          Abbreviated Dial Programming by Assigned Lists:
     Auto Abbreviated/Delayed Transition Interval(rings):
                  Protocol for Caller ID Analog Terminals: Bellcore
 Display Calling Number for Room to Room Caller ID Calls?
```

AAR/ARS Dial Tone Required

A second dial tone provides feedback to the user that additional dialing can occur.

Valid entries	Usage
y/n	Enter **y** to indicate a second dial tone is to be given to the calling party on a incoming tie or DID trunk call that is to be routed via AAR/ARS.

Abbreviated Dial Programming by Assigned Lists

Valid entries	Usage
y	Enter **y** to allow programming by station's assigned list.
n	Enter **n** if using Program Access code to indicate which personal list is to be programmed.

ACA Referral Calls

Indicates where ACA referral calls generate. This field only appears when the **Automatic Circuit Assurance (ACA) Enabled** field is **y**.)

Valid entries	Usage
local	Local referral calls generate on and for the local switch.
primary	Primary referral calls generate on the local switch for remote servers/switches as well as the local switch.
remote	Remote referral calls generate at another server in a DCS network. In this case, the remote node number must also be entered. The remote node number is the same node number as defined on the **Dial Plan** screen. Also, **ACA** button status transmits to other servers/ switches when in a DCS network.

ACA Referral Destination

The specified extension should be equipped with a display module. This field only appears if ACA Referral Calls is **local** or **primary**.

Valid entries	Usage
An extension	Enter the extension on a local server running Communication Manager that is to receive the ACA referral call.
attd	Enter **attd** for attendant.

ACA Remote PBX Identification

This field only appears if **ACA Referral Calls** is **remote**.

Valid entries	Usage
1 to **63**	Enter a number to identify the switch in a DCS network that makes the referral call. Do not define the remote server/switch identified in this field as **local** on the system's **Dial Plan** screen.

ACA Short Holding Time Originating Extension and ACA Long Holding Time Originating Extension

Valid entries	Usage
An unassigned extension	Do not use the same extension number for both fields. The specified extensions are assigned automatically by the system when the screen is submitted. These fields only display if **ACA Referral Calls** is **local** or **primary**.

Auto Abbreviated/Delayed Transition Interval (rings)

Valid entries	Usage
1 to **16**	Enter the number of rings before an automatic abbreviated/ delayed transition is triggered for a call.

Automatic Callback — No Answer Timeout Interval (rings)

Valid entries	Usage
2 to **9**	Enter the number of times the callback call rings at the calling station before the callback call is canceled.

Automatic Circuit Assurance (ACA) Enabled

If **Automatic Circuit Assurance (ACA) Enabled** is **n**, associated ACA fields will not display.

Must have an **aca-halt** button administered on the user's station. If you enable this feature, complete the following ACA-related fields.

Valid entries	Usage
y	Enter **y** if ACA measurements will be taken.
n	Otherwise, enter **n**.

Call Park Timeout Interval (minutes)

Valid entries	Usage
1 to **90**	Enter the number of minutes a call remains parked before it cancels.

DID/Tie/ISDN/SIP Intercept Treatment

Valid entries	Usage
A recorded announcement extension	Toll charges do not apply to DID and private network calls routed to an announcement. **Note:** If entering a Multi-Location Dial Plan shortened extension, note the following: When entering a Multi-Location Dial Plan shortened extension in a field designed for announcement extensions, certain administration end validations that are normally performed on announcement extensions are not done, and resultant warnings or submittal denials do not occur. The shortened extensions also do not appear in any display or list that shows announcement extensions. Extra care should be taken to administer the correct type of announcement for the application when assigning shortened extensions.
attd	For system security, Avaya recommends entering **attd** in this field. This routes intercept calls to the attendant and, if the attendant receives several of these, they will know a problem exists.

Display Calling Number for Room to Room Caller ID Calls

Valid entries	Usage
y/n	Enter **y** to display the calling number for room to room hospitality calls.

Internal Auto-Answer of Attd-Extended/Transferred Calls

This only applies to digital telephones (except BRI) with a headset or speakerphone capability.

Valid entries	Usage
attd-extended	Enter **attd-extended** to enable IAA for only attendant extended calls.
both	Enter **both** to enable IAA for station transferred and attendant extended calls.
none	Enter **none** to disable IAA for all calls.
transferred	Enter **transferred** to enable IAA for only station transferred calls.

Music (or Silence) On Transferred Trunk Calls

Valid entries	Usage
all	Enter **all** to allow all transferred trunk calls to receive music until the call is answered if the Music-on-Hold feature is available.
no	Enter **no** if trunk callers are to hear music (or silence if Music-on-Hold is not administered) while waiting to be transferred, and then ringback as soon as the transfer is completed till the call is answered.
call-wait	Enter **call-wait** if trunk calls transferred to stations that require the call to wait hear music (if administered); all other transferred trunk calls receive ringback tone.

Music/Tone on Hold

If you use equipment that rebroadcasts music or other copyrighted materials, you may be required to obtain a copyright license from, or pay fees to, a third party. You can purchase a Magic OnHold system, which does not require such a license, from Avaya or our business partners. This field does not appear if Tenant Partitioning is **y** on the **System-Parameters Customer-Options** screen. In that case, use the **Tenant** screen to establish Music on Hold.

Valid entries	Usage
music **tone** **none**	Indicates what a caller hears while on hold.

Off-Premises Tone Detect Timeout Interval (seconds)

Valid entries	Usage
5 to **25**	The number of seconds a call progress tone receiver (CPTR) tries to detect dial tone from a trunk during dialing. Once the time-out interval occurs, the call either outpulses on the trunk or gets intercept treatment depending on the setting of the **Outpulse Without Tone** field on page 6 of this screen.

Port

Indicates the port number that provides Music-on-Hold access. This requires a port on a TN763 Auxiliary Trunk circuit pack or any supported Analog Line circuit pack. Appears when **Music/ Tone on Hold** is **music**. Enter the necessary characters.

Valid entries	Usage
01 to **44** (DEFINITY R) or **01** to **03** (DEFINITY CSI, SI) or **1** to **64** (S8700 or S8710 IP-Connect / S8700 or S8710 Multi-Connect / S8300 Media Server)	First and second characters are the cabinet number
A to **E**	Third character is the carrier
0 to **20**	Fourth and fifth character are the slot number
01 to **04** (Analog TIE trunks) **01** to **31**	Six and seventh characters are the circuit number

1 of 2

Valid entries	Usage
1 to **80** (DEFINITY R, CSI) or **1** to **10** (DEFINITY SI) or **1** to **250** (S8700 or S8710 IP-Connect / S8700 or S8710 Multi-Connect / S8300 Media Server)	Gateway
V1 to **V9**	Module
01 to **31**	Circuit
	2 of 2

Protocol for Caller ID Analog Terminals

Determines the protocol/tones sent to a Caller ID telephone.

Valid entries	Usage
Bellcore	Enter **Bellcore** for Bellcore protocol with 212 modem protocol tones. Used in the U.S. and similar countries.
V23-Bell	Enter **V23-Bell** for Bellcore protocol with V.23 modem tones. Used in Bahrain and similar countries.

Self Station Display Enabled

Use this field to control the use of the **inspect** button for digital display phones.

Self Station Display allows a user to display the primary extension associated with a digital display telephone. There are two methods: (1) enter a feature access code (FAC), and (2) use the **inspect** button. In either case, the display shows the primary extension associated with the telephone where the FAC or **normal** or **exit** button is entered. In the case of the FAC, the display continues until a display-altering event occurs (for instance, going on-hook or receiving an incoming call). In the case of the **inspect** button, the display continues until the user presses the **normal** or **exit** button or until a display-altering event occurs.

Valid entries	Usage
y	The primary extension does display when the **inspect** button is pressed.
n	The extension does not display when the **inspect** button is pressed.

Trunk-to-Trunk Transfer

Regulations in some countries control the settings for this field. See your Avaya technical support representative for assistance.

Valid entries	Usage
all	Enter **all** to enable all trunk-to-trunk transfers. This allows telephone users to set up trunk-to-trunk transfer, go on-hook without disconnecting the call, and forward the call to a remote location.
restricted	Enter **restricted** (restricted public) to restrict all public trunks (CO, WATS, FX, CPE, DID, and DIOD).
none	Enter **none** to restrict all trunks (except CAS and DCS) from being transferred.

Field descriptions for page 2

Figure 336: Feature-Related System Parameters screen

```
change system-parameters features                              page 2 of x
                     FEATURE-RELATED SYSTEM PARAMETERS
LEAVE WORD CALLING PARAMETERS
              Maximum Number of Messages Per Station: 10
  Maximum Number of External Calls Logged Per Station: 0
       Message Waiting Indication for External Calls? n
  Stations with System-wide Retrieval Permission (enter extension)
    1: 34430___ 3: attd___ 5: _________7: _________9: ________
    2: 34412___ 4: ________ 6: _________8: _________10: ________
WARNING!   SEE USER DOCUMENTATION BEFORE CHANGING TTI STATE
            Terminal Translation Initialization (TTI) Enabled? _
             TTI State: _______    TTI Security Code: _______
         Customer Telephone Activation (CTA) Enabled? _
      Record CTA/PSA/TTI Transactions in History Log? _
                       COR for PSA Dissociated Sets: _
                  CPN, ANI for PSA Dissociated Sets: _
      Prohibit Bridging Onto Calls with Data Privacy? _
           Enhanced Abbreviated Dial Length (3 or 4)? _
       Record All Submission Failures in History Log? _
           Record PMS/AD Transactions in History Log?_
              Record IP Registrations in History Log?
Default Multimedia Outgoing Trunk Parameter Selection: 2x64
```

LEAVE WORD CALLING PARAMETERS

Maximum Number of Messages Per Station

Valid entries	Usage
0 to **125**	The maximum number of LWC Messages that can be stored by the system for a telephone at a given time.

Maximum Number of External Calls Logged Per Station

When an external call is not answered, the server running Communication Manager keeps a record of up to 15 calls (provided information on the caller identification is available) and the telephone's message lamp lights. The telephone set displays the names and numbers of unsuccessful callers

.

Valid entries	Usage
0 to **15**	The maximum number of calls that can be logged for each user. The assigned number cannot be larger than the entry in the **Maximum Number of Messages Per Station (when MSA not in service)** field.

Message Waiting Indication for External Calls

Provides a message waiting indication when external calls are logged.

Valid entries	Usage
y	The message waiting indication for a particular station is on whenever an external call is logged.
n	The log of external calls has no impact on the message waiting indication.

COR for PSA Dissociated Sets

Valid entries	Usage
1 to **95** or blank	Enter the class of restriction to apply to calls made from dissociated telephones. See "Personal Station Access" in *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205, for more information.

CPN, ANI for PSA Dissociated Sets

Appears when the **COR for PSA Dissociated Sets** field is non-blank. Specifies the ISDN calling party number (CPN), R2-MFC ANI, and CAMA CESID applied to calls made from PSA dissociated sets, if no system-wide calling party information has been administered for those protocols on their respective administration screens.

Valid entries	Usage
1 to 20 digits	Enter the calling party number or automatic number identification for calls made from dissociated telephones.

Customer Telephone Activation (CTA) Enabled

Valid entries	Usage
y/n	If **y**, you can use #* to automatically activate telephones after administration. See Installing New Phones for more information.

Default Multimedia Outgoing Trunk Parameter Selection

Does not appear on S8700 or S8710 IP-Connect.

Valid entries	Usage
2x56	Sets default parameter for bandwidth and bearer for all video calls.
2x64	

Enhanced Abbreviated Dial Length (3 or 4)

The administrator may not be able to use all entry slots because of system capacity constraints.

Valid entries	Usage
3	A value of **3** makes 1000 Enhanced List entries available to the administrator
4	A value of **4** makes 10,000 entries available.

Prohibit Bridging Onto Calls with Data Privacy

Valid entries	Usage
y/n	Enter **y** to protect calls from bridge-on by any party, including Service Observing, Intrusion, Verify, and Bridging.

Record All Submission Failures in History Log

Allows submission failures to be recorded on the history log.

Valid entries	Usage
y/n	Enter **y** to record submission failures on the history log.

Record CTA/PSA/TTI Transactions in History Log

Use this field to record when extensions and physical phones move between ports without additional administration from the administrator of Avaya Communication Manager.

Valid entries	Usage
y/n	Enter **y** to start records for ACTR, TTI, and PSA transactions (extension and telephone moves between ports) in the history log. You access these transactions via the List History command.

Record IP Registrations in History Log

Allows the logging of IP registrations in the history log.

Valid entries	Usage
y/n	Enter **y** to record IP registrations on the history log.

Record PMS/AD Transactions in History Log

Allows PMS and abbreviated dialing button transactions to be recorded on the history log.

Valid entries	Usage
y/n	Enter **y** to record PMS or abbreviated dialing button transactions on the history log.

Stations With System-wide Retrieval Permission (enter extension)

An extension must be removed from this list before the station is removed from the system. The server running Communication Manager refers to the extensions on this list as "super-retrievers."

Valid entries	Usage
An assigned extension.	Enter up to 10 telephone extension numbers that can retrieve LWC Messages or External Call Log records for all other telephones. A VDN extension is not allowed.
attd	An entry of **attd** gives retrieval permission to all attendants.

Field descriptions for page 3

Figure 337: Feature-Related System Parameters screen

```
change system-parameters features                              page 3 of x

                        FEATURE-RELATED SYSTEM PARAMETERS
TTI/PSA PARAMETERS

   WARNING! SEE USER DOCUMENTATION BEFORE CHANGING TTI STATE

        Terminal Translation Initialization (TTI) Enabled?  __
              TTI State: ________        TTI Security Code:  ________
           Record CTA/PSA/TTI Transactions in History Log? __
         Enhanced PSA Location/Display Information Enabled? __
                     Default COR for PSA Dissociated Sets: ___

              Customer Telephone Activation (CTA) Enabled? __
```

Customer Telephone Activation (CTA) Enabled

Valid entries	Usage
y	Enter **y** if you want the Customer Telephone Activation feature for your system.
n	Enter n if you do not want the Customer Telephone Activation feature for your system.

Default COR for PSA Dissociated Sets

Valid entries	Usage
0 to **95** or blank	Specify the Class of Restriction (COR) that the system uses for calls made from dissociated telephones.

Enhanced PSA Location/Display Information Enabled

The system displays this field only when the when the **Terminal Translation Initialization (TTI)** field is **y**

Valid entries	Usage
y	Enter **y**, if you want the system to display: • PSA login and associated station information when a station is PSA associated. • PSA logout and the port when a station is PSA dissociated.
n	Enter n if you do not want the sytsem to display PSA information.

Record CTA/PSA/TTI Transaction in History Log

Valid entries	Usage
y	Enter **y** if you want the system to record Cutomer Telephone Activation (CTA), Personal Station Activation (PSA), and TTI transactions in the system history log.
n	Enter **n** if you do not want the system to record Customer Telephone Activation (CTA), Personal Station Activation (PSA), and TTI transactions in the system history log.

Terminal Translation Initialization (TTI) Enabled

For more information on TTI, see "Terminal Translation Initialization" in *Feature Description and Implementation for Avaya Communication Manager*, 555-245-205. You should contact your Avaya technical support representative before making changes to TTI settings.

Valid entries	Usage
y	Enter **y** to start ACTR, TTI, and PSA transactions (extension and telephone moves between ports). You can administer this field only if the **Terminal Trans. Init. (TTI)** field on the **Customer Options** screen is **y**.
n	Enter **n** to remove existing TTI port translations and make sure no new TTI port translations are generated.

TTI Security Code

Valid entries	Usage
1 to 7 digits	Enter a one-digit to seven-digit number that a TTI user must use when the user accesses TTI from a telephone or data terminal. The system displays this field only when the **Terminal Translation Initialization (TTI)** field is **y**.

TTI State

Enter the type of port translation that you want for the system to use for unadministered digital ports. The default is voice.

Valid entries	Usage
data	Enter **data**, if you want a stand-alone data module to the be the TTI port translation for your system. The activation and deactivation sequence is entered at data terminal.
resume	Enter **resume**, if you want TTI to be available after TTI has been manually suspended. The state of TTI returns to the state that it was in before TTI was manally suspended.
suspend	Enter **suspend** to make TTI voice or TTI data translataions temporarily unavailable. The system does not remove existing TTI translations.
voice	Enter **voice**, if you want voice or voice/data terminal to be the TTI port translation for the system. The activation and deactivation sequence is entered from a telephone.

Field descriptions for page 4

Figure 338: Feature-Related System Parameters screen

```
change system-parameters features                              page 4 of x
                     FEATURE-RELATED SYSTEM PARAMETERS
                      Reserved Slots for Attendant Priority Queue: 5_
                                       Time Before Off-Hook Alert: 10__
                                 Emergency Access Redirection Extension: _________
              Number of Emergency Calls Allowed in Attendant Queue: __
                                            Call Pickup Alerting? n
                      Temporary Bridged Appearance on Call Pickup? y
                                    Call Pickup on Intercom Calls? y
                                            Directed Call Pickup? n
                                       Extended Group Call Pickup: flexible
                 Deluxe Paging and Call Park Timeout to Originator? n
     Controlled Outward Restriction Intercept Treatment: tone________
    Controlled Termination Restriction (Do Not Disturb): tone________
             Controlled Station to Station Restriction: tone________
AUTHORIZATION CODE PARAMETERS           Authorization Code Enabled? y
                                         Authorization Code Length: 7
                              Authorization Code Cancellation Symbol? #
                                           Attendant Time Out Flag? n
                                       Display Authorization Code? _
                  Controlled Toll Restriction Replaces: station-station
          Controlled Toll Restriction Intercept Treatment: extension     3000
```

Call Pickup Alerting

This provides pickup group members with a visual indication on the Call Pickup status lamp of calls eligible to be answered via Call Pickup

Valid entries	Usage
y/n	Enter **y** to enable Call Pickup Alerting on a system-wide basis.

Call Pickup on Intercom Calls

Valid entries	Usage
y	Enter **y** to allow a user's or Agent LoginID's call, ringing as an intercom call, to be picked up using the Call Pickup or Directed Call Pickup features. This field controls the use of this feature throughout the system.
n	Enter **n** to prevent the use of these features to pickup an intercom call.

Controlled Outward Restriction Intercept Treatment

Enter the type of intercept treatment the caller receives when the call is outward restricted.

Valid entries	Usage
announcement	Provides a recorded announcement to calls that cannot be completed as dialed. You select and record the message. The calling party receives indication that the call is receiving Intercept Treatment. Enter the extension number for the announcement in the associated field. **Note:** If entering a Multi-Location Dial Plan shortened extension, note the following: When entering a Multi-Location Dial Plan shortened extension in a field designed for announcement extensions, certain administration end validations that are normally performed on announcement extensions are not done, and resultant warnings or submittal denials do not occur. The shortened extensions also do not appear in any display or list that shows announcement extensions. Extra care should be taken to administer the correct type of announcement for the application when assigning shortened extensions..
attendant	Allows attendants to provide information and assistance to outgoing calls that cannot be completed as dialed or that are transferred to incomplete or restricted stations.
extension	Enter the extension number for the extension in an associated field. May not be a VDN extension.
tone	Provides a siren-type tone to internal calls that cannot be completed as dialed

Controlled Station-to-Station Restriction

Enter the type of intercept treatment the caller receives when the call is placed to a restricted telephone.

Valid entries	Usage
announcement	If announcement is entered, an associated extension number field displays. Enter the extension of the restricted telephone in the field.
attendant	Intercepted calls are redirected to the attendant.
extension (may not be a VDN extension)	If extension is entered, an associated extension number field displays. Enter the extension of the restricted telephone in the field.
tone	Intercepted calls receive intercept (siren) tone.

Controlled Termination Restriction (Do Not Disturb)

Enter the type of intercept treatment the caller receives when the call is placed to a termination restricted telephone.

Valid entries	Usage
announcement	If **announcement** is entered, complete an associated extension number field.
attendant	Redirects intercepted calls to the attendant.
coverage	Redirects intercepted calls to coverage.
extension	If **extension** is entered, complete an associated extension number field. May not be a VDN extension,
tone	Provides a siren-type tone to calls that cannot be completed as dialed.

Deluxe Paging and Call Park Timeout to Originator

Paged calls that are to be parked require separate activation of the Call Park feature. All parked calls that time out return to the attendant.

Valid entries	Usage
y	Enter **y** to enable the Loudspeaker Paging - Deluxe feature that essentially integrates the Loudspeaker Paging and Call Park features. All parked calls that time out (not answered by paged party) return to the parking party.
n	Enter **n** to enable the Loudspeaker Paging feature.

Directed Call Pickup

Feature use by individual stations, attendants, or EAS agents can be controlled by COR.

Valid entries	Usage
y	Enter **y** to allow use of the Directed Call Pickup feature across the system.
n	Enter **n** to prevent feature use.

Emergency Access Redirection Extension

Valid entries	Usage
An assigned extension	Enter the assigned extension number (can be a VDN) where emergency queue overflow will redirect.

Extended Group Call Pickup

Enables call pickup groups to answer calls directed to another call pickup group.

Valid entries	Usage
flexible	Flexible feature version supporting a one-to-n (pickup group-to-extended pickup group) mapping.
simple	Simple feature version with a one-to-one pickup group-to-extended pickup group mapping supported.
none	Extended group call pickup not supported.

Number of Emergency Calls Allowed in Attendant Queue

Valid entries	Usage
0 to **75**	Enter the number of emergency calls allowed in the attendant queue before additional calls are routed to the backup extension.

Reserved Slots for Attendant Priority Queue

Valid entries	Usage
2 to **75**	Enter the number of calls that can go in to the emergency queue

Temporary Bridged Appearance on Call Pickup

Valid entries	Usage
y	Enter **y** to allow a temporary bridged appearance for calls answered with the Call Pickup or Directed Call Pickup features. This field controls this capability on a system-wide basis.
n	Enter **n** to prevent the temporary bridged appearance of calls answered with these features.

Time Before Off-Hook Alert

Valid entries	Usage
1 to **3000** seconds	Enter the time in seconds that a telephone with an Off-Hook Alert Class of Service can remain off-hook (after intercept tone has started) before an emergency call is sent to the attendant.

AUTHORIZATION CODE PARAMETERS

Attendant Time Out Flag

If this field is not enabled, the caller receives Intercept tone. This flag affects only remote users or incoming calls over trunks requiring an authorization code. This field only appears if **Authorization Codes Enabled** is **y**.

Valid entries	Usage
y/n	Enter **y** if a call is to be routed to the attendant if the caller does not dial an authorization code within 10 seconds or dials an invalid authorization code.

Authorization Code Cancellation Symbol

Enter the symbol a caller must dial to cancel the 10-second wait period during which the user can enter an authorization code. This field only appears when **Authorization Code** is **y**.

Valid entries	Usage
#	Enter the cancellation code # if the main and tandem servers/switches are both of the same type.
1	Enter the cancellation code **1** if an Avaya System 85 or DIMENSION PBX switch is part of the complex/ network.

Authorization Code Length

This field only appears and must be completed if **Authorization Codes Enabled** is **y**. This is the number of digits that must be assigned to the **Authorization Code (AC)** field on the **Authorization Code** screen.

SECURITY ALERT:
You enhance your system's security by using the maximum length for your authorization code.

Valid entries	Usage
4 to 13 digits	Enter a number that defines the number of digits (length) in the **Authorization Code** field.

Authorization Codes Enabled

This field cannot be administered if Authorization Codes is not enabled on the **System-Parameters Customer-Options** screen.

SECURITY ALERT:
To maintain system security, Avaya recommends that Authorization Codes be used.

Valid entries	Usage
y/n	Enter **y** to enable Authorization Codes on a systemwide basis.

Controlled Toll Restriction Intercept Treatment

Appears when the **Controlled Toll Restriction Replaces** field is **outward** or **station-to-station**. This field applies an intercept treatment to a toll call during the call processing.

Valid entries	Usage
announcement	A sub-field appears to the right if **announcement** is used. If the entry is **announcement**, enter the assigned announcement extension.
attendant	Intercepted calls are redirected to the attendant.
extension	A sub-field appears to the right if **extension** is used. If the entry is **extension**, enter the extension assigned to station or individual attendant.
tone	Intercepted calls receive intercept (siren) tone.

Controlled Toll Restriction Replaces

This field activates the Controlled Toll Restriction feature.

Valid entries	Usage
outward **station-station** **none**	The value that you choose for this field will be replaced by controlled toll restriction. In other words, if you choose station-station, you will not be able to use station-station restrictions unless you reset this field.

Display Authorization Code

This field applies only to DCP, not to BRI or hybrid sets.

SECURITY ALERT:
To enhance your system's security, set **Display Authorization Code** to **n**.

Valid entries	Usage
y	Enter **y** to allow authorization code digits to display on the set during the dialing.
n	Enter **n** if these digits should not display.

Field descriptions for page 5

Figure 339: Feature-Related System Parameters screen

```
change system-parameters features                                  page 5
                        FEATURE-RELATED SYSTEM PARAMETERS

SYSTEM PRINTER PARAMETERS
              System Printer Endpoint: ____                Lines Per Page: 60
                  EIA Device Bit Rate: 9600

Emergency Extension Forwarding (min): 10

SYSTEM-WIDE PARAMETERS
                        Switch Name: _________________



MALICIOUS CALL TRACE PARAMETERS
               Apply MCT Warning Tone? n     MCT Voice Recorder Trunk Group: ___
      Delay Sending Release (seconds)?
SEND ALL CALLS OPTIONS
           Send All Calls Applies to: station
      Auto Inspect on Send All Calls? n

UNIVERSAL CALL ID
     Create Universal Call ID (UCID)? n
                                          UCID Network Node ID: ___
```

SYSTEM PRINTER PARAMETERS

The system printer is the printer dedicated to support scheduled reports.

EIA Device Bit Rate

This field does not appear for DEFINITY R, S8700 or S8710 IP-Connect, S8700 or S8710 Multi-Connect

Valid entries	Usage
1200 **2400** **4800** **9600**	Enter the required printer speed setting.

Lines Per Page

Valid entries	Usage
24 to **132**	Enter the number of lines per page required for the report.

System Printer Endpoint

Note:

The **eia** option is not available for DEFINITY R.

Valid entries	Usage
Data module extension	Does not appear for S8700 or S8710 IP-Connect. Associated with the System printer.
eia	Does not appear for S8700 or S8710 IP-Connect. If the DCE jack is used to interface the printer.
SYS_PRNT	Use this value if the system printer is connected over a TCP/IP link, and the link is defined as SYS_PRNT on the **IP Services** screen.
blank	

Emergency Extension Forwarding (min)

If an emergency call should drop (get disconnected), the public safety personnel will attempt to call back. If the ELIN that was sent was not equivalent to the caller's extension number, the return call would ring some other set than the one that dialed 911. To overcome that limitation, you can automatically forward that return call to the set that placed the emergency call for an administered period of time.

This Emergency Extension Forwarding only applies if the emergency location extension number is an extension on the same PBX as the extension that dialed 911. Customers who have several PBXs in a campus should assign emergency location extensions accordingly.

This field sets the Emergency Extension Forwarding timer for all incoming trunk calls if an emergency call gets cut off (drops).

Valid entries	Usage
0 to **999**	Type a number between 0 and 999 that represents the time (in minutes) that an incoming trunk call will forward to the extension that made the initial 911 call. The default value for both new installs and upgrades is 10.

Note:

If a user at the emergency location extension (the extension that made the initial 911 call) manually turns off the Call Forwarding feature, the feature is off no matter how many minutes might remain on the timer.

SYSTEM-WIDE PARAMETERS

Switch Name

Valid entries	Usage
Any keyboard character	Enter up to 20 alpha-numeric characters for identification.

MALICIOUS CALL TRACE PARAMETERS

Apply MCT Warning Tone

Valid entries	Usage
y/n	Enter **y** to provide an audible tone to the controlling station when an MCT recorder is actively recording a malicious call.

Delay Sending RELease (seconds)

This field specifies the amount of time DEFINITY waits before sending an ISDN release message in response to receiving an ISDN disconnect message. This field appears only if, on the **System-Parameters Customer-Options** screen, the **Malicious Call Trace** field is **y**.

Valid entries	Usage
0 to **30**	Enter the number in increments of 10.

MCT Voice Recorder Trunk Group

Assign the trunk group for MCT voice recorders.

Valid entries	Usage
1 to **666** or blank	group number for DEFINITY R, CSI, SI
1 to **2000** or blank	group number for S8700 or S8710 IP-Connect

SEND ALL CALLS OPTIONS

Auto Inspect on Send All Calls

Valid entries	Usage
y	If set to **y**, allows you to be presented automatically with Calling Party information for calls which are silently alerting their station because of the Send-All-Calls feature.
n	If set to **n**, you are not guaranteed a Calling Party display for calls sent directly to Coverage by the Send-All-Calls feature.

Send All Calls Applies to

Valid entries	Usage
station	If set to **station**, any call to that station, regardless of the number dialed, causes calls to that station's own extension to be sent immediately to Coverage, or causes calls to different extensions assigned to the station as bridged appearances to become Ring-Ping notification if **Redirect Notification** field is **y**.
extension	When set to **extension**, only the calls sent to that extension are placed to coverage.

UNIVERSAL CALL ID

Create Universal Call ID (UCID)

Valid entries	Usage
y	If set to **y**, DEFINITY will generate UCID for each call when necessary.
n	If set to **n**, the DEFINITY will not generate a UCID for any call.

UCID Network Node ID

Enter a number unique to this server/switch in a network of switches.

Valid entries	Usage
1 to **32767** or blank	This number is an important part of the UCID tag and must be unique to the server/switch.

Field descriptions for page 6

Figure 340: Feature-Related System Parameters screen

```
change system-parameters features                                  page 6
                        FEATURE-RELATED SYSTEM PARAMETERS
       Public Network Trunks on Conference Call: 5              Auto Start? n
    Conference Parties with Public Network Trunks: 6             Auto Hold? n
 Conference Parties without Public Network Trunks: 6        Attendant Tone? y
         Night Service Disconnect Timer (seconds): 180       Bridging Tone? n
                  Short Interdigit Timer (seconds): 3       Conference Tone? n
               Unanswered DID Call Timer (seconds): ____     Intrusion Tone? n
               Line Intercept Tone Timer (seconds): 30    Special Dial Tone? n
                  Long Hold Recall Timer (seconds):     Mode Code Interface? n
                       Reset Shift Timer (seconds): 0
     Station Call Transfer Recall Timer (seconds): 0
                             DID Busy Treatment: tone
       Invalid Number Dialed Intercept Treatment: Announcement _____
              Allow AAR/ARS Access from DID/DIOD? _
                Allow ANI Restriction on AAR/ARS? _
     Use Trunk COR for Outgoing Trunk Disconnect? _
                 7405ND Numeric Terminal Display? n           7434ND? n
DISTINCTIVE AUDIBLE ALERTING
             Internal: 1   External: 2   Priority: 3
                         Attendant Originated Calls:
   DTMF Tone Feedback Signal to VRU - Connection: _    Disconnection: _
```

7405ND Numeric Terminal Display

Valid entries	Usage
y/n	If enabled, this allows you to use 7405ND in the **Type** field of the **Station** screen. This is not an actual telephone type, but you can use this to define ports for certain types of Octel Messaging Division voice messaging systems. This numeric display setting sends only numbers, and not names, to the Octel system.

7434ND

Valid entries	Usage
y/n	If enabled, this allows you to use 7434ND in the **Type** field of the **Station** screen. This is not an actual telephone type, but you can use this to define ports for certain types of Octel Messaging Division systems. Use this value if your voice messaging system operates in Bridged Mode.

Allow AAR/ARS Access from DID/DIOD

Valid entries	Usage
y/n	Enter **y** to allow calls for DID and DIOD type trunk groups to complete calls using ARS or AAR.

Allow ANI Restriction on AAR/ARS

(For Russia only) If a call is placed over a Russian shuttle trunk or a Russian rotary trunk via an AAR or ARS entry with the **ANI Req** field set to **r**, then ANI is requested just like a **y** entry. However, if the ANI request fails, the call immediately drops. All other trunk types treat the **r** entry as a **y**.

Valid entries	Usage
y	The **ANI Req** field on the AAR and ARS Digit Analysis Table and the AAR and ARS Digit Conversion Table permits the additional value of **r**(estricted).
n	The **ANI Req** field on the two screens takes only the current values of **n** and **y**.

Attendant Tone

Valid entries	Usage
y/n	Enter **y** to provide call progress tones to the attendants.

Auto Hold

Valid entries	Usage
y/n	Enter **y** to enable the Automatic Hold feature on a system-wide basis.

Auto Start

If this field is enabled, the **Start** buttons on all attendant consoles are disabled.

Valid entries	Usage
y/n	Enter **y** to enable the Automatic Start feature.

Bridging Tone

Valid entries	Usage
y/n	Enter **y** to apply a bridging tone when calls are bridged on primary extensions.

Conference Parties with Public Network Trunks

If the value of the **Public Network Trunks on Conference Call** field is **0**, this field will not appear on the screen.

Valid entries	Usage
3 to **6**	Specify the maximum number of parties allowed in a conference call involving a public network subscriber.

Conference Parties without Public Network Trunks

Valid entries	Usage
3 to **6**	Enter a number to specify the maximum number of parties allowed in a conference call involving no public network trunks.

Conference Tone

Note:

Bridging and Conference Tones are not supported by all countries. If these tones are enabled for countries other than Italy, Belgium, United Kingdom, or Australia, the tones will be equivalent to no tone (silence) unless the tone is independently administered or customized on the **Tone Generation** screen.

Valid entries	Usage
y/n	Enter **y** to provide conference tone as long as three or more calls are in a conference call.

DID Busy Treatment

Specifies how to handle a direct inward dialing (DID) call to a busy station.

Valid entries	Usage
attendant	Call is routed to attendant.
tone	Caller hears a busy tone.

Intrusion Tone

Valid entries	Usage
y/n	Enter **y** to apply an intrusion tone (executive override) when an attendant intrudes on the call.

Invalid Number Dialed Intercept Treatment

Enter the type of intercept treatment the end-user hears after dialing an invalid number.

Valid entries	Usage
announcement	Provides a recorded announcement when the end-user dials an invalid number. You select and record the message. Enter the extension number for the announcement in the associated field.
tone	Default. Provides intercept tone when the end-user dials an invalid number.

Line Intercept Tone Timer (seconds)

Valid entries	Usage
0 to **60**	Enter a number to specify how long an analog station user can wait after hearing warning tone without going on hook, before the station is placed in the lockout state.

Long Hold Recall Timer (seconds)

You can administer the system to remind a user that a call has been on hold for too long.

Valid entries	Usage
0 to **999**	Enter a number between **0** and **999**; **0** deactivates the timer. This value is the number of seconds a call can be on hold before the system re-alerts the user to remind them of the call.

Mode Code Interface

Note:

If you make a change to this field, you must log off and log back on to effect the permission changes to get to the Mode Code Related System Parameters on page 1283.

Valid entries	Usage
y/n	A **y** entry allows you to use the Mode Code Voice Mail System Interface to connect the server running Communication Manager over a DTMF interface to INTUITY AUDIX or other vendors' voice-mail systems.

Night Service Disconnect Timer (seconds)

Valid entries	Usage
10 to **1024** or blank	Enter a number or blank to indicate how long a trunk call can be unanswered during night service before being disconnected. The trunk must not have Disconnect Supervision for this timer to apply.

Public Network Trunks on Conference Call

Indicates the number of public network trunks allowed on a conference call

Valid entries	Usage
0 to **5**	If this field is **0**, the **Conference Parties with Public Network Trunks** field will not appear on the screen.

Reset Shift Timer (seconds)

Used only for station-to-station calls or private network calls using ISDN trunks.

Valid entries	Usage
0 to **255**	Specifies the number of seconds that reset shift dial tone is audible before busy tone is heard. Reset shift dial tone allows the user to dial a new extension by dialing one new digit that replaces the last digit of the extension previously dialed. The new digit replaces the last digit of the extension previously dialed. Enter **0** to disable this feature.

Short Interdigit Timer (seconds)

Valid entries	Usage
3 to **9**	Enter a number to limit the time that digit analysis will wait for the next digit when it has predicted that all the digits have already been collected.

Special Dial Tone

Special dial tone notifies an analog-telephone user if certain features are still active when the user goes off-hook. These features include:

- Call Forwarding
- Send All Calls
- Do Not Disturb

Valid entries	Usage
y/n	Enter **y** to use the Special Dial Tone. You must have a TN2182 circuit pack.

Station Call Transfer Recall Timer (seconds)

Allows a user-transferred call (station-to-station, a trunk call, or a DCS call) to re-terminate with priority ringing back to the station user who initiates the transfer operation if the transfer-to party does not answer the call within the administered Station Call Transfer Recall timer.

Valid entries	Usage
0 to **999**	Enter the time in seconds before a call redirects back to the station user who initiated the transfer operation. Enter **0** to disable this feature.

Unanswered DID Call Timer (seconds)

Enter number or blank to limit how long a DID call can remain unanswered before routing to the DID/TIE/ISDN Intercept Treatment feature. This timer interacts with the nonadministrable 50 second Wait for Answer Supervision Timer (WAST). The WAST timer overrides this field. Thus if this field is set to a value equal to or greater than 50 seconds, the caller receives intercept tone instead of the normal attendant or announcement treatment that is given when the Unanswered DID Call Timer expires before the WAST. If the Unanswered DID Call Timer expires while the DID call is being processed by call vectoring, the timer is ignored. See Wait Answer Supervision Timer in this section.

Valid entries	Usage
A number between **10** and **1024**	Enter a number to indicate how long a DID call can remain unanswered before routing to the DID/TIE/ISDN Intercept Treatment feature.
blank	Disables the timer.

Use Trunk COR for Outgoing Trunk Disconnect

Use this field to indicate whether the Outgoing Trunk Disconnect Timer is set based on the COR of the originating station or of the trunk group. If enabled, the timer is based on the COR of the trunk, not the originating caller's station.

Valid entries	Usage
n	Default. The Outgoing Trunk Disconnect Timer to be set based on the COR of the originating station.
y	Enter **y** to enable the Outgoing Trunk Disconnect Timer to be set based on the COR of the trunk instead of the originating station.

DISTINCTIVE AUDIBLE ALERTING

Attendant Originated Calls

Valid entries	Usage
internal **external** **priority**	Indicates which type of ringing (defined above) applies to attendant-originated calls.

Distinctive Audible Alerting (Internal, External, Priority)

This is also known as Distinctive Ringing.

Valid entries	Usage
1 to **3**	Enter the number of rings for internal, external and priority calls. For virtual stations, this applies to the mapped-to physical telephone.

DTMF Tone Feedback Signal to VRU - Connection, Disconnection

This field appears only if **DTMF Feedback Signals for VRU** on the **Customer-Options System Parameters** screen is **y**.

Valid entries	Usage
0 to **9, *, #, A, B, C, D**	Enter the code to connect or disconnect the VRU. This can be a single digit, or a combination such as *99 to connect, #99 to disconnect. The tones must be programmed at the VRU as well.
blank	Blank means that no tone is to be sent to the VRU.

Field descriptions for page 7

Figure 341: Feature-Related System Parameters screen

```
change system-parameters features                                    page 7
                         FEATURE-RELATED SYSTEM PARAMETERS

CONFERENCE/TRANSFER

               Abort Transfer? n             No Dial Tone Conferencing? n
        Transfer Upon Hang-Up? n   Select Line Appearance Conferencing? n
Abort Conference Upon Hang-Up? n                                Unhold? n
    No Hold Conference Timeout: 60


ANALOG BUSY AUTO CALLBACK
                Without Flash?                      Announcement:
                                          Voice Mail Hunt Group Ext:

AUDIX ONE-STEP RECORDING
Apply Ready Indication Tone To Which Parties In The Call? all
  Interval For Applying Periodic Alerting Tone (seconds): 15
```

CONFERENCE/TRANSFER

Abort Conference Upon Hang-Up

Allows DCP, hybrid, IP, wireless, or ISDN-BRI telephone users to abort the conference operation when they hang up.

Valid entries	Usage
y/n	Enter **y** to change a call placed on soft-hold in the conference-pending status to hard-held status if the user hangs up.

Abort Transfer

Stops the transfer operation whenever a user presses a non-idle call appearance button in the middle of the transfer operation, or when they hang up. If both the **Abort Transfer** and **Transfer Upon Hang-Up** fields are **y** and you press the **transfer** button and then dial the complete transfer-to number, hanging up the telephone transfers the call. You must select another non-idle call appearance to abort the transfer. If the **Transfer Upon Hang-Up** field is **y**, hanging up completes the transfer. Requires DCP, Hybrid, IP, ISDN-BRI or wireless phones.

Valid entries	Usage
y/n	Enter **y** to abort the transfer a call by pressing the **Transfer** button, dialing the desired extension, and then hanging up or selecting another non-idle call appearance. The user must press the **Transfer** button again to complete the process unless **Transfer Upon Hang-up** is also set to **y**.

No Dial Tone Conferencing

When another line is on hold or alerting, No Dial Tone Conferencing eliminates dial tone while setting up a conference. See GuideBuilder for more information.

Valid entries	Usage
y/n	Enter **y** to activate No Dial Tone Conferencing.

No Hold Conference Timeout

Controls the timeout of No Hold Conference call setup. The system Answer Supervision timer should be set to a value less than this.

Valid entries	Usage
20 to **120**	Enter the number of seconds.

Select Line Appearance Conferencing

Select Line Appearance Conference changes the capabilities of the conference buttons and line appearance buttons on digital phones. See GuideBuilder for more information.

Valid entries	Usage
y/n	Enter **y** to activate Select Line Appearance Conferencing

Transfer Upon Hang-Up

Allows DCP, hybrid, IP, wireless, or ISDN-BRI telephone users to complete a transfer operation by hanging up.

Valid entries	Usage
y/n	Enter **y** so users may transfer a call by pressing the **Transfer** button, dialing the desired extension, and then hanging up. The user may also wait to hang up, speak with the other party, then press **Transfer** again to complete the process. With this field set to **y**, users of the Call Park FAC can park a call without having to press the **Transfer** button a second time.

Unhold

Allows the user to press the **hold** button on a telephone to release a hold (if no other line appearance is on hold or alerting). This does not apply to BRI phones or attendant consoles.

Valid entries	Usage
y/n	Enter **y** to activate the unhold capability

ANALOG BUSY AUTO CALLBACK

With the Analog Busy Auto Callback Without Flash (ACB) feature enabled, when a caller places a call through an analog station, and the called station is busy and has no coverage path nor forwarding, then an announcement plays, announcing that the station is busy and prompting the caller to enter **1** for ACB or **2** to cover to a voice mail hunt group extension.

Announcement

Appears only if the **Without Flash** field is **y**.

Valid entries	Usage
Extension number	Enter the extension of the announcement you want to play for the ACB feature. This field cannot be left blank.

Voice Mail Hunt Group Ext

Appears only if the **Without Flash** field is **y**.

Valid entries	Usage
Extension number	Enter a voice mail hunt group extension to which the call is to be forwarded if the user enters 2 at the ACB announcement prompt..

Without Flash

Provides automatic callback for analog stations without flashing the hook. It is applied only when the called station is busy and has no other coverage path or call forwarding. The caller can enable the automatic callback without flashing the hook or entering the feature access code.

Note:

If the **Analog Busy Auto Callback Without Flash** field is set to **y**, the **Busy Auto Callback without Flash** field on the **Station** screen defaults to **y** (enabled) for all analog station types that allow Analog Auto Callback.

Valid entries	Usage
y/n	Enter **y** to provide automatic callback for a calling analog station without flashing the hook.

AUDIX ONE-STEP RECORDNG

On stations administered with this feature button, this allows users to activate and deactivate the recording of active calls to their Audix with the press of one button.

Apply Ready Indication Tone To Which Parties In The Call

This field is for administering who hears the Audix recording ready tone.

Valid entries	Usage
all, initiator, or **none**	Enter a value for which party or parties on the call should hear the ready-to-record indication tone. The default is **all**. This field cannot be left blank.

Interval For Applying Periodic Alerting Tone (seconds)

Appears only if the **Apply Ready Indication Tone To Which Parties In The Call** field is set to **all**.

Valid entries	Usage
0 to **60**	Enter a number from zero to 60 for the number of seconds desired between alerting tones, where zero disables the tone. The default value is a 15 second interval.

Field descriptions for page 8

Figure 342: Feature-Related System Parameters screen

```
change system-parameters features                                    page 8
                          FEATURE-RELATED SYSTEM PARAMETERS

ISDN PARAMETERS

Send Non-ISDN Trunk Group Names as Connected Name?
 Display Connected Name/Number for ISDN DCS Calls?
       Send ISDN Trunk Group Name on Tandem calls?

               Send Custom Messages Through QSIG?
                                  QSIG TSC Extension:
MWI - Number of Digits Per Voice Mail Subscriber:

                               National CPN Prefix:
                          International CPN Prefix:
                         Pass Prefixed CPN to ASAI:
   Unknown Numbers Considered Internal for AUDIX?
             UNSI Calling Name for Outgoing Calls?
               Path Replacement with Measurements?
                  QSIG Path Replacement Extension:
            Path Replace While in Queue/Vectoring?
```

ISDN PARAMETERS

Display Connected Name/Number for ISDN DCS Calls

Valid entries	Usage
y/n	Enter **y** to display the connected name/number (if received) for ISDN DCS calls.

Feature Plus Ext

Valid entries	Usage
A valid extension	Administration of this field is required for proper termination of some Feature Plus signaling. For example, Message Waiting Indication (MWI) requires this extension in order to send the indication to the appropriate server running Communication Manager. Appears only if the **ISDN Feature Plus** field is **y** on the **System-Parameters Customer-Options** screen.

International CPN Prefix

Allows you to apply prefixes to international calling numbers for display at receiving phones. This is useful for those phones that use or implement call back features based on incoming call numbers. When an ISDN-PRI call arrives, the incoming call setup is analyzed for: (1) whether the Type of Address (TOA) is national or international, and (2) whether the Numbering Plan Identifier (NPI) is Unknown or ISDN/Telephony. This administered prefix is applied to international calls. Prefixing applies to any subsequent display on the same server when the call is transferred, covered, or forwarded. The same prefixing applies to outgoing ISDN-PRI calls when the connected number information is returned and meets the same TOA and NPI criteria. The prefix plus the calling/connected number digit string is limited to 15 digits, with truncation occurring at the least significant digits.

Valid entries	Usage
1 to 5 digits, (**0** to **9**), * and # or blank	Enter a number that allows you to apply prefixes to international calling numbers for display.

Maximum Length

Appears only if the **Unknown Numbers Considered Internal for AUDIX** field is **y**. Indicates the maximum length of an unknown private number. Any unknown number longer than the administered value is considered external. This field cannot be blank when it appears.

Valid entries	Usage
1 to **20**	Enter a number for the maximum length of an unknown private number.

MWI - Number of Digits Per Voice Mail Subscriber

Appears only if the **Basic Supplementary Services** field or the **ISDN Feature Plus** field on the **System-Parameters Customer-Options** screen is **y**. This field provides an indication of the number of digits per AUDIX subscriber.

Note:

For QSIG-MWI only. These routing digits and inserted digits must screen a digit string that, when analyzed and processed, routes to a Signaling Group supporting QSIG-TSCs. Once a QSIG TSC is established (from a message-center server/switch to a served-user switch), then every lamp update message places the **Inserted Digits** field (from the **Message Waiting Indication Subscriber Number Prefixes** screen) in front of the AUDIX subscriber number to screen a complete QSIG network number for the served user.

Note:

For Feature Plus MWI only. The routing digits and inserted digits must screen a digit string that routes over an SSF trunk to the Feature Plus extension on the remote (served user) switch. The **Inserted Digits** field must include the Feature Plus extension.

Valid entries	Usage
3 to **7**	Enter a value that corresponds to the digit string length of subscribers translated in the Message Center entity. For instance, if the Message Center entity is AUDIX, the value in this field must match the value of the **Extension Length** field on the **Switch Interface Administration** screen of AUDIX.

National CPN Prefix

Allows you to apply prefixes to national calling numbers for display at receiving phones. This is useful for those phones that use or implement call back features based on incoming call numbers. When an ISDN-PRI call arrives, the incoming call setup is analyzed for: (1) whether the Type of Address (TOA) is national or international, and (2) whether the Numbering Plan Identifier (NPI) is Unknown or ISDN/Telephony. This administered prefix is applied to national calls. Prefixing applies to any subsequent display on the same server when the call is transferred, covered, or forwarded. The same prefixing applies to outgoing ISDN-PRI calls when the connected number information is returned and meets the same TOA and NPI criteria. The prefix plus the calling/connected number digit string is limited to 15 digits, with truncation occurring at the least significant digits.

Valid entries	Usage
1 to 5 digits, (**0** to **9**), * and # or blank	Enter a number that allows you to apply prefixes to national calling numbers for display.

Pass Prefixed CPN to ASAI

Passes Calling Party Number information (CPN) to ASAI. The prefixed number is not passed on to other adjuncts, Call Detail Recording, or servers/switches.

Valid entries	Usage
y/n	Enter **y** to pass CPN information to ASAI.

Path Replacement While in Queue/Vectoring

Valid entries	Usage
y/n	Enter **y** to allow Path Replacement after queue/vector processing has started. Depending on the version of CMS you are using, some calls can go unrecorded if you enable this capability. Please see your Avaya technical support representative for more information.

Path Replacement with Measurements

Valid entries	Usage
y/n	Allows QSIG path replacement or DCS with Reroute to be attempted on measured calls.

QSIG Path Replacement Extension

Enter the extension for the system to use as part of the complete number sent in the Path Replacement Propose message.

Valid entries	Usage
Extension	Enter an unused extension that conforms to your dial plan.

QSIG TSC Extension

Valid entries	Usage
Enter any valid, unassigned extension.	This is the phantom endpoint extension for QSIG Call Independent Signaling Connections (CISCs), which are similar to NCA Temporary Signaling Connections (TSCs) (both incoming and outgoing).

Send Custom Messages Through QSIG?

Valid entries	Usage
y/n	Enter **y** to provide appropriate display information, for example for the Posted Messages feature, over QSIG links.

Send ISDN Trunk Group Name on Tandem Calls

Valid entries	Usage
y/n	Enter **y** to provide consistent display information regardless of trunk type. If set to **y**, provides only trunk group name.

Send Non-ISDN Trunk Group Name as Connected Name

Valid entries	Usage
y/n	Enter **y** to send a name of the non-ISDN trunk group as the connected name when a call routes from ISDN to non-ISDN and the call is answered.

Unknown Numbers Considered Internal for AUDIX

Appears when, on the **System-Parameters Customer-Options** screen, either the **ISDN-PRI** or **ISDN-BRI Trunks** field is **y**. This field controls the treatment of an ISDN number whose numbering plan identification is "unknown" in a QSIG centralized AUDIX arrangement. This field works in conjunction with the Calling Party Number to INTUITY AUDIX field on the **Hunt Group** screen. The **Calling Party Number to INTUITY AUDIX** field on the **Hunt Group** screen must be **y** for this field to have an effect.

Valid entries	Usage
y	The unknown number is considered "internal" and AUDIX tries to find a calling party name match for the digit string. If a name match is found, AUDIX provides the calling party's name. If no name is found, AUDIX provides the calling party's telephone number.
n	The unknown number is considered "external" and AUDIX provides the calling party's telephone number.

USNI Calling Name for Outgoing Calls?

Valid entries	Usage
y/n	Enter **y** to send a name on outgoing calls over NI PRI trunks. Important: Be sure you have validated that your service provider's central office is capable of accepting calling name information from Communication Manager in this way. For example, if the central office has a 5ESS, it must be a generic 5EXX or later. Failure to validate the central office capability may cause the central office to drop outgoing calls from your media server. In this case, change the value in this field to **n**. Enter **n** to prevent sending calling name information with outgoing calls over NI PRI trunks. **n** is this field overrides a **y** in the **Send Name** field of the outgoing **Trunk Group** screen.

Field descriptions for page 9

Figure 343: Feature-Related System Parameters screen

```
change system-parameters features                              page 9 of x
                          FEATURE-RELATED SYSTEM PARAMETERS

CPN/ANI/ICLID PARAMETERS

    CPN/ANI/ICLID Replacement for Restricted Calls:
   CPN/ANI/ICLID Replacement for Unavailable Calls:

SEPARATION OF BEARER AND SIGNALING (SBS) PARAMETERS
               Local Country Code: 1
        International Access Code: 011
```

CPN/ANI/ICLID PARAMETERS

CPN/ANI/ICLID Replacement for Restricted Calls

Valid entries	Usage
up to 15 characters	Enter a text string to replace the restricted numbers on the display.

CPN/ANI/ICLID Replacement for Unavailable Calls

Valid entries	Usage
up to 15 characters	Enter a text string to replace the unavailable numbers on the display.

SEPARATION OF BEARER AND SIGNALING (SBS) PARAMETERS

Local Country Code

Valid entries	Usage
1 to 3 digits or blank	Enter a valid country code for this node. The default is blank (no SBS signaling trunk groups are administered). For example, for an SBS node in the United States, enter **1** .

International Access Code

Valid entries	Usage
1 to 5 digits or blank	Enter the access code required by the PSTN to route calls out of the country. This code will be included with the telephone number received from the SBS terminating node if the Local Country Codes of the originating and terminating nodes are different. The default is blank (no access code is needed). For example, for an SBS node in the United States, enter **011** .

Note:

Once administered, these fields cannot be cleared until all trunk groups administered for SBS signaling have been removed. For details, see the **Trk Grp(s)** and **Signaling Group** screens.

Field descriptions for page 10

Figure 344: Feature-Related System Parameters screen

```
change system-parameters features                                  page 10 of x
                        FEATURE-RELATED SYSTEM PARAMETERS

                 Pull Transfer: n            Update Transferred Ring Pattern? n
         Outpulse Without Tone? y              Wait Answer Supervision Timer? n
         Misoperation Alerting? n               Repetitive Call Waiting Tone? y
    Allow Conference via Flash? y     Repetitive Call Waiting Interval (sec): _
 Vector Disconnect Timer (min): __    Network Feedback During Tone Detection? y
   Hear Zip Tone Following VOA? y    System Updates Time On Station Displays? n
      Intercept Treatment on Failed Trunk Transfers? n
                    Station Tone Forward Disconnect: silence
                             Level Of Tone Detection: precise
           Charge Display Update Frequency (seconds): 30
         Date Format on 607/2400/4600/6400 Terminals: mm/dd/yy
On-hook Dialing on 607/2400/4600/6400/8400 Terminals? n

ITALIAN DCS PROTOCOL
  Italian Protocol Enabled? y
   Apply Intercept Locally? _      Enforce PNT-to-PNT Restrictions? _
```

Allow Conference via Flash

Valid entries	Usage
y	Enter **y** to allow an analog station to use flash to conference calls together.
n	Enter **n** to prevent this.

Charge Display Update Frequency (seconds)

This applies only if you use Advice of Charge or Periodic Pulse Metering with display functions.

Valid entries	Usage
10 to **60** or blank	The amount of time (in seconds) between charge-display updates. Frequent display updates may have considerable performance impact. If the duration of a call is less than the Charge Display Update Frequency, the display will not automatically show charge information. To see charge information for a call, the user must have a **disp-chrg** button and must press the button before the call drops.

Date Format on 607/2400/4600/6400 Terminals

The format of the date as displayed on the telephones.

Valid entries	Usage
mm/dd/yy	month/day/year
dd/mm/yy	day/month/year
yy/mm/dd	year/month/day

Hear Zip Tone Following VOA?

This tone alerts a telephone user that the announcement has completed and a caller is now connected. CallMaster set and attendant console users hear double zip tone following the announcement. All other telephone users hear single zip tone.

Note:

This field does not effect auto-answer zip tone heard prior to the VOA.

Valid entries	Usage
y	Enter **y** to play zip tone following a VDN of Origin Announcement (VOA).
n	Enter **n** if you do not want zip tone following a VOA.

Intercept Treatment on Failed Trunk Transfers

Valid entries	Usage
y	Enter **y** to provide intercept treatment to calls failing trunk transfers.
n	Enter **n** to drop these calls.

Level of Tone Detection

For the most part, this option is no longer required in today’s switching environment. It may be useful if your users are having difficulty placing outgoing calls due to inaccurate detection of network dial tone.

Valid entries	Usage
broadband	This is the least exact of the levels of tone detection. If Avaya Communication Manager detects any tone at all, it interprets this as dial tone.
medium	The server running Avaya Communication Manager interprets any tone which has a continuous "on" period of longer than 1 second as dial tone. Otherwise, the server accepts whatever the tone detector circuit pack reports.
precise	Communication Manager accepts whatever the tone detector circuit pack reports.

Misoperation Alerting

Misoperation Alerting should not be enabled if Call Prompting is optioned.

Valid entries	Usage
y	Enter **y** for misoperation recall alerting on multi-appearance stations, analog stations, and attendant consoles.
n	Enter **n** for standard misoperation handling without recall alerting.

Network Feedback During Tone Detection

Valid entries	Usage
y/n	Enter **y** to provide audible feedback to the user while the system attempts to detect dial tone.

On-hook Dialing on 607/2400/4600/6400/8400 Terminals

For 6400/8400, 607, 2420, 2410, and 4600 telephone users with speakerphones.

Valid entries	Usage
y/n	Enter **y** allows users to dial without lifting the handset. If you enable this, users hear dial tone when they press the **Speaker** button, even if the handset is on-hook.

The next four fields control station-to-switch recall signal timing. If a flashhook interval (recall window) is required, the upper and lower bounds of the interval can be administered. An on-hook that lasts for a period of time greater than or equal to the lower bound and less than or equal to the upper bound will be treated as a recall flash. If an interval is not required, the **Disconnect Timing** value must be administered. An on-hook that lasts for a period of time less than this value will be ignored; greater than or equal to this value will be regarded as a disconnect. Regardless, an on-hook lasting 50 to 150 ms coming from a 2500-type set will always be treated as a digit pulse unless **Ignore Rotary Digits** is **y** for that station.

Outpulse Without Tone

Valid entries	Usage
y	Enter **y** to indicate the server will outpulse digits even when a dial tone has not been received.
n	Enter "**n**" if the calling party should receive intercept tone if no dial tone is detected.

Pull Transfer

Valid entries	Usage
y/n	Enter **y** to enable the Pull Transfer feature on a system-wide basis. This allows either the transferring or transferred-to party to press the **Transfer** button to complete the transfer operation

Repetitive Call Waiting Interval (sec)

This field appears when the **Repetitive Call Waiting Tone** field is **y**.

Valid entries	Usage
1 to **99**	Enter a number to specify the number of seconds between call waiting tones.

Repetitive Call Waiting Tone

Valid entries	Usage
y/n	Enter **y** to indicate that a repetitive call waiting tone be provided to the called party for all types of call waiting access.

Station Tone Forward Disconnect

Tone Forward Disconnect applies to any station other than one administered as a data endpoint, an attendant console, a BRI telephone, an auto answer, or as an Outgoing Call Management (OCM) agent.

Valid entries	Usage
busy **intercept** **silence**	When a station is the last party remaining off-hook on a call, that station receives the indicated tone or silence until that station is placed on-hook, or until the tone has played for 45 seconds and is followed by silence.

System Updates Time On Station Displays

This does not apply to phones (such as BRI phones) where the user sets the time.

Valid entries	Usage
y/n	Enter **y** to have the system automatically update the time on display phones when background maintenance is run (for example, when the set is plugged in).

Update Transferred Ring Pattern

Valid entries	Usage
y/n	Enter **y** to change the ringing pattern from internal to external when an internal station transfers an external call. If most of your calls go through an attendant, you might want to set this to **y**, so your users will be able to distinguish an external call.

Vector Disconnect Timer (min)

Enter the number of minutes, or blank that a trunk should remain connected to a vector.

Valid entries	Usage
1 to **240**	The number of minutes that you enter determines when the trunk will be disconnected if the **Disconnect Supervision-In** or **Disconnect Supervision-Out** fields on the **Trunk Group** screen are **n**.
blank	Enter blank if you do not want Avaya Communication Manager to initiate a disconnect.

Wait Answer Supervision Timer

See Unanswered DID Call Timer (seconds) for more information.

Valid entries	Usage
y	Enter **y** to enable this feature on a systemwide basis. When **y** is entered in this field, calls to stations unanswered after 50 seconds are dropped.
n	When **n** is entered in this field, unanswered calls drop only when the calling party goes on-hook.

ITALIAN DCS PROTOCOL

The next three fields control the Italian DCS Protocol feature.

Apply Intercept Locally

This field appears if the **Italian Protocol Enabled** field is **y**.

Valid entries	Usage
y/n	Enter **y** to indicate that DID/CO intercept treatment will be applied locally instead of on the originating server/ switch.

Enforce PNT-to-PNT Restrictions

This field appears if the **Italian Protocol Enabled** field is **y**.

Valid entries	Usage
y/n	Enter **y** to indicate that restrictions and denial of PNT-to-PNT connections will be enforced when the EDCS message is unavailable. A **y** in this field means restrictions will be enforced.

Italian Protocol Enabled

Valid entries	Usage
y/n	Enter **y** to enable the Italian DCS feature on a systemwide basis.

Field descriptions for page 11

Figure 345: Feature-Related System Parameters screen

```
change system-parameters features                              page 11 of x
                      FEATURE-RELATED SYSTEM PARAMETERS

CALL CENTER SYSTEM PARAMETERS
  EAS
        Expert Agent Selection (EAS) Enabled? n
       Minimum Agent-LoginID Password Length:
         Direct Agent Announcement Extension: _________          Delay: ___
   Message Waiting Lamp Indicates Status For: station
VECTORING
                   Converse First Data Delay: 0       Second Data Delay: 2
              Converse Signaling Tone (msec): 100          Pause (msec): 70_
                    Prompting Timeout (secs): 10
                Interflow-qpos EWT Threshold: 2
   Reverse Star/Pound Digit For Collect Step? n
         Available Agent Adjustments for BSR? _
SERVICE OBSERVING
             Service Observing: Warning Tone? n      or Conference Tone?
    Service Observing Allowed with Exclusion? n
```

CALL CENTER SYSTEM PARAMETERS

EAS

Direct Agent Announcement Delay

Only appears if Expert Agent Selection (EAS) or **ASAI Link Core Capabilities** on the **System-Parameters Customer-Options** screen is **y**.

Valid entries	Usage
0 to **99** or blank	Enter the number of seconds the caller will hear ringback before the Direct Agent Announcement is heard by the calling party.

Direct Agent Announcement Extension

Valid entries	Usage
Valid extension	Enter the extension of the direct agent announcement.

Expert Agent Selection (EAS) Enabled

To enable this field, either no ACD or vectoring hunt groups may exist or, existing ACD or vectoring hunt groups must be "skilled." Only appears if **Expert Agent Selection (EAS)** on the **System-Parameters Customer-Options** screen is **y**.

Valid entries	Usage
y/n	Enter **y** to enable Expert Agent Selection.

Message Waiting Lamp Indicates Status For

Only appears if **Expert Agent Selection (EAS)** on the **System-Parameters Customer-Options** screen is **y**.

Valid entries	Usage
station	Since you only have one message waiting lamp on a telephone, you need to indicate if the message is for at the telephone extension or the loginID.
loginID	Expert Agent Selection (EAS) must be enabled to use this option.

Minimum Agent-LoginID Password Length

Enter the minimum number of digits that must be administered as an EAS Agent's LoginID password. Only appears if **Expert Agent Selection (EAS)** on the **System-Parameters Customer-Options** screen is **y**.

Valid entries	Usage
0 to **9**	Entering a **0** or blank indicates no password is required.

VECTORING

Available Agent Adjustments for BSR

Controls the use of BSR available agent adjustments. The **Vectoring (Best Service Routing)** field must be **y** on the **System-Parameters Customer-Options** screen.

Valid entries	Usage
y/n	Enter **y** to allow adjustments to available agents.

Converse First Data Delay/Second Data Delay

The First Data Delay prevents data from being outpulsed (as a result of a converse vector step) from the system to CONVERSANT before CONVERSANT is ready. The delay commences when the CONVERSANT port answers the call. The Second Data Delay is used when two groups of digits are being outpulsed (as a result of a converse vector step) from the system to CONVERSANT. The Second Data Delay prevents the second set from being outpulsed before CONVERSANT is ready. The delay commences when the first group of digits has been outpulsed. Only appears if **Vectoring (Basic)** on the **System-Parameters Customer-Options** screen is **y**.

Valid entries	Usage
0 to **9**	Number of seconds for the delay.

Converse Signaling Tone/Pause

Only appears if **Vectoring (Basic)** and **DTMF** on the **System-Parameters Customer-Options** screen are **y**. In the **Signaling Tone** field, enter the length in milliseconds of the digit tone for digits being passed to the CONVERSANT. In the **Pause** field, enter the length in milliseconds of the delay between digits being passed. The optimum timer settings for the CONVERSANT or IR are 60 msec tone and 60 msec pause.

Valid entries	Usage
40 to **2550** (in increments of 10).	Values entered in the **Tone/Pause** fields are rounded up or down depending upon the type of circuit pack used to outpulse the digits.
100	• **TN742B or later suffix analog board** — Tone and pause round up or down to the nearest 25 msecs. For example a 130 msec tone rounds down to 125 msecs, a 70 msec pause rounds up to 75 msec for a total of 200 msecs per tone. • **TN464F, TN767E or later suffix DS1 boards** — Tone and pause round up to the nearest 20 msecs. For example a 130 msec tone rounds up to 140 msecs, a 70 msec pause rounds up to 80 msecs for a total of 220 msecs per tone. If a circuit pack has been used for end-to-end signalling to the CONVERSANT, and has then been used to send digits to a different destination, the CONVERSANT timers may stay in effect. To reset your timers to the system default, pull and reseat the circuit pack.

Interflow-qpos EWT Threshold

Displays only if, on the **System-Parameters Customer-Options** screen, the **Lookahead Interflow (LAI)** field is **y**. Part of enhanced Look-Ahead Interflow. Any calls predicted to be answered before this threshold will not be interflowed (therefore saving CPU resources).

Valid entries	Usage
0 to **9** or blank	Number of seconds for this threshold

Prompting Timeout (secs)

Only appears if **Vectoring (Prompting)** on the **System-Parameters Customer-Options** screen is **y**.

Valid entries	Usage
4 to **10**	Enter the number of seconds before the **`Collect Digits`** command times out for callers using rotary dialing.

Reverse Star/Pound Digit for Collect Step

The "*" is interpreted as a "caller end-of-dialing indicator and the "#" is an indicator to clear all digits previously entered by the caller for the current "collect" vector step.

Valid entries	Usage
y/n	Enter **y** to reverse the star and pound digits by the "collect" vector step. If set to **y**, it does not affect any other DEFINITY vector step or other non-ACD DEFINITY feature (such as ARS) in that the "*" and "#" digit-processing is unchanged.

SERVICE OBSERVING

or Conference Tone

Service Observing (Basic) on the **System-Parameters Customer-Options** screen must be **y** before this field may be administered.

Valid entries	Usage
y/n	Enter **y** to assign a conference tone to be given to telephone users and calling parties whenever their calls are being monitored using the Service Observing feature.

Service Observing: Warning Tone

Service Observing (Basic) on the **System-Parameters Customer-Options** screen must be **y** before this field may be administered.

CAUTION:

The use of Service Observing features may be subject to federal, state, or local laws, rules or regulations or require the consent of one or both of the parties to the conversation. Customers should familiarize themselves and comply with all applicable laws, rules, and regulations before using these features.

Valid entries	Usage
y/n	Enter **y** to assign a warning tone to be given to telephone users and calling parties whenever their calls are being monitored using the Service Observing feature.

Service Observing Allowed with Exclusion

Allows Service Observing of a station with Exclusion active, either by Class Of Service or by manual activation of Exclusion. Default is **n**.

Valid entries	Usage
y	Enter **y** to allow Service Observing of a station with Exclusion active, either by COS or by manual activation of Exclusion.
n	Default. Observing towards a station with Exclusion active is denied, or if Exclusion is activated by a station while being observed, all bridged parties including the observer are dropped.

Field descriptions for page 12

Figure 346: Feature-Related System Parameters screen

```
change system-parameters features                              page 12 of x
                        FEATURE-RELATED SYSTEM PARAMETERS

AGENT AND CALL SELECTION
                     MIA Across Splits or Skills? n
                      ACW Agents Considered Idle? y
                      Call Selection Measurement: current-wait-time
Service Level Supervisor Call Selection Override? y
                              Auto Reserve Agents:
        Copy ASAI UUI During Conference/Transfer?
ASAI
Call Classification After Answer Supervision? n      Send UCID to ASAI? n

CALL MANAGEMENT SYSTEM
                              Adjunct CMS Release:
                 ACD Login Identification Length: 0
                             BCMS/VuStats LoginIDs?
               BCMS/VuStats Measurement Interval: hour
       BCMS/VuStats Abandon Call Timer (seconds):
                 Validate BCMS/VuStats Login IDs? n
                        Clear VuStats Shift Data: on-login
             Remove Inactive BCMS/VuStats Agents? n
```

AGENT AND CALL SELECTION

ACW Agents Considered Idle

Valid entries	Usage
y/n	Enter **y** to have agents who are in After Call Work included in the Most-Idle Agent queue. This means that ACW is counted as idle time. Enter **n** to exclude ACW agents from the queue.

Auto Reserve Agents

When a critical skill is not meeting its service level, auto-reserve puts agents in standby for their other skills to ensure that there is an available agent when the next call arrives for the critical skill. When an agent becomes available, all of his or her assigned skills are checked to see if any auto-reserve skills are not meeting their target service level. If so, the agent is made available only in those skills.

Valid entries	Usage
all	Puts an agent on stand-by for all skills.
none	Agent is not on stand-by for any additional skills.
secondary-only	Puts an agent on stand-by only for secondary skills.

Call Selection Measurement

This field determines how Avaya Communication Manager selects a call for an agent when the agent becomes available and there are calls in queue.

For information on CentreVu Advocate, please contact your Avaya Account Executive or see the *Avaya Business Advocate Release 11 User Guide,* 585-210-711.

Valid entries	Usage
current-wait-time	**Current Wait Time** selects the oldest call waiting for any of the agent's skills.
predicted-wait-time.	**Predicted Wait Time** is a feature of CentreVu Advocate.

Copy ASAI UUI During Conference/Transfer

Displays when, on the **System-Parameters Customer-Options** screen, either the **ASAI Interface** or **ASAI Proprietary Adjunct Links** field is **y**.

Valid entries	Usage
y/n	Enter **y** to copy user-to-user (UUI) information during a conference or transfer calls.

Note:

When this field is set to **y**, the system actually copies *all* UUI information, not just ASAI UUI. Copying only occurs during a human-initiated conference or transfer. Communication Manager does not copy the UUI if the conference or transfer is initiated by ASAI.

MIA Across Splits or Skills

Valid entries	Usage
y/n	Enter **y** to remove an agent from the MIA queue for all the splits/skills/hunt groups that he or she is available in when the agent answers a call from any of his or her splits/skills/hunt groups.

Service Level Supervisor Call Selection Override

This field determines whether Avaya Communication Manager changes agents' call handling preferences when a skill using Service Level Supervisor exceeds its Level 1 threshold.

For information on CentreVu Advocate, please contact your Avaya Account Executive or see the *Avaya Business Advocate Release 3 Version 11 User Guide*.

Valid entries	Usage
y	Enter **y** if you want to override the normal call handling preferences of a skill's assigned agents in this situation.
n	Enter **n** if you do not want to override agents' normal call handling preferences when the skill exceeds its Level 1 threshold. Service Level Supervisor requires Expert Agent Selection and CentreVu Advocate.

ASAI

Call Classification After Answer Supervision?

For use with ASAI Outbound Call Management (OCM).

Valid entries	Usage
y/n	Enter **y** to force the server running Communication Manager to rely on the network to provide answer/busy/drop classification to the server. After the call has been answered, a call classifier can be added to perform answering machine, modem, and voice answering detection. The default value **n** always connects a classifier after call setup for determining call progress and answer. ISDN progress messages generally take precedence.

Send UCID to ASAI

Valid entries	Usage
y/n	Enter **y** to enable transmission of Universal Call ID (UCID) information to ASAI.

CALL MANAGEMENT SYSTEM

ACD Login Identification Length

Enter the number of digits for an ACD Agent Login ID if **Expert Agent Selection (EAS)** on the **System-Parameters Customer-Options** screen is **n**. If **BCMS/VuStats Login IDs** is **y**, the ACD Login ID length must be greater than 0. This field identifies an ACD agent to CMS. The number you enter in this field must equal the number of characters in the agent's login ID.

Valid entries	Usage
0 to **9**	For CMS, this field cannot be 0.

Adjunct CMS Release

Specifies the release of the CMS adjunct used with the system.

Valid entries	Usage
R3V6 **R3V8** **R3V9** **R3V11** **R3V12**	For CMS, this field cannot be blank.

BCMS/VuStats Measurement Interval

You can enter **half-hour** or **hour** for polling and reporting measurement data if the **BCMS (Basic)** and/or the **VuStats** on the **System-Parameters Customer-Options** screen is **y**.

Valid entries	Usage
half-hour	There are a maximum of 25 time slots available for measurement intervals. If **hour** is specified, an entire day of traffic information will be available for history reports; otherwise, only half a day will be available. This does not affect daily summaries as they always reflect traffic information for the entire day. The interval may be changed at any time, but will not go into effect until the current interval completes.
hour	

BCMS/VuStats LoginIDs

This feature can be used when EAS is not optioned, or in addition to EAS login IDs. When this field is **y**, both BCMS and CMS use the same login ID for an agent.

Valid entries	Usage
y/n	Enter **y** to administer valid agent login IDs to monitor call activity by agent.

BCMS/VuStats Abandon Call Timer (seconds)

Valid entries	Usage
1 to **10** or blank	Specify the number of seconds before calls are considered abandoned. Calls with talk time that is less than this number (and that are not held) are tracked by BCMS and displayed by VuStats as ABAND calls.

Clear VuStats Shift Data

Valid entries	Usage
on-login	Enter **on-login** to clear shift data for an agent when the agent logs in.
at-midnight	Enter **at-midnight** to clear shift data for all agents at midnight.

Remove Inactive BCMS/VuStats Agents

Valid entries	Usage
y	Agents are removed from reports when they have no staff time during the previous 7 days.
n	Agents remain on the report even if they have no staff time for any period of time.

Validate BCMS/VuStats Login IDs

Valid entries	Usage
y	Enter **y** to allow entry only of login-IDs that have been entered on the **BCMS Login-ID** screen.
n	Enter **n** to allow entry of any ACD login of the proper length.

Field descriptions for page 13

Figure 347: Feature-Related System Parameters screen

```
change system-parameters features                              page 13 of x
                         FEATURE-RELATED SYSTEM PARAMETERS
REASON CODES
                                 Aux Work Reason Code Type:
                                   Logoff Reason Code Type:

REDIRECTION ON IP CONNECTIVITY FAILURE

                Switch Hook Query Response Timeout:
            Auto-answer IP Failure AUX Reason Code:

MAXIMUM AGENT OCCUPANCY PARAMETERS

         Maximum Agent Occupancy Percentage:
   Maximum Agent Occupancy AUX Reason Code:
```

REASON CODES

Aux Work Reason Code Type

Valid entries	Usage
none	Enter **none** if you do not want an agent to enter a Reason Code when entering AUX work.
requested	Enter **requested** if you want an agent to enter a Reason Code when entering AUX mode but do not want to force the agent to do so. To enter **requested** the **Reason Codes** and **EAS** on the **System-Parameters Customer-Options** screen must be **y**.
forced	Enter **forced** to force an agent to enter a Reason Code when entering AUX mode. To enter **forced**, the **Reason Codes** and **EAS** on the **System-Parameters Customer-Options** screen must be **y**.

Logout Reason Code Type

Valid entries	Usage
none	Enter **none** if you do not want an agent to enter a Reason Code when logging out.
requested	Enter **requested** if you want an agent to enter a Reason Code when logging out but do not want to force the agent to do so. To enter **requested** the **Reason Codes** and **EAS** on the **System-Parameters Customer-Options** screen must be **y**.
forced	Enter **forced** to force an agent to enter a Reason Code when logging out. Enter **forced** to force an agent to enter a Reason Code when entering AUX mode. To enter **forced**, the **Reason Codes** and **EAS** on the **System-Parameters Customer-Options** screen must be **y**.

REDIRECTION ON IP CONNECTIVITY FAILURE

Switch Hook Query Response Timeout

Valid entries	Usage
500 to **5000** (msec)	Assign the time on a system basis that the call processing will wait for a response to the switch hook query before Return on IP Connectivity Failure (ROIF) is triggered. For details on selecting an appropriate timeout period, see *Call Center Software Automatic Call Distribution (ACD) Guide*, 07-300185.
blank	ROIF is not active.

Auto-answer IP Failure AUX Reason Code

Valid entries	Usage
0 to **9**	Enter the reason code assigned for auto-answer IP failure, as the reason the agent was put into Aux Work.

MAXIMUM AGENT OCCUPANCY PARAMETERS

The Maximum Agent Occupancy (MAO) threshold is a system-administered value that is applied across all administered agents and is based on the total percentage of agent time in call service. MAO data is derived from the same calculations that are used to derive Least Occupied Agent (LOA).

When an agent who exceeds the specified MAO threshold attempts to become available, he or she is automatically placed in AUX mode for the reason code administered for this purpose. When the occupancy for such pending agents drops below the MAO, they are released from AUX mode and made available. To use MAO, Expert Agent Selection (EAS) must be enabled. For more information on MAO, see the *Avaya Communication Manager Call Center Software Call Vectoring and Expert Agent Selection (EAS) Guide*, 07-300186.

Maximum Agent Occupancy Percentage

Valid entries	Usage
0 to **100**	Enter the percentage for MAO. Default is 100%.

Maximum Agent Occupancy AUX Reason Code

Valid entries	Usage
0 to **9**	Enter a reason code value. Default is **9**. A different reason code can be used for this purpose, but Avaya recommends that you do *not* use reason code 0.

Field descriptions for page 14

Figure 348: Feature-Related System Parameters screen

```
change system-parameters features                        Page 14 of x
                        FEATURE-RELATED SYSTEM PARAMETERS

AUTOMATIC EXCLUSION PARAMETERS

                       Automatic Exclusion by COS? y
                 Automatic Exclusion Coverage/Hold? y
              Automatic Exclusion with Whisper Page? y
                            Recall Rotary Digit: 2
                    Password to Change COR by FAC: *
          Duration of Call Timer Display (seconds): 3
WIRELESS PARAMETERS
  Radio Controllers with Download Server Permission (enter board location)
    1.         2.         3.         4.         5.

IP PARAMETERS
                   Direct IP-IP Audio Connections? n
                              IP Audio Hairpinning? n
RUSSIAN MULTI-FREQUENCY PACKET SIGNALING
                                           Re-try?
     T2 (Backward Signal) Activation Timer (secs):
```

AUTOMATIC EXCLUSION PARAMETERS

Automatic Exclusion by COS

Activates automatic exclusion automatically by class of service when a user goes off-hook on a station with an assigned **Exclusion** button. This works only for stations on the local server running Communication Manager.

Valid entries	Usage
y	Enables automatic exclusion by a class of service.
n	Exclusion operates normally. See Exclusion on Telephone feature Buttons on page 123 for more information.

Automatic Exclusion Coverage/Hold

Appears when the **Automatic Exclusion by COS** field is **y**.

Valid entries	Usage
y	The principal can bridge onto the call by pressing the appropriate bridged appearance button. And, if the coverage point places the exclusion call on hold, the principal can retrieve the call.
n	If a coverage point has answered a call and there is active exclusion on the call, the principal cannot bridge onto the call. And, if the coverage point places the exclusion call on hold, the principal cannot retrieve the call.

Automatic Exclusion with Whisper Page

Appears when the **Automatic Exclusion by COS** field is **y**.

Valid entries	Usage
y	The whisper page goes through to an excluded call.
n	The whisper page is denied when a station attempts to whisper page to a station that is on an excluded call.

Duration of Call Timer Display

Administer a call timer button on the **Station** screen.

Valid entries	Usage
3 to **30**	Enter the length of time (in 3 second increments) that the call information remains on display after the call is terminated.

Password to Change COR by FAC

Appears if, on the **System-Parameters Customer-Options** screen, the **Change COR by FAC** field is **y**. Avaya recommends using this password option.

Valid entries	Usage
4 to 8 digits	Requires the password option.
blank	Disables the password option.

Recall Rotary Digit

This establishes the digit to use for rotary phones to receive recall dial tone. Dialing this digit simulates switch-hook flash so that users of rotary phones can use features such as conference and transfer. The telephone must also be administered to use the recall rotary digit.

Valid entries	Usage
0 to **9**	Enter the digit users can dial to generate recall dial tone. Use a number that is not the first digit in normal dialing patterns.

WIRELESS PARAMETERS

Radio Controllers with Download Server Permission

Enter the necessary characters for the port location of the circuit pack containing the radio controllers with download server permission.

Valid entries	Usage
01 to **44** (DEFINITY R) or **01** to **03** (DEFINITY CSI, SI) or **1** to **64** (S8700 or S8710 IP-Connect / S8700 or S8710 Multi-Connect / S8300 Media Server)	First and second characters are the cabinet number
A to **E**	Third character is the carrier
0 to **20**	Fourth and fifth characters are the slot number

IP PARAMETERS

Direct IP-IP Audio Connections

Allows direct audio connections between IP endpoints.

Valid entries	Usage
y/n	Enter to **y** to save on bandwidth resources and improve sound quality of voice over IP transmissions.

IP Audio Hairpinning

Allows IP endpoints to be connected through the IP circuit pack in the server.

Valid entries	Usage
y/n	Enter **y** to allow IP endpoints to be connected through the IP circuit pack in the media server in IP format, without going through the Avaya DEFINITY TDM bus.

RUSSIAN MULTI-FREQUENCY PACKET SIGNALING

Re-try

The **Re-try** field applies to outgoing Russian MFP trunks. It allows the server running Communication Manager to resend Russian MFP calling party number and dialed number information to the CO. The server resends the information only once over another outgoing trunk port of the same trunk group if Communication Manager receives a message that the information was received incorrectly by the CO. The switch also sends Russian MFP information over another trunk port if Communication Manager does not receive a timely response for the information.

Valid entries	Usage
y/n	Enter **y** to resend address information on outgoing Russian MFP trunks.

T2 (Backward Signal) Activation Timer (secs)

The **T2 (Backward Signal) Activation Timer (secs)** field applies to outgoing Russian MFP trunks. This field sets the number of seconds that Communication Manager waits for confirmation after sending calling party number and dialed number information on outgoing Russian MFP trunks

Valid entries	Usage
5 to **20**	Enter the number of seconds the system waits to receive confirmation after sending the address information on outgoing Russian MFP trunks.

Firmware Station Download

Use this screen to download firmware to multiple stations of the same telephone type, either 2420 or 2410 DCP telephones. Download firmware to as many as 1000 stations per download schedule. You can schedule a specific time for the download, or you can administer the download to run immediately.

Figure 349: Firmware Station Download screen

```
 change firmware station-download

                              FIRMWARE STATION DOWNLOAD

Source File:

Schedule Download? y
     Start Date/Time:/: Stop Date/Time:/:
Continue Daily Until Completed? y
Download Set Type: 2420

Beginning Station: Ending Station:
```

Source File

Valid entries	Usage
up to 32 alphanumeric characters	Display only field. This field displays the name of the file specified on the **TFTP Server Configuration** screen, and which exists in system memory.

Schedule Download

Valid entries	Usage
y/n	Enter **y** to schedule a time for firmware download to multiple DCP stations..

Start Date/Time

This field appears only when the **Schedule Download** field is set to **y**.

Valid entries	Usage
mm, dd, yyyy; hh, mm	Enter the month, day, year, and time at which you want the firmware download to begin.

Stop Date/Time

This field appears only when the **Schedule Download** field is set to **y**.

Valid entries	Usage
mm, dd, yyyy; hh, mm	Enter the month, day, year, and time at which you want the firmware download to end.

Continue Daily Until Completed

Valid entries	Usage
y/n	Enter **y** if you want the system to execute the firmware download each day at the scheduled time until all specified telephones have received the firmware.

Download Set Type

Valid entries	Usage
2410 DCP	Display only. Indicates the set type (2410 or 2420) of DCP telephones to which firmware is to be downloaed.
2420 DCP	

Beginning Station

Valid entries	Usage
up to 8 digits	Enter the first extension number in the range of telephones to which you want to download the firmware. Up to 1000 stations can be included in a scheduled download.

Ending Station

Valid entries	Usage
up to 8 digits	Enter the last extension number in the range of telephones to which you want to download firmware. Up to 1000 stations can be included in a scheduled download.

Group Paging Using Speakerphone

Use this screen to assign digital speakerphones to a paging group. Users can page all the phones in the group simultaneously by dialing the group's extension.

Figure 350: Group Paging Using Speakerphone screen

```
add group-page next                                          Page 1 of 1
                    GROUP PAGING USING SPEAKERPHONE
   Group Number: 1                            Group Extension: 3210
     Group Name: Sales staff                               COR: 5
GROUP MEMBER ASSIGNMENTS
   Ext      Name                              Ext      Name
 1: 2009    B. Smith                       17:
 2: 2010    R. Munoz                       18:
 3: 2011    Y. Lu                          19:
 4: 2012    A. Sullivan                    20:
 5:                                        21:
 6:                                        22:
 7:                                        23:
 8:                                        24:
 9:                                        25:
10:                                        26:
11:                                        27:
12:                                        28:
13:                                        29:
14:                                        30:
15:                                        31:
16:                                        32:
```

COR

Valid entries	Usage
0 to **95**	Enter a class of restriction. In order to page the group, users' class of restriction must give them calling permission for the group's class of restriction.

Ext

Valid entries	Usage
An extension number	Assign a telephone to the group by entering its extension number in this field.

Group Extension

Valid entries	Usage
An extension number	Assign the extension users will dial to page the members of this group.

Group Name

Valid entries	Usage
1 to 27 characters	Enter a name that's informative to users, because it appears on callers' telephone displays when they page the group.

Group Number

This field displays the identifying number the server running Communication Manager assigns to the group when it is created.

Name

When you save your changes, Avaya Communication Manager fills in this display field with the name assigned to each extension on the **Station** screen.

Related topics

See Paging Over Speakerphones on page 498 for complete instructions.

See "Group Paging" in *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205, for more information.

Holiday Table

Use this screen to define individual holidays or holiday ranges for use with Holiday Vectoring. To use and administer Holiday Vectoring, **Vectoring (Holiday)** must be **y** on the **System-Parameters Customer-Options** screen. Contact your Avaya representative for assistance.

Figure 351: Holiday Table screen

```
 change holiday table 1                                      Page   1 of 1
                                  HOLIDAY TABLE

Number:  1                    Name:

______START______       ______END________
Month Day Hour Min      Month Day Hour Min     Description
__    __  __   __       __    __  __   __      ____________________________________
__    __  __   __       __    __  __   __      ____________________________________
__    __  __   __       __    __  __   __      ____________________________________
__    __  __   __       __    __  __   __      ____________________________________
__    __  __   __       __    __  __   __      ____________________________________
__    __  __   __       __    __  __   __      ____________________________________
__    __  __   __       __    __  __   __      ____________________________________
__    __  __   __       __    __  __   __      ____________________________________
```

Description

Valid entries	Usage
Up to 27 characters.	Enter a phrase to describe the holiday.

End Day

Valid entries	Usage
1 to **31**	Enter the ending day of the holiday.

End Hour

Valid entries	Usage
0 to **23**	Enter the ending hour of the holiday using a 24-hour clock.

End Min

Valid entries	Usage
0 to **59**	Enter the ending minute of the holiday.

End Month

Valid entries	Usage
1 to **12**	Enter the ending month of the holiday.

Name

Display-only field identifying the name of the table.

Valid entries	Usage
Up to 27 characters	Enter a description of the holiday table.

Number

Display-only field identifying the holiday table number.

Valid display	Usage
1 to **10**	Holiday table number.

Start Day

Valid entries	Usage
1 to **31**	Enter the starting day of the holiday.

Start Hour

Valid entries	Usage
0 to **23**	Enter the starting hour of the holiday using a 24-hour clock.

Start Min

Valid entries	Usage
0 to **59**	Enter the starting minute of the holiday.

Start Month

Valid entries	Usage
1 to **12**	Enter the starting month of the holiday.

Hospitality

This screen is used to implement the system parameters associated with the hospitality features. To use and administer the Hospitality-related features, **Hospitality** must be **y** on the **System-Parameters Customer-Options** screen. Contact your Avaya representative for assistance.

Field descriptions for page 1

Figure 352: Hospitality screen

```
change system-parameters hospitality                          Page   1 of   3
                              HOSPITALITY

                        Message Waiting Configuration: act-nopms
                Controlled Restrictions Configuration: act-nopms
                Housekeeper Information Configuration: act-nopms
                      Number of Housekeeper ID Digits: 0
                                   PMS Log Endpoint:
                             Journal/Schedule Endpoint:
             Client Room Coverage Path Configuration: act-nopms
              Default Coverage Path for Client Rooms:
             Forward PMS Messages to Intuity Lodging? n

                               PMS LINK PARAMETERS
                                         PMS Endpoint:
                                 PMS Protocol Mode: transparent ASCII mode? n
                   Seconds before PMS Link Idle Timeout: 20
 Milliseconds before PMS Link Acknowledgment Timeout: 500
                         PMS Link Maximum Retransmissions: 3
               PMS Link Maximum Retransmission Requests: 3
                         Take Down Link for Lost Messages? y
```

Client Room Coverage Path Configuration

This indicates whether the server and the Property Management System (PMS) exchange coverage path information for guest stations.

Valid entries	Usage
act-nopms	The message is acknowledged (MESSAGE ACK), but no action is taken.
act-pms	If active (**act-pms**), the server and PMS exchange and accept coverage path information. This field does not apply to normal mode. When upgrading from a release that does not support this feature, the field is set to **act-pms** if the PMS protocol mode is administered for transparent or ASCII mode.

Controlled Restrictions Configuration

This indicates whether controlled restriction information is being exchanged between the server and the PMS.

Valid entries	Usage
act-nopms	The message is acknowledged (MESSAGE ACK), but no action is taken.
act-pms	The server and the PMS exchange and accept controlled restriction information.

Default Coverage Path for Client Rooms

This applies only to stations with a "client room" class of service in the "occupied" mode. This field is used for transparent or ASCII mode. The value in this field is also used during a translation save as the coverage path for each station with "client room" class of service.

Valid entries	Usage
1 to **9999** or blank	Enter the coverage path assigned when the server receives a check-out message for a valid extension or a new check-in.

Forward PMS Message to INTUITY Lodging

This field is used only in ASCII mode.

Valid entries	Usage
y	PMS-to-INTUITY messages are sent through the server.
n	PMS-to-INTUITY messages are sent directly to the Avaya INTUITY Lodging system.

Housekeeper Information Configuration

This indicates whether housekeeper information is being exchanged between the server and the PMS.

Valid entries	Usage
act-nopms	The message is acknowledged (MESSAGE ACK), but no action is taken.
act-pms	If active (**act-pms**), the server and PMS exchange and accept housekeeper information.

Journal/Schedule Endpoint

This is a valid data extension number that is assigned to the data module connected to the Journal/Schedule printer.

Valid entries	Usage
Valid data extension number	Cannot be a VDN extension. This extension can be the same as the PMS/Log printer and both sets of reports can be printed on the same printer. This extension is dialed by the server to send journal information or schedule reports to the printer.
PMS_LOG	Use this value if the printer is connected over a TCP/IP link, and this link is defined as PMS_LOG on the **IP Services** screen.
PMS_JOURNAL	Use this value if the printer is connected over a TCP/IP link, and this link is defined as PMS_JOURNAL on the **IP Services** screen.
blank	

Message Waiting Configuration

This indicates whether message waiting notification requests and changes are being exchanged between the server and the PMS.

Valid entries	Usage
act-nopms	The message is acknowledged (MESSAGE ACK), but no action is taken.
act-pms	Message waiting is active on the server and information between the PMS and server is being transmitted.

Number of Housekeeper ID Digits

Valid entries	Usage
0 to **6**	Enter the number of digits that the housekeeper must dial for identification.

PMS Log Endpoint

This is a valid data extension number that is assigned to the data module connected to the PMS/Log printer.

Valid entries	Usage
Valid data extension	Cannot be a VDN extension. This extension is dialed by the server to send housekeeping and PMS events to the printer.
PMS_LOG	Use this value if the printer is connected over a TCP/IP link, and this link is defined as PMS_LOG on the **IP Services** screen.
PMS_JOURNAL	Use this value if the printer is connected over a TCP/IP link, and this link is defined as PMS_JOURNAL on the **IP Services** screen.
blank	

PMS LINK PARAMETERS

ASCII mode

The **PMS Protocol Mode** field must be **transparent**.

Valid entries	Usage
y/n	Enter **y** when the ASCII-only mode is being used for the PMS message set.

Milliseconds Before PMS Link Acknowledgment Timeout

This regulates how quickly the system responds to a message from the PMS (also known as "pace timing.") This value is also used as the "inquiry message" (ENQ) time-out value. In most cases, keep this as short as possible.

Valid entries	Usage
100 to **1500**	Enter the time in milliseconds the system waits for an acknowledgment from the PMS indicating it correctly received a message.

PMS Link Maximum Retransmission Requests

Valid entries	Usage
1 to **5**	Enter the number of times that the server will allow the PMS to request acknowledgment for a message that it sent.

PMS Link Maximum Retransmissions

Valid entries	Usage
1 to **5**	Enter the number of times that the server retransmits a message to the PMS in response to a negative acknowledgment, or sends an inquiry for acknowledgment from the PMS before giving up on the message.

PMS Log Endpoint

Valid entries	Usage
Valid extension	Enter the data extension number the server dials to access PMS. Cannot be a VDN extension.
PMS	Use this value if the PMS is connected over a TCP/IP link, and this link is defined as PMS on the **IP Services** screen.
blank	

PMS Protocol Mode

This indicates the message protocol mode used between the server and PMS. Coordinate this option with your PMS vendor.

Valid entries	Usage
normal	
transparent	

Seconds Before PMS Link Idle Timeout

Valid entries	Usage
5 to **20**	Enter the idle time in seconds that the server waits for an acknowledgment from the PMS before the server enters link failure mode from the PMS transmission link.

Take Down Link for Lost Messages

Valid entries	Usage
y/n	Enter **y** to cause the PMS link to come down if messages are being lost. Monitor your PMS error log if you use n.

Field descriptions for page 2

Figure 353: Hospitality screen

```
change system-parameters hospitality                          Page   2 of 3

                                    HOSPITALITY
          Dual Wakeup? y    Daily Wakeup? y  VIP Wakeup? y
                              VIP Wakeup Per 5 Minutes: _____
                        Room Activated Wakeup With Tones?
                  Time of Scheduled Wakeup Activity Report: _______
                   Time of Scheduled Wakeup Summary Report: _______
        Time of Scheduled Emergency Access Summary Report: _______
                                      Announcement Type:
                     Integrated Announcement Extension:
      Length of Time To Remain Connected To Announcement: ____
         Extension To Receive Failed Wakeup LWC Messages: _______
        Routing Extension On Unavailable Voice Synthesis: _______
                Display Room Information in Call Display?
                      Automatic Selection of DID Numbers?
                    Custom Selection of VIP DID Numbers?
                               Number of Digits from PMS:
                                     PMS Sends Prefix?
                  Number of Digits in PMS Coverage Path:
                               Digit to Insert/Delete:
```

Announcement Ports

This field appears only when the **Announcement Type** field is **voice-synthesis**. For the **voice-synthesis** announcement type, this indicates the equipment location of two ports on the voice synthesizer circuit pack. Enter the necessary characters.

Valid entries	Usage
01 to **44** (DEFINITY R) or **01** to **03** (DEFINITY CSI, SI) or **1** to **64** (S8700 or S8710 IP-Connect / S8700 or S8710 Multi-Connect / S8300 Media Server)	First and second characters are the cabinet number
A to **E**	Third character is the carrier
0 to **20**	Fourth and fifth character are the slot number
01 to **04** (Analog TIE trunks) **01** to **31**	Six and seventh characters are the circuit number
1 to **80** (DEFINITY R, CSI) or **1** to **10** (DEFINITY SI) or **1** to **250** (S8700 or S8710 IP-Connect / S8700 or S8710 Multi-Connect / S8300 Media Server)	Gateway
V1 to **V9**	Module
01 to **31**	Circuit

Announcement Type

This indicates the type of automatic wakeup announcement the hotel guest will receive. Allowable entries are as follows:

Valid entries	Usage
external	Applicable when using an announcement adjunct. If **external** is used, complete the **Auxiliary Board for Announcement** field.
integrated	Applicable when using the TN750B or TN750C announcement circuit pack. If **integrated** is used, complete the **Integrated Announcement Extension** field. The extension you enter must be a valid integrated announcement extension (administered on the **Recorded Announcements** screen) or a VDN. If you enter an invalid extension, the server displays an error message.
mult-integ	Multi-integrated; applicable when using the TN750B or TN750C announcement circuit pack. **mult-integ** allows the automatic wakeup feature to use integrated announcement circuit packs to play any one of multiple announcements to different extensions duiring a wakeup call. If **mult-integ** is used, complete the **Default Announcement Extension** field. The extension you enter must be a valid integrated announcement extension (administered on the **Recorded Announcements** screen) or a VDN. If you enter an invalid extension, the server displays an error message.
voice-synthesis	If **voice-synthesis** is used, complete the **Announcement Ports** field.
music-on-hold	If **music-on-hold** is used, no other field appears.
silence	If **silence** is used, no other field appears.

Automatic Selection of DID Numbers

This field assigns a 2 to 5-digit number to a guest's telephone number that is not associated with the room number.

Valid entries	Usage
y/n	Enter **y** to use the Automatic Selection of DID Numbers for Guest Rooms feature.

Auxiliary Board for Announcement

This field appears only when the **Announcement Type** field is **external**. This indicates the equipment location of an auxiliary trunk circuit that connects to the external announcement equipment. Enter the necessary characters.

Valid entries	Usage
01 to **44** (DEFINITY R) or **01** to **03** (DEFINITY CSI, SI) or **1** to **64** (S8700 or S8710 IP-Connect / S8700 or S8710 Multi-Connect / S8300 Media Server)	First and second characters are the cabinet number
A to **E**	Third character is the carrier
0 to **20**	Fourth and fifth character are the slot number
01 to **04** (Analog TIE trunks) **01** to **31** **1** to **80** (DEFINITY R, CSI) or **1** to **10** (DEFINITY SI) or **1** to **250** (S8700 or S8710 IP-Connect / S8700 or S8710 Multi-Connect / S8300 Media Server)	Six and seventh characters are the circuit number Gateway
V1 to **V9**	Module
01 to **31**	Circuit

Custom Selection of VIP DID Numbers

This field allows you to select the DID number assigned to a room when a guest checks in. This field can be accessed only if the **Automatic Selection of DID Numbers** field is **y**.

Valid entries	Usage
y/n	Enter **y** to allow you to select the DID number assigned to a room when a guest checks in.

Daily Wakeup

Valid entries	Usage
y/n	Enter **y** if each extension can request daily wakeup calls.

Default Announcement Extension

This field appears only when the **Announcement Type** field is **mult-integ**. This indicates the default wakeup announcement extension when using the integrated announcement circuit pack.

Valid entries	Usage
valid extension or VDN	Enter the extension of the announcement you want to use for default wakeup calls.

Digit to Insert/Delete

Enter the leading digit that may be deleted and inserted back as described in the following text. The current PMS message set uses the extension number as the room identifier. In many customer configurations, the leading digit of the extension number is dropped to screen the room number. In order to accommodate PMS devices that are based on room number and not extension, this leading digit may be deleted on messages from Avaya Communication Manager to the PMS, and then inserted back on messages from the PMS to Communication Manager.

Note:

The PMS interface supports 3-, 4-, or 5-digit extensions, but prefixed extensions do not send the entire number across the interface. Only the assigned extension number is sent. Therefore, you should not use prefixed extensions for numbers that are also going to use the Digit to Insert/Delete function.

Valid entries	Usage
0 to **9**	Enter the leading digit that may be deleted and inserted back as described in the following text.

Display Room Information in Call Display

This indicates the type of guest room information displayed on telephone displays.

Valid entries	Usage
y	If this field is set to **y**, the phones will display the name and room number. The extension number and room number are not always the same number.
n	If this field is set to **n**, the phones will display the name and extension number.

Dual Wakeup

Valid entries	Usage
y/n	Enter **y** if each extension can request two wakeup calls within one 24-hour time period.

Extension to Receive Failed Wakeup LWC Messages

This indicates where unsuccessful wakeup LWC messages will be stored. This is usually administered to an unassigned extension (cannot be a VDN extension) or to the attendant (attd). In addition, a LWC lamp for that extension is usually assigned to the attendant console as an indication of failed wakeup calls.

Valid entries	Usage
Unassigned extension	Enter the extension where unsuccessful wakeup LWC messages will be stored.

Integrated Announcement Extension

This field appears only when the **Announcement Type** field is **integrated**. This indicates the wakeup announcement extension when using the integrated announcement circuit pack.

Valid entries	Usage
Valid extension or VDN	Enter the extension of the announcement you want to use for wakeup calls.

Length of Time to Remain Connected to Announcement

This applies only after the guest has heard the announcement completely one time, but continues to listen.

Valid entries	Usage
0 to **300**	Enter the length of time in seconds that a hotel guest will be connected to an announcement.

Number of Digits from PMS

This indicates the number of digits being sent from the PMS to the server to identify room numbers.

Note:

If the **Number of Digits from PMS** field is blank and the **PMS Sends Prefix** field is set to **n**, the server will not support an extension that starts with **0**.

Valid entries	Usage
1 to **4**	When using normal mode, digits **1** through **4** are valid.
1 to **5**	When using transparent or ASCII mode, digits **1** through **5** are valid.
blank	If using mixed numbering in the server, leave this field blank.

Number of Digits in PMS Coverage Path

This indicates whether the coverage paths are **3** or **4** digits long. There can be up to 9999 coverage paths.

Valid entries	Usage
3 to **4**	Indicate whether the coverage paths are 3 or 4 digits long.

PMS Sends Prefix

This indicates if the PMS sends a prefix digit to the server as part of the room numbering plan.

Note:

If the **PMS Sends Prefix** field is set to **n** and the **Number of Digits from PMS** field is blank, the server will not support an extension that starts with **0**.

Valid entries	Usage
y/n	Enter **y** or **n** to indicate if the PMS sends a prefix digit to the server as part of the room numbering plan.

Room Activated Wakeup with Tones

Valid entries	Usage
y/n	Enter **y** if wakeup calls can be activated via tones that prompt users for the time they wish to waken. This allows room activated wakeup calls without the use of a speech synthesizer or a display telephone.

Routing Extension on Unavailable Voice Synthesis

Valid entries	Usage
Assigned extension (cannot be a VDN extension) or **attd**	A call will be placed to this extension if a voice systhesis port is not available during voice synthesis entry of wakeup requests.

CAUTION:

Set the following reports for a time other than when the system does its scheduled maintenance tests. To make sure the times do not overlap, enter the command display system-parameters maintenance and check when the system is set to run tests.

Time of Scheduled Emergency Access Summary Report

This indicates the time of day that the Emergency Access Summary Report will be printed on the Journal/ Schedule printer.

Valid entries	Usage
hh:mm:am/pm	Enter the time where hh=hour, mm=minute, am/pm=A.M. or P.M.

Time of Scheduled Wakeup Activity Report

This indicates the time of day that the Wakeup Activity Report will be printed on the Journal/ Schedule Printer. This report summarizes the wakeup activity for each extension that had wakeup activity for the past 24 hours.

Valid entries	Usage
hh:mm:am/pm	Enter the time where hh=hour, mm=minute, am/pm=A.M. or P.M.

Time of Scheduled Wakeup Summary Report

This indicates the time of day that the Wakeup Summary Report will be printed on the Journal/Schedule printer. This report gives an hour-by-hour summary of the number of scheduled wakeup calls and a list of extensions to which wakeup calls were attempted but did not complete during the hour.

Valid entries	Usage
hh:mm:am/pm	Enter the time where hh=hour, mm=minute, am/pm=A.M. or P.M.

VIP Wakeup

Valid entries	Usage
y/n	Enter **y** if each extension can request VIP wakeup calls.

VIP Wakeups Per 5 Minutes

This field appears if the **VIP Wakeup** field is **y**.

Valid entries	Usage
1 to **50**	Enter the number of VIP Wakeup calls allowed in a 5-minute interval.

Field descriptions for page 3

Figure 354: Hospitality screen

```
change system-parameters hospitality                          Page 3 of 3

                                   HOSPITALITY

ROOM STATES:

                  Definition for Rooms in State 1: Rooms in State 1
                  Definition for Rooms in State 2: Rooms in State 2
                  Definition for Rooms in State 3: Rooms in State 3
                  Definition for Rooms in State 4: Rooms in State 4
                  Definition for Rooms in State 5: Rooms in State 5
                  Definition for Rooms in State 6: Rooms in State 6

HOSPITALITY FEATURES

                                              Suite Check-in? n
```

ROOM STATES

Definition for Rooms in State 1 through 6

Enter up to a 30-character definition for each room status. For example, you could define state 1 as ‘clean, ready to use’ and state 2 as ‘occupied, needs cleaning.’

The definitions for room states (Field descriptions for page 3), are for Attendant Room Status only. If you are not using Attendant Room Status, you do not need to complete these fields.

HOSPITALITY FEATURES

Suite Check-in

This field allows attendants to have the system automatically check-in several related extensions with one check-in command.

Valid entries	Usage
y/n	Enter **y** to use the Suite Check-in feature. See "Hospitality" in *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205, for more information.

Hunt Group

Hunt groups allows calls to be answered by users (agents) at a predefined group of telephones or devices.

Use the **Hunt Group** screen to create a hunt group, identified by a hunt group number, and to assign hunt group member users by their extension numbers.This screen can also be used to implement associated features such as Automatic Call Distribution (ACD) and Hunt Group Queuing. The total number of pages can vary, depending on system configuration. See the *Hardware Guide for Avaya Communication Manager* for the maximum number of hunt groups supported by each configuration.

When a call comes into a hunt group, the system checks for the busy or idle status of extension numbers in the hunt group when answering. A Uniform Call Distribution (UCD) type hunt group selects the "most idle" extension in the group when answering a new call. A Direct Department Calling (DDC) type hunt group selects the first available extension (in the administered sequence) when answering a new call. Expert Agent Distribution (EAD), used only with Expert Agent Selection (EAS), selects the "most idle" agent or the "least occupied" agent with the highest skill level for the call’s skill.

Note:

Vector controlled splits/skills can be called directly via the split/skill extension (instead of calling a VDN mapped to a vector that will terminate the call to a vector controlled split/skill); however, the calls will not receive any announcements, be forwarded, redirect to coverage, or intraflow/interflow to another hunt group.

The **Hunt Group** screen can vary according to system configuration and values populating particular fields.The following figures show several ways that page 1 of the **Hunt Group** screen might appear. The descriptions that follow the figures include all fields shown in all variations of Page 1.

Field descriptions for page 1

Figure 355: Hunt Group screen when Queue is y

```
change hunt-group 4                                          Page 1 of X
                                    HUNT GROUP

           Group Number: 4__                                      ACD? _
             Group Name: _____________________________          Queue? y
                                                          Queue Limit:____
        Group Extension: _________                             Vector? _
             Group Type: ________                    Coverage Path: _____
                     TN: ___                 Night Service Destination:
____
                    COR: _                              MM Early Answer? _
          Security Code: _____
       ISDN Caller Disp: _________


Calls Warning Threshold: ___   Port: x____  Extension: _____
 Time Warning Threshold: ___   Port: x____  Extension: _____
```

Figure 356: Hunt Group screen when Queue and Vector are n

```
change hunt-group x                                          Page 1 of X
                                    HUNT GROUP

            Group Number: ___                                     ACD? n
              Group Name: _____________________________           Queue? n
         Group Extension: ____                                   Vector? n
              Group Type: _________                   Coverage Path: _____
                      TN: ___                 Night Service Destination:
_____
                     COR: _                             MM Early Answer? _
           Security Code: _____
     ISDN Caller Display: _________
```

The two **Extension** fields display only when the **Calls Warning Port** and the **Time Warning Port** fields are **x**.

Figure 357: Hunt Group screen when Queue and Vector are y

```
change hunt-group x                                          Page 1 of X
                                    HUNT GROUP

           Group Number: ___                                    ACD? n
             Group Name: _____________________________          Queue? n
        Group Extension: _____                                  Vector? y
             Group Type: _______
                     TN: ___
                    COR: _                              MM Early Answer?
          Security Code: ____
    ISDN Caller Display: _________


 Calls Warning Threshold: ___   Port: x___   Extension: ____
  Time Warning Threshold: ___   Port: x___   Extension: ____
```

The two **Extension** fields display only when the **Calls Warning Port** and the **Time Warning Port** fields are x.

Figure 358: Hunt Group screen when Queue is y and Vector is n

```
change hunt-group x                                          Page 1 of X
                                    HUNT GROUP

           Group Number: ___                                    ACD? n
             Group Name: _____________________________          Queue? n
        Group Extension: _____                                  Vector? n
             Group Type:                                Coverage Path: ____
                     TN:                       Night Service Destination:
_____
                    COR:                                MM Early Answer?
          Security Code: ____
       ISDN Caller Disp: _________


Calls Warning Threshold: ___   Port: x___   Extension: ____
 Time Warning Threshold: ___   Port: x___   Extension: ____
```

ACD

Indicates whether Automatic Call distribution is used. This field cannot be set to **y** if, on the **System-Parameters Customer-Options** screen, the **ACD** field is **n**.

Valid entries	Usage
y	The hunt group will function as an ACD split/skill. AUDIX hunt groups can function as ACD splits/skills.
n	This feature is not desired, even if, on the **System-Parameters Customer-Options** screen, the **ACD** field is **y**. When the hunt group is assigned as an ACD split/skill, the hunt group members serve as ACD agents. The agents in this split/skill must log in to receive ACD split/skill calls. If this hunt group is on a remote server running Communication Manager and using the AUDIX in a DCS feature, then enter **n**.

Calls Warning Threshold

Appears if the **Queue** field is **y**. Enter the number of calls that can be queued before the System flashes the queue status (feature buttons assigned on agents phones) and the optional Auxiliary Queue Call Warning Threshold lamp assigned to the split/skill. These lamps are lighted steadily when at least one call is in queue and the threshold has not yet been reached

Valid entries	Usage
1 to **999** and must be less than or equal to the queue length or blank	This field must not be left blank if **Calls Warning Port** is assigned a port number.

(Calls Warning) Extension

Appears if the **Queue** field is **y** and when the **Calls Warning Port** and the **Time Warning Port** fields are **x**. An extension is needed when an X is placed in **Calls Warning Port**. This extension can be used by the Terminal Translation Initialization (TTI) feature to assign a port to this extension from the port itself. Once **Calls Warning Port** is assigned a valid port (either via TTI or the **change hunt-group** command), then the extension is removed and considered unassigned.

Valid entries	Usage
Extension	Enter an unassigned extension. This field may not be blank.

(Calls Warning) Port

Appears if the **Queue** field is **y**. Enter the seven-character port number assigned to connect the optional external Auxiliary Queue Call Warning Threshold lamp that will flash when the number of calls in queue has exceeded the queue warning threshold (assigned in **Calls Warning Threshold**).

This port is assigned to an Analog Line circuit pack or given an **x** designation if an extension is used. Enter the necessary characters.

Valid entries	Usage
01 to **44** (DEFINITY R) or **01** to **03** (DEFINITY CSI, SI) or **1** to **64** (S8700 or S8710 IP-Connect / S8700 or S8710 Multi-Connect / S8300 Media Server)	First and second characters are the cabinet number
A to **E**	Third character is the carrier
0 to **20**	Fourth and fifth character are the slot number
01 to **04** (Analog TIE trunks) **01** to **31**	Six and seventh characters are the circuit number

Note:

For example, 01A0612 is in cabinet 01, carrier A, slot 06, and circuit number (port) 12.

COR

Enter the class of restriction (COR) number that reflects the desired restriction for the hunt group. If this is a hunt group supporting the AUDIX in a DCS feature, the CORs on the **Hunt Group** screen on each server running Communication Manager must be the same.

Valid entries	Usage
0 to **95**	Enter the class of restriction (COR) number that reflects the desired restriction for the hunt group.

Coverage Path

Enter a coverage path number. This assigns a coverage path for the hunt group. The coverage path is assigned using the **Coverage Path** screen. Does not appear if the **Vector** field is **y**.

Valid entries	Usage
1 to **999**	Enter a coverage path number.
t1 to **t999**	Time of day table
blank	

Group Extension

Enter an unused extension number to be assigned to the hunt group. The field may not be blank.

Valid entries	Usage
1 to 7 digits	Unassigned extension

Group Name

Enter a character string that uniquely identifies the group (for example, "parts dept," "purchasing," or "sales dept").

Valid entries	Usage
27-character string	

Note:

For 4620SW or 4610SW telephones, the **Group Name** field is supported by Unicode language display. For more information on Unicode language display, see Administering Unicode display on page 181.

Group Number

This is a display-only field when the screen is accessed using an administration command such as **add** or **change**.

Group Type

The group types available depend on what is administered on your **System-Parameters Customer-Options** screen for **Automatic Call Distribution (ACD)**, **Expert Agent Selection (EAS)** and **CentreVu Advocate**. The table below shows what group types are available depending on your configuration.

Each option uses a different method to select an extension or agent for a call when two or more extensions or agents are available. The second table shows how calls are handled for each group type.

	circ	ddc	ucd-mia	ead-mia	ucd-loa	ead-loa	pad	slm
ACD=n	x	x						
ACD, Split, Vector = n/y		x	x					
ACD, Skill, Vector = n/y			x	x				x
ACD, Skill, Vector = y Advocate or Elite			x	x	x	x		
ACD, Skill, Vector = y Dynamic Advocate			x	x	x	x	x	

Valid entries	Usage
circ	Enter circ (circular) when the call should be routed in a "round-robin" order. The order in which you administer the extensions determines the order that calls are directed. The server running Communication Manager keeps track of the last extension in the hunt group to which a call was connected. The next call to the hunt group is offered to the next extension in the circular list, independent of how long that extension has been idle. You cannot use circular hunting with automatic call distribution, queues, or vectors.
ddc	Enter ddc when the call should be routed to the first extension or ACD agent assigned in the ACD split. Group type ddc is also known as "hot seat" distribution. "ddc" distribution is not available when the group is administered as a skill.

1 of 2

Valid entries	Usage
ucd-mia **ucd-loa**	When ucd-mia or ucd-loa is entered, a call routes to the most-idle agent based on when the agent finished the most recent call ("ucd-mia"), or the least occupied agent based on agent occupancy ("ucd-loa"). Enter ucd-mia or ucd-loa if the hunt group has an AUDIX message. One of these entries is required when supporting the Outbound Call Management feature and when the **Controlling Adjunct** field is **asai**.
ead-mia **ead-loa**	When ead-mia or ead-loa is entered, a call routes to the available agent with the highest skill level for the call. If two or more agents with equal skill levels are available, Communication Manager routes the call to the most-idle agent based on when the agent finished the most recent call ("ead-mia"), or the least occupied agent based on agent occupancy ("ead-loa"). This allows a call to be distributed to an agent best able to handle it if multiple agents are available.
pad	Enter pad (percent allocation distribution) to select an agent from a group of available agents based on a comparison of the agent's work time in the skill and the agent's target allocation for the skill.
slm	Enter slm when you want to: • Compare the current service level for each SLM-administered skill to a user-defined call service level target and identify the skills that are most in need of agent resources to meet their target service level. • Identify available agents and assess their overall opportunity cost, and select only those agents whose other skills have the least need for their service at the current time.

2 of 2

ISDN Caller Disp

This field is required if, on the **System-Parameters Customer-Options** screen, the **ISDN-PRI** or **ISDN-BRI Trunks** field is **y**.

Valid entries	Usage
grp-name	Enter **grp-name** or **mbr-name** to specify whether the hunt group name or member name, respectively, will be sent to the originating user.
mbr-name	
blank	If the **ISDN-PRI** or the **ISDN-BRI Trunks** field is **n**, this field must be blank.

MM Early Answer

This field applies for systems using Multimedia Call Handling only.

Valid entries	Usage
y/n	The system begins to answer an H.320 call and establish an audio channel before offering the conversion call to the hunt group. This starts billing for the call when the call is first put into queue.

Night Service Destination

Enter the destination where calls to this split will redirect when the split is in the night service mode. Not all features will work correctly if this is not a local extension. Does not appear if the **Vector** field is **y**.

Valid entries	Usage
An assigned extension number (can be a VDN extension)	
attd	An attendant group code.
blank	

Queue

Specifies a queue for the hunt group.

Valid entries	Usage
y/n	Enter **y** so the hunt group will be served by a queue.

Queue Limit

This field appears when the **Queue** field is set to **y**.

Valid entries	Usage
1 to **999**	Enter a limit to the number of calls that will queue.
unlimited	The system dynamically allocates the queue slots from a common pool on an as-needed basis.

Security Code

Enter a security code (password) used for the Demand Print feature.

Valid entries	Usage
3 to 8 digits	Enter the password for the Demand Print feature.

Time Warning Threshold

Appears if the **Queue** field is **y** and when the **Calling Warning Port** and the **Time Warning Port** fields are **x**. Enter the time in seconds that a call can remain in the queue before the system flashes the Queue status lamps (feature buttons assigned members phones) and the Auxiliary Queue Time Warning lamp assigned to this split/skill.

Valid entries	Usage
0 to **999** or blank	An entry of **0** provides a warning whenever a call is queued.

(Time Warning) Extension

Appears if the **Queue** field is **y**. An extension is needed when an **x** is placed in **Time Warning Port**. This extension can be used by the Terminal Translation Initialization (TTI) feature to assign a port to this extension from the port itself. Once **Time Warning Port** is assigned a valid port (either via TTI or the **change hunt-group** command), then the extension is removed and considered unassigned.

Valid entries	Usage
Extension	Enter an unassigned extension. This field may not be blank.

(Time Warning) Port

Appears if the **Queue** field is **y**. Enter the seven-character port number assigned to the Auxiliary Queue Time Warning lamp that flashes when the time entered in **Time Warning Threshold** has been reached by a call in queue.

Note:

This port is assigned to an Analog Line circuit pack or given an **X** designation if an extension is used. Enter the necessary characters.

Valid entries	Usage
01 to **44** (DEFINITY R) or **01** to **03** (DEFINITY CSI, SI) or **1** to **64** (S8700 or S8710 IP-Connect / S8700 or S8710 Multi-Connect / S8300 Media Server)	First and second characters are the cabinet number
A to **E**	Third character is the carrier
0 to **20**	Fourth and fifth character are the slot number
01 to **04** (Analog TIE trunks) **01** to **31**	Six and seventh characters are the circuit number

For example, 01A0612 is in cabinet 01, carrier A, slot 06, and circuit number (port) 12.

TN

Valid entries	Usage
1 to **100**	Enter the Tenant Partition number.

Vector

See example screens for fields that display when this field is **y**.

Valid entries	Usage
y/n	Enter **y** to indicate that this hunt group will be vector controlled. On the **System-Parameters Customer-Options** screen, the **Vectoring-Basic** field must be **y** before **y** can be entered here.

Field descriptions for page 2

Page 2 of the **Hunt Group** screen can vary according to values for particular fields on Page 1. The screen shown in Figure 359 appears only when the **ACD** field on page 1 is **y**, and the **Queue** and **Vector** fields are **n**. If the **ACD** field is **n**, the **Messenge Center** page shown in Figure 361 becomes Page 2 and all subsequent page numbers are decreased by one. The **Timed ACW Interval** field shown below appears only if, on the **System-Parameters Customer-Options** screen, the **Timed ACW** field is **y**. The screen shown in Figure 360 appears when the **Queue** and **Vector** fields are **y**.

The descriptions that follow the figures include all fields in both variations of Page 2.

Figure 359: Hunt Group screen when ACD is y and Queue and Vector are n

```
Change hunt-group 4                                              Page 2 of X
                                HUNT GROUP
                  Skill? _           Expected Call Handling Time (sec): ___
                    AAS? _             Service Level Target (% in sec):80 in 20
              Measured: ____
   Supervisor Extension: ____
  Priority on Intraflow? _

    Controlling Adjunct: ____




Timed ACW Interval (sec): ___              Maximum Auto Reserve Agents: 0
  Multiple Call Handling: _____________
                                       Redirect on No Answer (rings): ___
                                                  Redirect to VDN: _____
                        Forced Entry of Stroke Counts or Call Work Codes? _
```

Figure 360: Hunt Group screen when Queue and Vector are y

```
change hunt-group x                                              Page 2 of X
                                HUNT GROUP
                  Skill?           Expected Call Handling Time (sec):___
                    AAS? _           Service Level Target (% in sec):80 in 20
              Measured: internal
    Supervisor Extension: ____



     Controlling Adjunct: ____



      VuStats Objective: ____
Timed ACW Interval (sec): ___ ___              Maximum Auto Reserve Agents: 0
  Multiple Call Handling: _____________
                                       Redirect on No Answer (rings): ___
                                                  Redirect to VDN: _____
                      Forced Entry of Stroke Counts or Call Work Codes? _
```

AAS

Appears when the **ACD** field is **y**

Valid entries	Usage
y/n	Enter **y** if this hunt group is to serve as an Auto-Available Split.

Acceptable Service Level (sec)

Appears if the **ACD** field is **y** and the **Measured** field is **internal** or **both**. This allows BCMS and/or VuStats to report a percentage of calls that were answered within the specified time. This entry is also used by the CentreVu Advocate Service Objective feature.

Valid entries	Usage
0 to **9999** seconds	Enter the number of seconds within which calls to this hunt group should be answered.

Adjunct CTI Link

Appears when the **ACD** field is **y** and the **Controlling Adjunct** field is **asai** or **adjlk**. Enter the appropriate ASAI CTI Link. This field cannot be blank.

Controlling Adjunct

Appears only if the **ACD** field is **y**. If the controlling adjunct is a CONVERSANT voice system (requires an ASAI link), then enter **asai** in this field. (On the **System-Parameters Customer-Options** screen, the **ASAI Link Core Capabilities** and **Computer Telephony Adjunct Links** fields must be **y** for CallVisor ASAI capability and for an entry other than **none**.)

Valid entries	Usage
none	Indicates that members of the split/skill or hunt group are not controlled by an adjunct processor.
asai	All agent logins are controlled by an associated adjunct and logged-in agents can only use their data terminal keyboards to perform telephone functions (for example, change work state).

1 of 2

Valid entries	Usage
adjlk	Computer Telephony Adjunct Links
asai-ip	Indicates ASAI links administered without hardware and used by the Co-Resident DLG application. You cannot use this selection unless, on the **System-Parameters Customer-Options** screen, the **Co-Res DEFINITY LAN Gateway** field is **y**.
adj-ip	Indicates ASAI adjunct links administered without hardware and used by the Co-Resident DLG application. You cannot use this selection unless, on the **System-Parameters Customer-Options** screen, the **Co-Res DEFINITY LAN Gateway** field is **y**.
	2 of 2

Dynamic Percentage Adjustment

Appears when **ACD** and **Group Type** fields on the **Hunt Group** screen are **pad** and the **Business Advocate** field is **y** on the **System-Parameters Customer-Options** screen.

Valid entries	Usage
y/n	Enter **y** to enable automatic adjustments to agents' target allocations as needed to help meet the administered service level targets.

Dynamic Queue Position

Appears when the **ACD** and **Skill** fields are **y** on the **Hunt Group** screen and the **Business Advocate** field is **y** on the **Feature-Related System Parameters** screen.

Valid entries	Usage
y/n	Enter **y** to apply the dynamic queue operation to the calls queued to the skill.

Dynamic Threshold Adjustment

Appears when the **ACD** and **Service Level Supervisor** fields on the **Hunt Group** screen are **y** and the **Business Advocate** field is **y** on the **System-Parameters Customer-Options** screen.

Valid entries	Usage
y/n	Enter **y** to enable automatic adjustments to overload thresholds to engage reserve agents a bit sooner or a bit later to meet the administered service levels.

Expected Call Handling Time (sec)

Appears if, on the **System-Parameters Customer-Options** screen, either the **Vectoring (Advanced Routing)** or **Business Advocate** field is **y**. and, on the **Hunt Group** screen, the **ACD** field is **y**.

Valid entries	Usage
1 to **9999** in increments of 1	Establishes the number of seconds for expected call handling. This value is used to initialize Expected Wait Time and is also used by the CentreVu Advocate Percent Allocation feature.

Forced Entry of Stroke Counts or Call Work Codes

Appears when the **ACD** field is **y** and **Controlling Adjunct** field is **none**.

Valid entries	Usage
y/n	Enter **y** so either a Stroke Count or Call Work Code must be entered for each call answered by an agent when in the Manual-In mode.

Inflow Threshold (sec)

Appears only when the **ACD** and **Queue** fields are **y** and **Vector** field is **n**. Enter the number of seconds that a call can remain in the queue before no more calls will be accepted by the queue. If **0** is entered, a call is redirected to this split/skill only if there is an available agent.

Valid entries	Usage
0 to **999**	Enter the number of seconds that a call can remain in the queue before no more calls will be accepted by the queue.

Level 1 Threshold (sec)

Enter the number of seconds corresponding to the Expected Wait Time (EWT) you want to set for this threshold. For example, if you enter 45 calls whose EWT exceeds 45 seconds will be classified as over threshold 1. This field is used with Service Level Supervisor and only appears if the **Service Level Supervisor** field is **y**.

Level 2 Threshold (sec)

Appears if the **ACD** field is **y**. Enter the number of seconds corresponding to the Expected Wait Time (EWT) you want to set for this threshold. For example, if you enter 60 calls whose EWT exceeds 60 seconds will be classified as over threshold 2. This field is used with Service Level Supervisor and only appears if the **Service Level Supervisor** field is **y**.

Maximum Auto Reserve Agents

Appears only if the **Group Type** field is **slm**. Set the maximum number of Auto Reserve Agents you want to be available for this skill (hunt group). Any time an auto-reserve skill is in danger of falling below its target service level percent, some of this skill's agents are auto-reserved (kept idle in other skills) so they will be available when a new call arrives for this skill. Valid values are 0 to 9. Default is 0.

Measured

Provides measurement data for the ACD split/skill collected (internal to the switch) for **VuStats** or **BCMS**. This measurement data is collected for **VuStats** and **BCMS** only if, on the **System-Parameters Customer-Options** screen, they are **y** and, on the **Hunt Group** screen, the **ACD** field is **y**.

Valid entries	Usage
internal	If you enter **internal** in this field and on the **System-Parameters Customer-Options** screen neither the **VuStats** or **BCMS** field is **y**, the system displays the following message: `<value> cannot be used; assign either BCMS or VuStats first` Contact your Avaya representative to assist with any changes you want to make on the **System-Parameters Customer-Options** screen.
external	Provides measurements made by the Call Management System (external to the server running Communication Manager).
both	Provides measurements collected both internally and externally.
none	Measurement reports for this hunt group are not required.

Multiple Call Handling

Appears only if, on the **System-Parameters Customer-Options** screen, the **Multiple Call Handling** field is **y** and the **ACD** field on this screen is **y**. This field defines whether the hunt group can have multiple call handling capabilities, and if so, what type.

Valid entries	Usage
none	Agents who are members of that split/skill can only receive an ACD call from that split/skill when the telephone is idle.
on-request	Agents in the Multiple Call Handling split/skill can place a non-ACD or an ACD call on hold and select an available work mode. A queued ACD split/skill or direct agent call then is routed to the agent.

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Valid entries	Usage
many-forced	An ACD call is delivered automatically to an idle line appearance if the agent is in the Auto-In/Manual-In (MI/AI) work mode and an unrestricted line appearance is available.
one-forced	An ACD call is delivered automatically to an idle line appearance if the agent has no other ACD call on the station, is in the Auto-In/Manual-In (MI/AI) work mode, and an unrestricted line appearance is available.
one-per-skill	An ACD call is delivered automatically to an idle line appearance if the agent has no other ACD call for that skill on the station, is in the Auto-In/Manual-In (MI/AI) work mode, and un unrestricted line appearance is available. Valid in an EAS environment and only when the **Skill** field is **y**.
	2 of 2

Priority On Intraflow

Appears if the **ACD** field is **y** and the **Vector** field is **n**.

Valid entries	Usage
y/n	Enter **y** for calls intraflowing from this split to a covering split to be given priority over other calls waiting in the covering split queue.

Redirect on No Answer (rings)

Appears if the **ACD** field is **y**.

Valid entries	Usage
1 to **20**	Enter the maximum number of rings before a call will redirect back to the split/skill, or to the administered VDN.
blank	Deactivates Redirect on No Answer.

Redirect to VDN

Appears if the **ACD** field is **y**. To redirect a RONA call to a VDN instead of to the split/skill, enter the extension number of the VDN. The administered VDN must be on-premises and must be administered on the system. The VDN can specify a vector that will in turn route to an off-premises VDN. You cannot enter an extension in this field if the **Redirection on No Answer (rings)** field is blank. Direct Agent calls go to the agent's coverage path if it is administered. If not, the calls go to a VDN.

Valid entries	Usage
Assigned VDN or blank	To redirect a RONA call to a VDN instead of to the split/skill, enter the extension number of the VDN.

Service Level Supervisor

Appears if, on the **System Parameters Customer-Options** screen, the **Business Advocate** field is **y** and, on the **Hunt Group** screen, the **ACD** and **Skill** fields are **y**. For information on CentreVu Advocate, please contact your Avaya Account Executive or see the *CentreVu Advocate User Guide*.

Valid entries	Usage
y/n	Enter **y** to use Service Level Supervisor for this skill.

Service Level Target (% in sec)

Appears when the **ACD** field and the **Dynamic Percentage Adjustment** or **Dynamic Threshold Adjustment** field on the **Hunt Group** screen is **y** and the **Business Advocate** field is **y** on the **System-Parameters Customer-Options** screen. Also appears when the **Group Type** field on the **Hunt Group** screen is **slm**, and on the **System-Parameters Customer-Options** screen, the **Service Level Maximizer** field is set to **y**, and the **Business Advocate** field is set to **n**.

Valid entries	Usage
1 to **99** (percentage) **1** to **9999** (time in seconds)	Enter the percentage and time components of the service level target. After 20 seconds, default service level target values are set at 80%.

Service Objective

Appears when the **Skill** and **Business Advocate** fields on the **Feature Related System Parameters** screen are **y** and, on the **Hunt Group** screen, the **ACD** field is **y**.

Valid entries	Usage
1 to **9999**	Enter the per-skill service objective.

Skill

Appears if, on the **System-Parameters Customer-Options** screen, the **Expert Agent Selection** field is **y** and, on the **Hunt Group** screen, the **ACD** field is **y**.

If this field is **y**, then the **Group Type** field must be **ucd** or **ead**.

Valid entries	Usage
y/n	Enter **y** if this hunt group is to be an EAS skill.

Supervisor Extension

Appears if the **ACD** field is **y**.

Valid entries	Usage
Valid extension	Enter the extension number (cannot be a VDN number) of the ACD split/skill supervisor that agents will reach when using the Assist feature

Timed ACW Interval (sec)

When a value is entered in this field, an agent in auto-in work mode who receives an ACD call from this hunt group is placed automatically into After Call Work (ACW) when the call drops. Enter the number of seconds the agent should remain in ACW following the call. When the administered time is over, the agent automatically becomes available. Timed ACW cannot be administered if the hunt group is adjunct controlled, is an AUDIX Message Center, or is an auto-available split. The **Timed ACW Interval** field appears if, on the **System Parameters Customer-Options** screen, the **Timed ACW** field is **y** and, on the **Hunt Group** screen, the **ACD** field is **y**.

Valid entries	Usage
1 to **9999** or blank	The number of seconds the agent should remain in ACW following the call.

VuStats Objective

Enter a numerical user-defined objective. An objective is a split or skill goal for the call. This could be an agent objective such as a specific number of calls handled or an average talk time. The objective could also be a percent within the service level. The objective appears on the VuStats display and allows agents and supervisors to compare the current performance against the value of the objective for the split or skill.

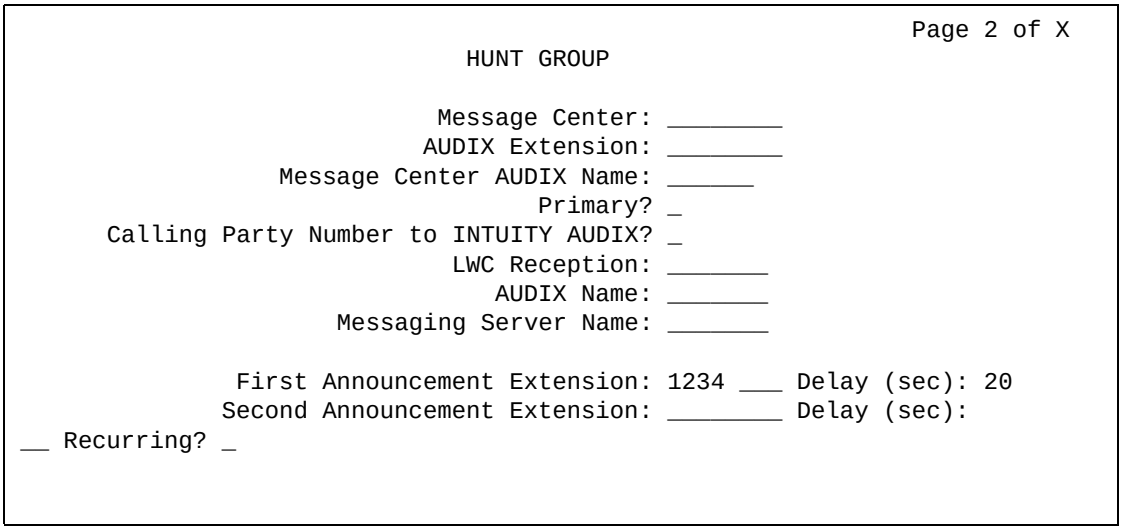
You can use this value in a customized VuStats display format if, on the **VuStats display format** screen, the **Object Type** field is either **agent**, **agent-extension**, or **split**.

Appears if, on the **System-Parameters Customer-Options** screen, the **VuStats** field is **y** and the **Measured** field is either **internal** or **both** and, on the **Hunt Group** screen, the **ACD** field is **y**.

Valid entries	Usage
0 to **99999**	Enter a split or skill objective.

Field descriptions for Message Center page

Figure 361: Hunt Group Message Center screen



AUDIX Name

Enter the name of the AUDIX machine as it appears on the **Node Names** screen. Only appears on DEFINITY Server R. Add the AUDIX name to the **Node Names** screen before entering it in this field. For more information on the **Node Names** screen, see *Administration for Network Connectivity for Avaya Communication Manager*, 555-233-504.

Calling Party Number to INTUITY AUDIX

Appears when the **Message Center** field is **audix** or **rem-vm**.

Valid entries	Usage
y/n	Enter **y** to send the calling party number to INTUITY AUDIX.

First Announcement Delay (sec)

Appears only if the **Queue** field is **y** and the **Vector** field is **n**. Enter the number of seconds that a call remains in queue before the associated first announcement is given the calling party. The call retains its place in the queue while the caller is listening to the recorded announcement. If the call hasn't been answered after the announcement, the caller hears music (for first announcement only) if Music-on-Hold is provided or ringing for as long as the call remains in queue. Appears only if the **Queue** field is **y** and the **Vector** field is **n**.

Valid entries	Usage
0 to **99**	When **0** is entered, the first announcement is provided immediately to the caller. This value is set automatically to **0** if there is no queue.
blank	This field must be blank if there is no first announcement.

First Announcement Extension

Appears when the **ACD** and **Queue** fields are **y** and the **Vector** field is **n**.

Note:

If entering a Multi-Location Dial Plan shortened extension, note the following: When entering a Multi-Location Dial Plan shortened extension in a field designed for announcement extensions, certain administration end validations that are normally performed on announcement extensions are not done, and resultant warnings or submittal denials do not occur. The shortened extensions also do not appear in any display or list that shows announcement extensions. Extra care should be taken to administer the correct type of announcement for the application when assigning shortened extensions.

Valid entries	Usage
Enter a recorded announcement extension number.	This is the announcement the caller receives after being in the queue for the time interval specified in First Announcement Delay. If the call hasn't been answered after the announcement, the caller hears music (only after the first announcement) if Music-on-Hold is provided, or ringing for as long as it remains in the queue. If this is the forced first announcement, the caller always hears ringback after the announcement; otherwise, the caller hears music (if provided).
blank	Leaving this field blank indicates there will be no announcement.

LWC Reception

Defines the destination for Leave Word Calling (LWC) messages left for the hunt group.

Valid entries	Usage
audix	If LWC is attempted, the messages are stored in AUDIX. The **Audix Name** field must be filled in too.
spe	If LWC is attempted, the messages are stored in the system processing element (spe).
none	

Message Center

Enter the type of messaging adjunct for the hunt group. Only one hunt group in the system can be administered as **audix**, one as **qsig-mwi**, one as **fp-mwi**, and one as **rem-vm**.

Valid entries	Usage
msa	Messaging Server Adjunct
msa-vm	For voice-mail system integrated using Mode Codes or Digital Station Emulation
rem-vm	DCS feature allowing voice mail to be located on another server
audix	For AUDIX located on this server running Avaya Communication Manager
qsig-mwi	QSIG network allowing voice mail to be located on another server
fp-mwi	Public network allowing AUDIX to be located on another switch; administrable only when the **ISDN Feature Plus** field on the **System-Parameters Customer-Options** screen is **y**.
none	Indicates the hunt group does not serve as a message hunt group.

Message Center AUDIX Name

Enter the name of the Message Center AUDIX. Appears on Avaya DEFINITY Server R, S8300 Media Server, S8700 or S8710 IP-Connect, and S8700 or S8710 Multi-Connect if the **Message Center** field is **audix** or **rem-vm**.

Message Center MSA Name

Note:

Administer the **IP Node Names** screen first.

Enter the name of the Message Center MSA. When it appears, it replaces the **Message Center AUDIX Name** field. Appears on DEFINITY R, S8300 Media Server, S8700 or S8710 IP-Connect, S8700 or S8710 Multi-Connect if the **Message Center** field is **msa**.

Primary

Appears on Avaya DEFINITY Server R, S8300 Media Server, S8700 or S8710 IP-Connect, S8700 or S8710 Multi-Connect if the **Message Center** field is **audix** or **rem-audix**.

Valid entries	Usage
y/n	Enter **y** to indicate that the specified AUDIX is the primary adjunct.

Second Announcement Extension

Appears only when the **ACD** and **Queue** fields both are **y** and the **Vector** field is **n**.

Note:

If entering a Multi-Location Dial Plan shortened extension, note the following: When entering a Multi-Location Dial Plan shortened extension in a field designed for announcement extensions, certain administration end validations that are normally performed on announcement extensions are not done, and resultant warnings or submittal denials do not occur. The shortened extensions also do not appear in any display or list that shows announcement extensions. Extra care should be taken to administer the correct type of announcement for the application when assigning shortened extensions.

Valid entries	Usage
Valid extension	Enter the extension number assigned to a recorded announcement.
blank	Leaving the field blank indicates there is no second announcement.

Second Announcement Delay (sec)

Appears only when the **ACD** and **Queue** fields both are **y** and the **Vector** field is **n**. Enter the time in seconds before the call in the queue receives a second recorded announcement or that the second announcement is repeated.

Valid entries	Usage
1 to **99**	Avaya recommends that, if this split/skill or hunt group is a coverage point for another split/skill, this delay should not be more than 15 seconds.
blank	Leave blank if there is no second announcement.

Second Announcement Recurring

Appears only when the **ACD** and **Queue** fields both are **y** and the **Vector** field is **n**.

Valid entries	Usage
y	The second announcement can be repeated.
blank	Leave blank if there is no second announcement.

Voice Mail Extension

Appears if the **Message Center** field is **rem-vm**.

Valid entries	Usage
extension	Enter the UDP extension of the voice-mail hunt group on the host server running Communication Manager.

Field descriptions for Hunt group for MWI screen

Figure 362: Hunt Group for MWI screen

```
change hunt-group 4                                                Page 3 of X
                                  HUNT GROUP

                            Message Center: _________
                          Voicemail Number: ____________________
Routing Digits (e.g. AAR/ARS Access Code): ____

                       Send Reroute Request? y

                              LWC Reception: none
                                 AUDIX Name: _______


              First Announcement Extension: ________ Delay (sec): __
             Second Announcement Extension: ________ Delay (sec): __ Recurring? _
```

Calling Party Number to INTUITY AUDIX

Only appears when the **Message Center** field is **audix** or **rem-audix**.

Valid entries	Usage
y/n	Enter **y** to send the calling party number to INTUITY AUDIX.

First Announcement Extension

Appears when the **ACD** and **Queue** fields are **y** and the **Vector** field is **n**.

Note:

If entering a Multi-Location Dial Plan shortened extension, note the following: When entering a Multi-Location Dial Plan shortened extension in a field designed for announcement extensions, certain administration end validations that are normally performed on announcement extensions are not done, and resultant warnings or submittal denials do not occur. The shortened extensions also do not appear in any display or list that shows announcement extensions. Extra care should be taken to administer the correct type of announcement for the application when assigning shortened extensions.

Valid entries	Usage
Enter a recorded announcement extension number.	This is the announcement the caller receives after being in the queue for the time interval specified in **First Announcement Delay**. If the call hasn't been answered after the announcement, the caller hears music (only after the first announcement) if Music-on-Hold is provided, or ringing for as long as it remains in the queue. If this is the forced first announcement, the caller always hears ringback after the announcement; otherwise, the caller hears music (if provided).
blank	Leaving this field blank indicates there will be no announcement.

First Announcement Delay (sec)

Enter the number of seconds that a call remains in queue before the associated first announcement is given the calling party. The call retains its place in the queue while the caller is listening to the recorded announcement. If the call hasn't been answered after the announcement, the caller hears music (for first announcement only) if Music-on-Hold is provided or ringing for as long as the call remains in queue. Appears only if the **Queue** field is **y** and the **Vector** field is **n**.

Valid entries	Usage
0 to **99**	When **0** is entered, the first announcement is provided immediately to the caller. This value is set automatically to **0** if there is no queue.
blank	This field must be blank if there is no first announcement.

LWC Reception

Defines the destination for Leave Word Calling (LWC) messages left for the hunt group.

Valid entries	Usage
audix	If LWC is attempted, the messages are stored in AUDIX. The **Audix Name** field must be filled in, too.
spe	If LWC is attempted, the messages are stored in the system processing element (spe).
none	

Message Center

Enter the type of messaging adjunct for the hunt group. Only one hunt group in the system can be administered as **audix**, one as **qsig-mwi**, one as **fp-mwi**, one as **rem-audix**, and as many as 6 as **qsig-mwi**.

Valid entries	Usage
rem-audix	DCS feature allowing AUDIX to be located on another server
audix	For AUDIX located on this server running Avaya Communication Manager
msa-vm	For voice-mail system integrated using Mode Codes or Digital Station Emulation
qsig-mwi	QSIG network allowing AUDIX, or another QSIG-integrated voice-mail system, to be located on another server/switch
fp-mwi	Public network allowing AUDIX to be located on another switch; administrable only when the **ISDN Feature Plus** field on the **System-Parameters Customer-Options** screen is **y**.
none	Indicates the hunt group does not serve as a message hunt group.

Routing Digits (e.g. AAR/ARS Access Code)

Appears only if the **Message Center** field is **qsig-mwi** or **fp-mwi**. Shows the AAR (most likely for a Message Center type of **qsig-mwi**) or ARS (most likely for a Message Center type of **fp-mwi**) access code which when prepended to the **AUDIX Complete Number** field defines a route to the Message Center switch hunt group containing the line ports to the AUDIX.

Valid entries	Usage
0 to **9**, *****, or #	Enter 1 to 4 digits.

Second Announcement Delay (sec)

Appears only when the **ACD** and **Queue** fields both are **y** and the **Vector** field is **n**. Enter the time in seconds before the call in the queue receives a second recorded announcement or that the second announcement is repeated.

Valid entries	Usage
1 to **99**	Avaya recommends that, if this split/skill or hunt group is a coverage point for another split/skill, this delay should not be more than 15 seconds.
blank	Leave blank if there is no second announcement.

Second Announcement Extension

Appears only when the **ACD** and **Queue** fields both are **y** and the **Vector** field is **n**.

Note:

If entering a Multi-Location Dial Plan shortened extension, note the following: When entering a Multi-Location Dial Plan shortened extension in a field designed for announcement extensions, certain administration end validations that are normally performed on announcement extensions are not done, and resultant warnings or submittal denials do not occur. The shortened extensions also do not appear in any display or list that shows announcement extensions. Extra care should be taken to administer the correct type of announcement for the application when assigning shortened extensions.

Valid entries	Usage
Valid extension	Enter the extension number assigned to a recorded announcement.
blank	Leaving the field blank indicates there is no second announcement.

Second Announcement Recurring

Appears only when the **ACD** and **Queue** fields both are **y** and the **Vector** field is **n**.

Valid entries	Usage
y	The second announcement can be repeated.
blank	Leave blank if there is no second announcement.

Send Reroute Request

Appears only when the **Message Center** field is type **QSIG-MWI** and **Supplementary Services with Rerouting** is **y** on the **System-Parameters Customer-Options** screen.

Valid entries	Usage
y	Default. Rerouting is invoked.
n	Rerouting is not invoked when a call covers through a qsig-mwi hunt group.

Voicemail Number

Appears only if, on the **System-Parameters Customer-Options** screen, the **Basic Call Setup** and **Basic Supplementary Services** fields are **y** and the **Message Center** field on this screen is **qsig-mwi** or **fp-mwi**. The qsig-mwi selection shows the complete number of the AUDIX hunt group on the Message Center server for QSIG MWI. The **fp-mwi** selection shows the public network number of the AUDIX hunt group on the Message Center server.

Valid entries	Usage
Up to 17 digits	Enter the complete AUDIX dial-up number.

Field descriptions for pages 4 through X

Figure 363: Hunt Group Group Member Assignments screen

```
change hunt-group 4                                              Page   4 of  xx
HUNT GROUP
          Group Number: 4 Group Extension: 3001        Group Type: ucd
  Member Range Allowed: 1 - 1500 Administered Members (min/max): 1 /9
                                          Total Administered Members: 9
GROUP MEMBER ASSIGNMENTS
      Ext      Name (24 characters)            Ext       Name (24 characters)

 1 :1011                                  14 :
 2 :1012                                  15 :
 3 :1013                                  16 :
 4 :                                      17 :
 5 :                                      18 :
 6 :                                      19 :
 7 :                                      20 :
 8 :                                      21 :
 9 :                                      22 :
10 :                                      23 :
11 :                                      24 :
12 :                                      25 :
13 :                                      26 :

More Members Exist
```

Administered Members (min/max)

Appears on all member pages. Indicates the minimum and maximum member number administered for this hunt group.

At End of Member List

This display-only field shows the current page is also the last page.

Group Extension

This display-only field shows the extension of the hunt group.

Group Number

This display-only field shows the number of a hunt group.

Group Type

This display-only field shows the type of the hunt group.

Member Range Allowed

The range of allowed members displays on all member pages. These values vary depending on the particular system and/or configuration.

More Members Exist

This display-only field shows there are more members than currently displayed (the current page is not the last page).

Total Administered Members

Appears on all member pages. Indicates the total number of members administered for this hunt group.

GROUP MEMBER ASSIGNMENTS

Ext

A display-only field if the **Controlling Adjunct** field is **asai**. Controlled Agent extensions must be entered on the **Adjunct Controlled Agent Table** screen. The extension cannot be a VDN. The data module cannot be a member of an ACD split/skill. Use this field when the **Controlling Adjunct** field is **none**.

Valid entries	Usage
.Valid extension.	Enter the assigned station or attendant console extension.

Name

This display-only field shows the name assigned to the above extension number when it is administered in the system.

Integrated Announcement Boards

You can move integrated announcement boards that have been previously administered on the **Announcements/Audio Sources** screen to a new board location. You can also display a list of all administered integrated announcement circuit packs.

Figure 364: Integrated Announcement Boards screen

```
display integrated-annc-boards                                  Page 1 of x

                          INTEGRATED ANNOUNCEMENT BOARDS
    Last Board Location Saved:

  Board                   Time                    Number of
  Location       Sfx      Remaining     Rate      Recordings        Checksum ID

1:01D09          C        206           32              7           047a
2:02817          C        248           32              1           00db
3:
4:
5:
```

Last Board Location Saved

Valid entries	Usage
Read-only	Applies to TN750 only; not applicable to VAL

Board Location

Valid entries	Usage
Read-only	The physical location of the integrated announcement circuit pack (UUCSS).

Sfx

Valid entries	Usage
Read-only	The circuit pack suffix letter(s).

Time Remaining

Valid entries	Usage
Read-only	The amount of recording time in seconds remaining on the circuit pack at the 64Kb rate.

Rate

Valid entries	Usage
Read-only	The announcement’s compression rate.

Number of Recordings

Valid entries	Usage
Read-only	The number of nonzero-length announcement recordings or files on the circuit pack.

Checksum ID

Valid entries	Usage
Read-only	Applies to TN750 only; not applicable to VAL.

Integrated Announcement Translations

Use this screen to change board locations currently administered on the **Announcements/ Audio Sources** screen to a new board location.

Figure 365: Change Integrated Announcement Translations screen

```
change integ-annc-brd-loc                                        Page 1 of 1

                    CHANGE INTEGRATED ANNOUNCEMENT TRANSLATIONS
          Change all board location translations from board:     to board:


Changing board locations using this command will change
all currently administered "from" board locations on the
Announcements/Audio Sources screen to the "to" board location.
```

Change all board location translations from board

Valid entries	Usage
board; cabinet 1 to 3; carrier A-E; slot 1 to 20; or gateway 1 to 10; module V1 to V9	Enter a TN2501 board that is currently administered on the **Announcements/Audio Sources** screen.

to board

Valid entries	Usage
board; cabinet 1 to 3; carrier A to E; slot 1 to 20; or gateway 1 to 10; module V1 to V9	Enter a TN2501 board to which you want to move announcement translations that are currently administered on the **Announcements/Audio Sources** screen.

Intercom Group

This screen assigns extensions to intercom groups.

Figure 366: Intercom Group screen

```
change intercom-group 1                                      Page 1 of 2
                                  INTERCOM GROUP
                                               Group Number: 1
                                      Length of Dial Code: _

GROUP MEMBER ASSIGNMENTS
               Ext       DC   Name
           1: _________ __
           2: _________ __
           3: _________ __
           4: _________ __
           5: _________ __
           6: _________ __
           7: _________ __
           8: _________ __
           9: _________ __
          10: _________ __
          11: _________ __
          12: _________ __
          13: _________ __
          14: _________ __
          15: _________ __
          16: _________ __
```

DC

This field assigns a dial code to an extension. The dial code is the code users must dial to make intercom calls to the corresponding extension.

Valid entries	Usage
1 or 2-digit code	The number of digits entered must exactly match the number assigned in the **Length of Dial Code** field. For example, if the **Length of Dial Code** field is set to 2, you must type 1 as 01 in the **DC** field. This field cannot be blank.

Ext

This field assigns an extension to the group.

Valid entries	Usage
an extension number	Enter a physical extension number. You may not enter a VDN in this field.

Group Number

This display-only field shows the group's ID number.

Length of Dial Code

This field sets the number of digits that users must dial to access an extension in the group. (On Page 2, this is a display-only field.

Valid entries	Usage
1	Enter **1** if there are 9 or fewer members.
2	Enter **2** if there are 10 or more members.

Name

Display-only field. The server running Communication Manager fills in this field with the name from the **Station** screen.

Related topics

See Using Telephones as Intercoms on page 502 for instructions.

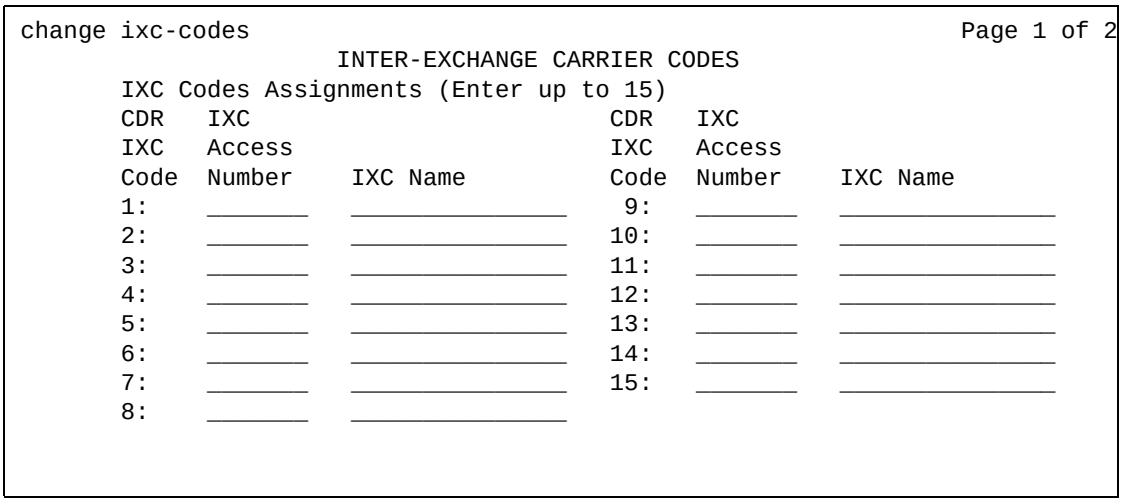
See "Intercom" in *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205, for more information.

Inter-Exchange Carrier (IXC) Codes

This screen allows identification of the IXC in the CDR record.

Field descriptions for page 1

Figure 367: Inter-Exchange Carrier Codes screen



```
change ixc-codes                                                Page 1 of 2
                      INTER-EXCHANGE CARRIER CODES
       IXC Codes Assignments (Enter up to 15)
       CDR   IXC                          CDR   IXC
       IXC   Access                       IXC   Access
       Code  Number    IXC Name           Code  Number    IXC Name
       1:    _______   ________________    9:   _______   ________________
       2:    _______   ________________   10:   _______   ________________
       3:    _______   ________________   11:   _______   ________________
       4:    _______   ________________   12:   _______   ________________
       5:    _______   ________________   13:   _______   ________________
       6:    _______   ________________   14:   _______   ________________
       7:    _______   ________________   15:   _______   ________________
       8:    _______   ________________
```

IXC Access Number

Valid entries	Usage
2 to 11 digits, **0** to **9** and *****	Enter the digits dialed or inserted by AAR/ARS into the outpulsed digit string to access the interexchange carrier. No duplicate access numbers are allowed in the table.

IXC Name

Valid entries	Usage
0 to 15 characters	Description to identify the IXC

Field descriptions for page 2

Figure 368: Inter-Exchange Carrier Codes screen

```
change ixc-codes                                              Page 2 of 2
                         IXC Prefix            IXC Code Format
                   1.    ___                   ____
                   2.    ___                   ____
                   3.    ___                   ____
                   4.    ___                   ____
                   5.    ___                   ____
```

IXC Code Format

Valid entries	Usage
1 to 4 digit code format	
*****	
x	
X	
xxxx	For line 1
xxx	For line 2

IXC Prefix

Valid entries	Usage
1 to 3 digit prefix	
*****	
101	For line 1
10	For line 2

Intra-Switch CDR

This screen administers extensions for which Intra-Switch CDR is to be enabled.

Note:

Attendants are not allowed to be optioned for the Intra-Switch CDR feature.

If your system can record more than 100 stations, the system only displays two pages of extensions (112 per page) at one time. When you enter the add command to add extensions, the system automatically begins after the last administered extensions. If you enter the change command, the system display begins with the first extension. If you enter the change command with an extension number, the system begins the display with that extension.

When you enter the command list intra-switch-cdr <extension> count x, the system lists "x" switch extensions administered for Intra-Switch CDR beginning with the extension specified by <extension>. For example, if you enter **`list intra-switch-cdr 81000 count 500`**, the system displays extension 81000 (if it is administered for Intra-Switch CDR) and the next 500 extensions that are administered for Intra-Switch CDR. The display command functions similarly to the change command.

Capacities

The Intra-Switch CDR extension capacities vary from server to server. For more information, see the *Hardware Guide for Avaya Communication Manager*.

Field descriptions for page 1

Figure 369: Intra-Switch CDR screen

```
change intra-switch cdr                                      Page 1 of 2
                          INTRA-SWITCH CDR

Assigned Members:   5     of  1000  administered
 1: 364-0000_ 19:_______ 37:_______ 55:_______ 73:_______ 91:_______
 2: 364-0001_ 20:_______ 38:_______ 56:_______ 74:_______ 92:_______
 3: 364-1000_ 21:_______ 39:_______ 57:_______ 75:_______ 93:_______
 4: 364-2001_ 22:_______ 40:_______ 58:_______ 76:_______ 94:_______
 5: 364-3003_ 23:_______ 41:_______ 59:_______ 77:_______ 95:_______
 6: __ _____ 24:_______ 42:_______ 60:_______ 78:_______ 96:_______
 7: ___ ____ 25:_______ 43:_______ 61:_______ 79:_______ 97:_______
 8: ____ ___ 26:_______ 44:_______ 62:_______ 80:_______ 98:_______
 9: _____ __ 27:_______ 45:_______ 63:_______ 81:_______ 99:_______
10: ______ _ 28:_______ 46:_______ 64:_______ 82:_______ 100._______
11: ______ _ 29:_______ 47:_______ 65:_______ 83:_______ 101:_______
12: _______ 30:_______ 48:_______ 66:_______ 84:_______ 102:_______
13: _______ 31:_______ 49:_______ 67:_______ 85:_______ 103:_______
14: _______ 32:_______ 50:_______ 68:_______ 86:_______ 104:_______
15: __ ____ 33:_______ 51:_______ 69:_______ 87:_______ 105:_______
16: ___ ___ 34:_______ 52:_______ 70:_______ 88:_______ 106:_______
17: ____ __ 35:_______ 53:_______ 71:_______ 89:_______ 107:_______
18: _____ _ 36:_______ 54:_______ 72:_______ 90:_______ 108:_______
```

Assigned Members

Displays the number of extensions currently administered for Intra-switch CDR.

1 through x

Valid entries	Usage
Any valid extension	Enter the local extensions you want to track with Intra-Switch CDR. The number of extensions you can track may vary from one system to the next.

IP Address Mapping

Note:

Enter data in either the **To IP Address** field or the **Subnet Mask** field.

This screen defines feature characteristics that depend on the IP address.

Field descriptions for page 1

Figure 370: IP Address Mapping screen

```
change ip-network-map                                          Page 1 of X

                     IP ADDRESS MAPPING

                                                                 Emergency
                                      Subnet            802.1Q   Location
FROM IP Address   (TO IP Address or Mask)   Region      VLAN     Extension
  1.__2.__3.__0    1.__2.__3.255    24       __1         ___3     ________
  1.__2.__4.__4    1.__2.__4.__4    32       __2         ___0     ________
  1.__2.__4.__5    1.__2.__4.__5    __       __3         ___0     ________
  1.__2.__4.__6    1.__2.__4.__9    __       __4         ___4     ________
___.___.___.___  ___.___.___.___    __       ___         ____     ________
___.___.___.___  ___.___.___.___    __       ___         ____     ________
___.___.___.___  ___.___.___.___    __       ___         ____     ________
___.___.___.___  ___.___.___.___    __       ___         ____     ________
___.___.___.___  ___.___.___.___    __       ___         ____     ________
___.___.___.___  ___.___.___.___    __       ___         ____     ________
___.___.___.___  ___.___.___.___    __       ___         ____     ________
___.___.___.___  ___.___.___.___    __       ___         ____     ________
___.___.___.___  ___.___.___.___    __       ___         ____     ________
```

From IP Address

Defines the starting IP address.

Valid entries	Usage
32-bit address (4 decimal numbers, each in the range 0 to 255)	See the *Administration for Network Connectivity for Avaya Communication Manager* for more information.

To IP Address

Defines the termination of a range of IP addresses.

If this field and the **Subnet Mask** fields are blank when submitted, the address in the **From IP Address** field is copied into this field.

The **Subnet Mask** field data is applied to the **From** field, creating the converted **To IP Address** field information.

Valid entries	Usage
32-bit address (4 decimal numbers, each in the range 0 to 255)	See the *Administration for Network Connectivity for Avaya Communication Manager* for more information.

To IP Address or Subnet Mask

The end of the IP address range can be specified by either entering the last IP address in the range or the **From IP Address** and the number of bits of the subnet mask.

If the **Subnet Mask** field is used, then:

- The mask is applied to the **From IP Address** field, placing zeros in the non-masked rightmost bits. This becomes the stored "From" address.
- The mask is applied to the **To IP Address** field, placing 1’s in the non-masked rightmost bits. This becomes the stored "To" address.

If this field and the **To IP Address** fields are blank when submitted, the address in the **From IP Address** field is copied into the **To IP Address** field.

Valid entries	Usage
0 to 32 or blank	Enter the last IP address in the range or the **From IP Address** and the number of bits of the subnet mask.

Region

Identifies the network region for the IP address range.

Valid entries	Usage
1 to **250** **1** to **80** for DEFINITY SI	Enter the network region number for this interface. This field must contain a non-blank value if the **From IP Address** field on the same row contains a non-blank value.

VLAN

Sends VLAN instructions to IP endpoints such as IP telephones and softphones. This field does not send VLAN instructions to the PROCR (S8300 Media Server, S8700 or S8710 IP-Connect, and S8700 or S8710 Multi-Connect), CLAN, and Media Processor boards.

Valid entries	Usage
0 to **4094**	Specifies the virtual LAN value.
n	Disabled

Emergency Location Extension

This field allows the system to properly identify the location of a caller who dials a 911 emergency call from this station. An entry in this field must be of an extension type included in the dial plan, but does not have to be an extension on the local system. It can be a UDP extension. The entry defaults to blank. A blank entry typically would be used for an IP softphone dialing in through PPP from somewhere outside your network.

If you populate the **IP Address Mapping** screen with emergency numbers, the feature functions as follows:

- If the **Emergency Location Extension** field in the **Station** screen is the same as the **Emergency Location Extension** field in the **IP Address Mapping** screen, the feature sends the extension to the Public Safety Answering Point (PSAP).
- If the **Emergency Location Extension** field in the **Station** screen is different from the **Emergency Location Extension** field in the **IP Address Mapping** screen, the feature sends the extension in the **IP Address Mapping** screen to the Public Safety Answering Point (PSAP).

Valid entries	Usage
0 to **9** (up to 7 digits)	Enter the emergency location extension for this station. Default is blank.

Note:

On the **ARS Digit Analysis Table** screen, you must administer 911 to be call type **emer** or **alrt** in order for the E911 Emergency feature to work properly.

IP Codec Set

The **IP Codec Set** screen allows you to specify the type of codec used for voice encoding and companding (compression/decompression). The main difference between codecs is in the compression algorithm used; some codecs compress the voice data more than others. A greater degree of compression results in lower bandwidth requirements on the network, but may also introduce transmission delays and lower voice quality. Codecs are used for VoIP links between any two VoIP resources or endpoints, e.g., IP telephone to IP telephone, IP telephone to Media Gateway, Media Gateway to Media Gateway, etc.

The default codec is set for G711MU. The G711MU provides the highest voice quality because it does the least amount of compression, but it uses the most bandwidth. The G711MU default setting can be changed to one of two other codecs (and their "flavors") if the G711MU does not meet your desired voice-quality/bandwidth trade-off specification. For example, if a far-end server is not running Avaya Communication Manager, you may need to change the codec to match one that is supported by that server’s software.

Field descriptions for page 1

The order in which the codecs are listed on this screen is the order of preference of usage. A trunk call between any two VoIP resources or endpoints will be set up to use the first common codec listed.

This screen allows you to define the allowed codecs and packet sizes used between VoIP resources. You can also enable silence suppression on a per-codec basis. This screen will dynamically display the packet size in milliseconds for each codec in the set, based on the number of frames you administer per packet.

Figure 371: IP Codec Set screen - page 1

```
change ip-codec-set 1                                          Page 1 of 2

                              IP Codec Set

      Codec Set: 1

     Audio        Silence        Frames    Packet
     Codec        Suppression    Per Pkt   Size (ms)
1: G.711MU_______    y              3         30
2:_______________    _              _
3:_______________    _              _
4:_______________    _              _
5:_______________    _              _
6:_______________    _              _
7:_______________    _              _

Media Encryption: none
```

Audio Codec

Specifies the audio codec used for this codec set.

Valid entries	Usage
G.711A (a-law) **G.711MU** (mu-law) **G.723-5.3** **G.723-6.3** **G.729** **G.729A** **G.729B** **G.729AB**	Enter the codec used for this codec set.

Codec Set

Enter the IP codec set from the menu of options.

Note:

Do not use codec G.723-5.3 with a 4624 IP Telephone or Avaya IP Softphone. The 4624 IP Telephone and Avaya IP Softphone do not work properly with audio codec G.723-5.3. Although you can use one of these phones with codec G.723-G5.3 to dial and connect to other users, you hear no audio on the call. That is, you hear no dial tone, ringing, or voice transmission. Therefore, if you have 4624 IP Telephones or IP Softphones connected to a media gateway, be sure the codec set used by that media gateway's network region does not use G.723-5.3 as its first option.

Frames Per Pkt

Specify the number of frames per packet up to a packet size of 60 milliseconds (ms)

Valid entries	Usage
1 to **6** or blank	The default frame size for G.711 and G.729 codecs is 2 frames (20 ms), and for G.723 is 3 frames (30 ms).

Media Encryption

This field appears only if the **Media Encryption over IP** feature is enabled. It specifies one of three possible options for the negotiation of encryption. The selected option for an IP codec set applies to all codecs defined in that set..

Valid entries	Usage
aes	Advanced Encryption Standard (AES), a standard cryptographic algorithm for use by U.S. government organizations to protect sensitive (unclassified) information. Use this option to encrypt these links: • Server-to-gateway (H.248) • Gateway-to-endpoint (H.323)
aea	Avaya Encryption Algorithm. Use this option as an alternative to AES encryption when: • All endpoints within a network region using this codec set must be encrypted. • All endpoints communicating between two network regions and administered to use this codec set must be encrypted.
none	Media stream is unencrypted. This is the default setting.

Packet Size (ms)

A display-only field showing the packet size in milliseconds.

Silence Suppression

Enables RTP-level silence suppression on the audio stream.

Valid entries	Usage
y/n	Enter **y** to enable RTP-level silence suppression on the audio stream.

Field descriptions for page 2

Use this screen to assign the following characteristics to a codec set:

- Whether or not endpoints in the assigned network region can route fax, modem, or TTY calls over IP trunks
- Which mode the system uses to route the fax, modem, or TTY calls
- Whether or not redundant packets will be added to the transmission for higher reliability and quality

These characteristics must be assigned to the codec set, and the codec set must be assigned to a network region for endpoints in that region to be able to use the capabilities established on this screen.

CAUTION:

If users are using Super G3 fax machines as well as modems, do *not* assign these fax machines to a network region with an IP Codec set that is modem-enabled as well as fax-enabled. If its Codec set is enabled for both modem and fax signaling, a Super G3 fax machine incorrectly tries to use the modem transmission instead of the fax transmission.

Therefore, assign modem endpoints to a network region that uses a modem-enabled IP Codec set, and assign the Super G3 fax machines to a network region that uses a fax-enabled IP Codec set.

Figure 372: IP Codec Set screen page 2

```
change ip-codec-set 1                                          Page   2 of   2

                              IP Codec Set


                     Mode        Redundancy

            FAX      relay         0

          Modem      off           0

        TDD/TTY      us            0
```

FAX Mode

Valid entries	Usage
off	Turn off special fax handling when using this codec set. In this case, the fax is treated like an ordinary voice call. With a codec set that uses G.711, this setting is required to send faxes to non-Avaya systems that do not support T.38 fax.
relay	For users in regions using this codec, use Avaya relay mode for fax transmissions over IP network facilities. This is the default for new installations and upgrades to Communication Manager R2.1.
pass-through	For users in regions using this codec, use pass-through mode for fax transmissions over IP network facilities. This mode uses G.711-like encoding.
t.38-standard	For users in regions using this codec, use T.38 standard signaling for fax transmissions over IP network facilities.

Note:

If you have a telephone that is on an IP trunk too close to a fax machine, the handset can pick up the tones from the fax machine and change itself into the fax mode. To prevent this, set the **FAX** field to **off**, and put the FAX machines in an ARS partition that uses only circuit switched trunks, even for IGW FAX calls.

Modem Mode

Valid entries	Usage
off	Turn off special modem handling when using this codec set. In this case, the modem transmission is treated like an ordinary voice call. This is the default for new installations and upgrades to Communication Manager R2.1. With a codec set that uses G.711, this setting is required to send modem calls to non-Avaya systems.
relay	For users in regions using this codec, use relay mode for modem transmissions over IP network facilities. **Note:** – Modem over VoIP in relay mode is currently available only for use by specific analog telephones that serve as Secure Telephone Units (STUs). Contact your Avaya technical representative for more information.
pass-through	For users in regions using this codec, use pass-through mode for modem transmissions over IP network facilities.

TDD/TTY Mode

Valid entries	Usage
off	Turn off special TTY handling when using this codec set. In this case, the TTY transmission is treated like an ordinary voice call. With a codec set that uses G.711, this setting is required to send TTY calls to non-Avaya systems. However, there may be errors in character transmissions.
US	For users in regions using this codec, use U.S. Baudot 45.45 mode for TTY transmissions over IP network facilities. This is the default for new installations and upgrades to Communication Manager R2.1.
UK	For users in regions using this codec, use U.K. Baudot 50 mode for TTY transmissions over IP network facilities.
pass-through	For users in regions using this codec, use pass-through mode for TTY transmissions over IP network facilities.

IP Interfaces

List IP Interface

Use this screen to view information about the IP interfaces administered on the system. The **all** command option displays information about all types of IP interfaces administered on the system. The **clan** command option displays information about administered C-LANS.

Figure 373: List IP Interface screen

```
list ip-interface                                          Page 1 of 4

                                IP INTERFACES
                                                                  Num
                                                                  Skts    Net
ON Type Slot Code Sfx  Node Name    Subnet Mask   Gateway Address   Warn    Rgn  VLAN
  _____ ____           ____________  255.255.255.0  ___.___.___.___   ___    ___    n
  _____ ____           ____________  255.255.255.0  ___.___.___.___   ___    ___    n
  _____ ____           ____________  255.255.255.0  ___.___.___.___   ___    ___    n
  _____ ____           ____________  255.255.255.0  ___.___.___.___   ___    ___    n
  _____ ____           ____________  255.255.255.0  ___.___.___.___   ___    ___    n
  _____ ____           ____________  255.255.255.0  ___.___.___.___   ___    ___    n
  _____ ____           ____________  255.255.255.0  ___.___.___.___   ___    ___    n
  _____ ____           ____________  255.255.255.0  ___.___.___.___   ___    ___    n
  _____ ____           ____________  255.255.255.0  ___.___.___.___   ___    ___    n
  _____ ____           ____________  255.255.255.0  ___.___.___.___   ___    ___    n
  _____ ____           ____________  255.255.255.0  ___.___.___.___   ___    ___    n
```

ON

A **y** in this field allows use of the ethernet port.

Type

Identifies the type of IP interface. This field appears only when the **all** command option is used.

Slot

Describes the physical port location of the interface on your system. First and second characters are the cabinet number, third character is the carrier, fourth and fifth character are the slot number, six and seventh characters are the circuit number.

Code/Sfx

These fields identify the circuit pack code number for each interface.

Node Name

The unique node name for the IP interface administered on the **Node Names** screen.

Subnet Mask

Identifies the subnet mask associated with the IP address for this IP interface. The subnet mask is a 32-bit binary number that divides the network ID and the host ID in an IP address.

Gateway Address

Shows the address of a network node that serves as the default gateway for each IP interface.

Num Skts Warn

Displays only when the **clan** command option is used. Shows the threshold number of sockets in use for IP endpoint registration on the specified C-LAN before a warning message is logged.

Net Rgn

Shows the network region number for each interface on the system.

VLAN

This field sends VLAN instructions to the PROCR (S8300 Media Server, S8700 or S8710 IP-Connect, and S8700 or S8710 Multi-Connect), CLAN, and Media Processor boards. It does not send VLAN instructions to IP endpoints such as IP telephones and softphones. This field cannot be administered for VAL boards.

Add/Change/Remove/Display IP Interface

Use this screen to assign a network region to an IP interface device, or to administer the threshold number of sockets to be used for endpoint registration of a C-LAN before a warning is logged.

Figure 374: Add IP Interface screen

```
add ip-interface 02c08                                          Page 1 of 1
                                 IP INTERFACES
                    Type: CLAN
                    Slot: 02c08
                Code Sfx: TN799 D
               Node Name: clan1xxxxxxxxxx
              IP Address: 123.456.789.012
             Subnet Mask: ___.___.___.___
          Gateway Address: ___.___.___.___
   Enable Ethernet Pt? n
       Network Region: 20
                  VLAN: 0


Number of CLAN Sockets Before Warning: 400
```

Type

Identifies the type of IP interface.

Valid entries	Usage
c-lan, **VAL**, **medpro, procr**	This field is auto-populated based on the slot location specified in the command line.

Slot

Describes the physical port location of the interface on your system. Auto-populated based on the slot location specified in the command line.

Valid entries	Usage
01 to **44** (DEFINITY R) or **01** to **03** (DEFINITY CSI, SI) or **1** to **64** (S8700 or S8710 IP-Connect / S8700 or S8710 Multi-Connect / S8300 Media Server)	First and second characters are the cabinet number.
A to **E**	Third character is the carrier.
0 to **20**	Fourth and fifth character are the slot number.

Code/Sfx

Display-only fields that identify the interface.

Valid display	Usage
TN799, **TN802**, **TN2302**, **2501AP**	Displays the circuit pack code number.

Node Name

The unique node name for the IP interface administered on the **Node Names** screen.

Valid entries	Usage
Character string (up to 15 characters max.)	Enter the name of the node. This entry must map to the node name administered on the **Node Names** screen.

IP Address

Display only, taken from the **IP Node Names** screen, based on the node name entered.

Valid entries	Usage
32-bit address (4 decimal numbers, each in the range 0 to 255)	Displays the IP address of the interface.

Subnet Mask

The subnet mask is a 32-bit binary number that divides the network ID and the host ID in an IP address.

Valid entries	Usage
255, 254, 252, 248, 240, 224, 192, 128, 0	Identifies the subnet mask associated with the IP address for this IP interface.

Gateway Address

Valid entries	Usage
0 to **255**	Enter the address of a network node that serves as the default gateway for the IP interface. Entering a value in this field poses a potential security risk. For more information on using a default gateway, see *Administration for Network Connectivity for Avaya Communication Manager* (555-233-204).

Enable Ethernet Pt?

Allows use of the ethernet port.

Valid entries	Usage
y/n	Enter **y** to use the ethernet port. Enter **n** before you make changes on this screen.

Net Rgn

Identify the network region for the specified interface.

Valid entries	Usage
1 to **250**	Enter the network region number for this interface.

VLAN

This field sends VLAN instructions to the PROCR (S8300 Media Server, S8700 or S8710 IP-Connect, and S8700 or S8710 Multi-Connect), CLAN, and Media Processor boards. It does not send VLAN instructions to IP endpoints such as IP telephones and softphones. This field cannot be administered for VAL boards.

Valid entries	Usage
0 to **4095**	Specifies the virtual LAN value.
n	Default. Disabled.

Number of CLAN Sockets Before Warning

Appears only when the value in the **Type** field is **C-LAN**. Set the threshold number of sockets in use for IP endpoint registration on the specified C-LAN. When the threshold is reached a warning message (error code 2820) is logged, but IP endpoints can still register with that C-LAN if additional sockets are available.

Valid entries	Usage
1 to **499**	Enter the number of sockets in use on a C-LAN before a warning message is logged. Default is 400.

IP Network Region

Use this screen to configure within-region and between-region connectivity settings for all VoIP resources and endpoints within a given IP region. The first page is used to modify the audio and QoS settings. The **Codec Set** field on this page reflects the CODEC set that must be used for connections between phones within this region or between phones and MedPro/Prowler boards and media gateways within this region. The ability to do NAT shuffling for direct IP-to-IP audio connections is also supported.

Field descriptions for page 1

Figure 375: IP Network Region screen

```
change ip-network-region 1                                    Page 1 of 19
                            IP NETWORK REGION
  Region: 1
Location:                  Home Domain:
    Name:
                                Intra-region IP-IP Direct Audio: no
AUDIO PARAMETERS                Inter-region IP-IP Direct Audio: no
   Codec Set: 1                           IP Audio Hairpinning? y
UDP Port Min: 2048
UDP Port Max: 3028                    RTCP Reporting Enabled? y
                              RTCP MONITOR SERVER PARAMETERS
DIFFSERV/TOS PARAMETERS       Use Default Server Parameters? y
 Call Control PHB Value: 34                Server IP Address:   .   .   .
        Audio PHB Value: 46                      Server Port: 5005
802.1P/Q PARAMETERS                   RTCP Report Period(secs): 5
 Call Control 802.1p Priority: 7
        Audio 802.1p Priority: 6      AUDIO RESOURCE RESERVATION PARAMETERS
H.323 IP ENDPOINTS                                    RSVP Enabled? y
  H.323 Link Bounce Recovery? y              RSVP Refresh Rate(secs): 15
 Idle Traffice Interval (sec): 20     Retry upon RSVP Failure Enabled? y
     Keep-Alive Interval (sec): 6              RSVP Profile: guaranteed-service
             Keep-Alive Count: 5      RSVP unreserved (BBE) PHB Value: 40
```

Note:

The **Display IP Network Region** screen reflects the values that you assign on this screen.

Home Domain

The name or IP address of the domain for which this network region is responsible (i.e., authoritative).

Valid entries	Usage
Up to 20 characters or blank.	Enter the name or IP address of the domain for which this network region is responsible. Note that this will appear in the "From" header of any SIP messages.

Name

Description of the region.

Valid entries	Usage
Up to 20 characters	Describes the region.

Region

A display-only field indicating the number of the region being administered.

MEDIA PARAMETERS

Codec Set

Specifies the codec set assigned to the region

Valid entries	Usage
1 to **7**	Enter the number for the codec set for the region.

Intra-region IP-IP Direct Audio

Allows direct audio connections between IP endpoints within a network region.

Valid entries	Usage
y/n	Enter **y** to save on bandwidth resources and improve sound quality of voice over IP transmissions. An **n** entry might be used if, for example, the IP phones within the region are behind two ro more firewalls.
native(NAT)	Enter **native(NAT)** if the IP address from which audio is to be received for direct IP-to-IP connections within the region is that of the telephone/softphone itself (without being translated by NAT). IP phones must be configured behind a NAT device *before* this entry is enabled.
translated(NAT)	Enter **translated(NAT)** if the IP address from which audio is to be received for direct IP-to-IP connections within the region is to be the one with which a NAT device replaces the native address. IP phones must be configured behind a NAT device *before* this entry is enabled.

Inter-region IP-IP Direct Audio

Allows direct audio connections between IP endpoints in different regions

Valid entries	Usage
y/n	Enter **y** to save on bandwidth resources and improve sound quality of voice over IP transmissions. An **n** entry might be used if, for example, the IP phones within the region are behind two ro more firewalls.
translated(NAT)	Enter **translated(NAT)** if the IP address from which audio is to be received for direct IP-to-IP connections between regions is to be the one with which a NAT device replaces the native address. IP phones must be configured behind a NAT device *before* this entry is enabled.
native(NAT)	Enter **native(NAT)** if the IP address from which audio is to be received for direct IP-to-IP connections between regions is that of the telephone itself (without being translated by NAT). IP phones must be configured behind a NAT device *before* this entry is enabled.

IP Audio Hairpinning

Allows IP endpoints to be connected through the IP circuit pack in the server.

Valid entries	Usage
y/n	Enter **y** to allow IP endpoints to be connected through the media server's IP circuit pack in IP format, without first going through the Avaya DEFINITY TDM bus.

Location

Specifies the location of the IP network region, allowing correct date and time information, and trunk routing based on IP network region.

Note:

If the Multinational Locations feature is enabled, and IP phones derive their network region from the IP Network Map, you must administer this field with a valid value (1 to 250). This allows the IP endpoint to obtain a VoIP resource.

Valid entries	Usage
1 to **44**	(For DEFINITY R, CSI, SI only.) Enter the number for the location for the IP network region. The IP endpoint uses this as its location number. This applies to IP telephones and softphones.
1 to **250**	(For Avaya S8300 Media Server, Avaya S8700 or S8710 Multi-Connect, and Avaya S8700 or S8710 IP-Connect only.) Enter the number for the location for the IP network region. The IP endpoint uses this as its location number. This applies to IP telephones and softphones.
blank	The location is obtained from the cabinet containing the CLAN or the media gateway that the endpoint registered with.

RTCP Reporting Enabled

Valid entries	Usage
y/n	Specifies whether you want to enable RTCP reporting. If this field is set to **y**, then the **RTCP Monitor Server Parameters** fields appear.

UDP Port Range

UDP Port Range Min

Specifies the minimum range of the UDP port number used for audio transport.

Valid entries	Usage
1024 to **65534** defaults: **2048** to **3028**	Enter the lowest UDP port number to be used for audio transport.

UDP Port Range Max

Specifies the maximum range of the UDP port number used for audio transport.

Valid entries	Usage
1025 to **65535** defaults: **2048** to **3028**	Enter the highest UDP port number to be used for audio transport.

RTCP MONITOR SERVER PARAMETERS

RTCP Report Period (secs)

This field only appears when the **Use Default Server Parameters** field is set to **n** and the **RTCP Reporting Enabled** field is set to **y**.

Valid entries	Usage
5 to **30**	Enter the report period for the RTCP Monitor server in seconds.

Server IP Address

This field only appears when the **Use Default Server Parameters** field is set to **n** and the and the **RTCP Enabled** field is set to **y**.

Valid entries	Usage
0 to **255** in nnn.nnn.nnn.nnn format	Enter the IP address for the RTCP Monitor server.

Server Port

This field only appears when the **Use Default Server Parameters** field is set to **n** and the and the **RTCP Enabled** field is set to **y**.

Valid entries	Usage
1 to **65535**	Enter the port for the RTCP Monitor server.

Use Default Server Parameters

This field only appears when the **RTCP Reporting Enabled** field is set to **y**.

Valid entries	Usage
y	Enter **y** to use the default RTCP Monitor server parameters as defined on the **IP Options System Parameters** screen. If set to **y**, you must complete the **Default Server IP Address** field on the **IP Options System Parameters** screen.
n	If you enter **n**, you need to complete the **Server IP Address**, **Server Port**, and **RTCP Report Period** fields that appear.

DIFFSERV/TOS PARAMETERS

Call Control PHB Value

Provides scalable service discrimination in the Internet without per-flow state and signaling at every hop. Use the **IP TOS** field to support the DiffServ codepoint.

Valid entries	Usage
0 to **63**	Enter the decimal equivalent of the Call Control PHB value.

Audio PHB Value

Provides scalable service discrimination in the Internet without per-flow state and signaling at every hop. Use the **IP TOS** field to support the Audio PHB codepoint.

Valid entries	Usage
0 to **63**	Enter the decimal equivalent of the DiffServ Audio PHB value.

Call Control 802.1p Priority

Provides Layer 2 priority for Layer 2 switches.

Valid entries	Usage
0 to **7**	Specifies the 802.1p priority value.

Audio 802.1p Priority

Provides Lay 2 priority for Layer 2 switches.

Valid entries	Usage
0 to **7**	Specifies the Audio 802.1p priority value.

AUDIO RESOURCE RESERVATION PARAMETERS

Retry upon RSVP Failure Enabled

This field only appears if the **RSVP Enabled** field is set to **y**.

Valid entries	Usage
y/n	Specifies whether to enable retries when RSVP fails.

RSVP Enabled

The entry in this field controls the appearance of the other fields in this section.

Valid entries	Usage
y/n	Specifies whether or not you want to enable RSVP.

RSVP Profile

This field only appears if the **RSVP Enabled** field is set to **y**. You set this field to what you have configured on your network.

Valid entries	Usage
guaranteed-service	This limits end-to-end queuing delay from sender to receiver. This setting is best for VoIP applications.
controlled-load	This subset of **guaranteed-service** provides for a traffic specifier, but not end-to-end queuing delay.

RSVP Refresh Rate (secs)

This field only appears if the **RSVP Enabled** field is set to **y**.

Valid entries	Usage
1 to **99**	Enter the RSVP refresh rate in seconds.

RSVP unreserved (BBE) PHB Value

Valid entries	Usage
0 to **63**	The BBE codpoint is used whenever an RSVP reservation is being obtained (pending), or has failed in some way, to provide better-than-best service to the voice stream.

H.323 IP ENDPOINTS

H.323 Link Bounce Recovery

A **y** entry in this field enables the H.323 Link Bounce Recovery feature for this network region. An **n** disables the feature.

Valid entries	Usage
y/n	Specifies whether to enable H.323 Link Bounce Recovery feature for this network region.

Idle Traffic Interval (seconds)

This field represents the maximum traffic idle time after which a TCP Keep-Alive (KA) signal is sent from the endpoint..

Valid entries	Usage
5 to **7200**	Enter the maximum traffic idle time in seconds. Default is **20**.

Keep-Alive Interval (seconds)

Use this field to set the interval between TCP Keep-Alive re-transmissions. When no ACK is received for all retry attempts, the local TCP stack ends the TCP session and the associated socket is closed.

Valid entries	Usage
1 to **120**	Specify the interval between KA retransmissions in seconds. Default is **5**.

Keep-Alive Count

Use this field to set the number of times the Keep-Alive message is transmitted if no ACK is received from the peer.

Valid entries	Usage
1 to **20**	Specify the number of retries when if no ACK is received. Default is **5**.

Field descriptions for Page 2

Beginning with Release 2.0 of Communication Manager, page 2 is the **LSP Names** screen. This page covers the information for the Local Survivable Processor names in priority order.

Figure 376: LSP Names screen

```
 change ip-network-region 3                                  Page 2 of 19

                                 IP NETWORK REGION

LSP NAMES IN PRIORITY ORDER
1
2
3
4
5
6
```

Note:

Earlier releases of Communication Manager display a different page 2.

Field descriptions for Page 3

Each page from page 3 on shows the inter-region connectivity for 15 region pairs. To accommodate the maximum of 250 regions for Linux platforms, up to 17 pages are available for this purpose.

Figure 377: Inter Network Region Connection Management screen

```
change ip-network-region 3                                    Page   3 of 19

                  Inter Network Region Connection Management

 src dst    codec    direct                                          Dynamic CAC
 rgn rgn     set      WAN     WAN-BW limits  Intervening-regions       Gateway
 3   1        1        y          256:Kbits
 3   2        1        n                      1    ___  ___  ___
 3   3        1
 3   4        1        n                      1    ___  ___  ___
 3   5        1        n                      6    ___  ___  ___
 3   6        1        y          :NoLimit
 3   7        1        y          10:Calls
 3   8
 3   9
 3   10
 3   11
 3   12
 3   13
 3   14
 3   15
```

src rgn

The entry in this field identifies the source region for this inter-network connection..

Valid entries	Usage
1 to **250**	Display-only. Shows the source region for this inter-network connection.

dst rgn

The entry in this field identifies the destination region for this inter-network connection..

Valid entries	Usage
1 to **250**	Display-only. Shows the destination region for this inter-network connection.

codec-set

Indicates which codec set is to be used between the two regions.

Valid entries	Usage
1 to **7**	If the two regions are not connected at all, this field should be blank. When the codec set is blank, the **direct-WAN**, **WAN-BW-limits**, and **Intervening-regions** entry fields are not displayed. This field cannot be blank if this route through two regions is being used by some non-adjacent pair of regions.

direct-WAN

The entry in this field indicates whether the two regions (source and destination) are directly connected by a WAN link.

Valid entries	Usage
y/n	The default value is **y(es)** if the **codec-set** field is not blank. If so, the **WAN-BW-limits** field displays, but the **Intervening-regions** fields do not. If the **direct-WAN** field is set to **n(o)**, then the **WAN-BW-limits** field does not display, but the **Intervening-regions** fields are displayed.

WAN-BW limits (value)

This field is used for entry of the bandwidth limits for direct WAN links.

Valid entries	Usage
1 to **9999**	Values for this field may be entered in the number of connections, bandwidth in Kbits/sec, bandwidth in Mbits/sec, or left blank. Default is blank. Note: For Release 2.0, The number must be less than or equal to 65 when the units part of the field is set to Mbits/sec.

WAN-BW limits (units)

The entry in this field the unit of measure corrsesponding to the value entered for bandwidth limitation.

Valid entries	Usage
Calls, Kbits/sec, Mbits/sec, NoLimit	This field allows you to limit bandwidth by number of connections, bandwidth in Kbits/sec, bandwidth in Mbits/sec, or it can be left blank. Default is blank.

Intervening-regions

The entry in this field allows entry of intervening region numbers between the two indirectly-connected regions.

Valid entries	Usage
1 to **250**	Enter up to four intervening region numbers between the two indirectly-connected regions. Note: Entry is not allowed for indirect region paths until all direct region paths have been entered. In addition, the order of the path through the regions must be specified starting from the source region to the destination region.

Dynamic CAC Gateway

This field only appears if the **WAN-BW- limit** field is set to **dynamic**. The gateway must be configured to be a CAC (Call Admission Control) gateway

Valid entries	Usage
1 to **250**	Set the gateway that reports the bandwidth-limit for this link. Default is blank.

IP Node Names

Figure 378: IP Node Names screen

```
change node-names ip                                        Page 1 of X

                              IP NODE NAMES

Name                  IP Address                Name          IP Address
 1. _______________   ___.___.___.___  17. _______________   ___.___.___.___
 2. _______________   ___.___.___.___  18. _______________   ___.___.___.___
 3. _______________   ___.___.___.___  19. _______________   ___.___.___.___
 4. _______________   ___.___.___.___  20. _______________   ___.___.___.___
 5. _______________   ___.___.___.___  21. _______________   ___.___.___.___
 6. _______________   ___.___.___.___  22. _______________   ___.___.___.___
 7. _______________   ___.___.___.___  23. _______________   ___.___.___.___
 8. _______________   ___.___.___.___  24. _______________   ___.___.___.___
 9. _______________   ___.___.___.___  25. _______________   ___.___.___.___
10. _______________   ___.___.___.___  26. _______________   ___.___.___.___
11. _______________   ___.___.___.___  27. _______________   ___.___.___.___
12. _______________   ___.___.___.___  28. _______________   ___.___.___.___
13. _______________   ___.___.___.___  29. _______________   ___.___.___.___
14. _______________   ___.___.___.___  30. _______________   ___.___.___.___
15. _______________   ___.___.___.___  31. _______________   ___.___.___.___
16. _______________   ___.___.___.___  32. _______________   ___.___.___.___
```

Note:

If you are using the Converged Communications Server for SIP Instant Messaging, enter the IP address for the SIP Proxy Server for your network in the corresponding fields on the **IP Node Names** screen.

Name

Identifies the name of the adjunct or server/switch node.

Valid entries	Usage
1 to 15 alpha-numeric characters	Used as a label for the associated IP address. The node names must be unique for each server/switch.

IP Address

The IP address for the node named in the previous field.

Valid entries	Usage
32-bit address (4 decimal numbers, each in the range 0 to 255)	A unique IP address is assigned to each port on any IP device that is used for a connection. See the *Administration for Network Connectivity for Avaya Communication Manager* for more information.

IP-Options System Parameters

Field descriptions for page 1

Figure 379: IP-Options System Parameters screen

```
display system-parameters ip-options                              Page 1 of x

                        IP-OPTIONS SYSTEM PARAMETERS

 IP MEDIA PACKET PERFORMANCE THRESHOLDS
    Roundtrip Propagation Delay (ms)    High: 800      Low: 400
                   Packet Loss (%)     High: 40       Low: 15
                   Ping Test Interval (sec): 20
    Number of Pings Per Measurement Interval: 10

 RTCP MONITOR SERVER
                  Default Server IP Address:
                        Default Server Port: 5005
            Default RTCP Report Period(secs): 5

AUTOMATIC TRACE ROUTE ON
            Link Failue? y

IP DTMF TRANSMISSION MODE
     Intra-System IP DTMF Transmission Mode: in-band-g711
     Inter-System IP DTMF: See Signaling Group Forms

 H.248 MEDIA GATEWAY               H.323. IP ENDPOINT
 Link Loss Delay Timer (min):5     Link Loss Delay Timer (min): 5
                                      Primary Search Time (sec): 75
```

IP-Options System Parameters screen

IP MEDIA PACKET PERFORMANCE THRESHOLDS

Number of Pings Per Measurement Interval

Specifies the number of test pings that comprise a measurement from which the performance values (delay and loss) are calculated.

Valid entries	Usage
10 to **100**	Enter the number. Default is **10**.

Packet Loss (%)

Specifies thresholds to be applied to packet loss rates (as measured by ping) for determining activation/deactivation of signaling group bypass.

High

Valid entries	Usage
0 to **100**	This value cannot be less than the minimum value. Default is **40**.

Low

Valid entries	Usage
0 to **100**	This value cannot be more than the maximum value. Default is **15**.

Ping Test Interval (sec)

Specifies the time between performance test pings for each testable signaling group.

Valid entries	Usage
10 to **999**	Enter the time. Default is **20**.

Roundtrip Propagation Delay (ms)

Specifies thresholds to be applied to roundtrip packet propagation delays (as measured by ping) for use in activating or clearing signaling group bypass.

High

Valid entries	Usage
10 to **9999**	This value cannot be less than the minimum value. Default is **800**.

Low

Valid entries	Usage
10 to **9999**	This value cannot be more than the maximum value. Default is **400**.

RTCP MONITOR SERVER

Default RTCP Report Period(secs)

In conjunction with the IP address and server port, this value tells the IP phones, IP softphones and VoIP media modules how often to send the information (RTCP packets) to the RTCP server

Valid entries	Usage
5 to **99**	Enter the desired number of seconds.

Default Server IP Address

The default server IP address that can be utilized by the **IP Network Region** screen for each administered region.

Valid entries	Usage
0 to **255** in nnn.nnn.nnn.nnn format	A unique IP address is assigned to each port on any IP device that is used for a connection.

Default Server Port

The RTCP monitor is a separate computer that receives RTCP packets from many devices. Avaya Communication Manager pushes these values to IP phones, IP softphones and VoIP media modules, such that they know where to send the data. The IP address is that of the RTCP server. The server port is the TCP/IP port of that RTCP server where the information should be sent.

Valid entries	Usage
1 to **65535**	Enter the port being used as an RTCP monitor.

Automatic Trace Route on Link Failure

In order to diagnose network problems, especially to determine where a network outage exists, Communication Manager initiates an automatic trace-route command when the connectivity between a server and its port networks, media gateways, or IP trunks is lost.

Valid entries	Usage
y	Enter **y** to turn the automatic trace-route command feature on.
n	Enter **n** to turn the automatic trace-route command feature off.

Note:

If you disable the feature, any automatic trace-route currently in progress finishes, and no subsequent trace-route commands are launched or logged (the link failure buffer is cleared).

IP DTMF TRANSMISSION MODE

Intra-System IP DTMF Transmission Mode

Enter the appropriate IP transmission mode.

Valid entries	Usage
in-band	DTMF digits encoded within existing RTP media stream for G.711/G.729 calls. G.723 is sent out-of-band.
in-band-g711	DTMF digits encoded within existing RTP media stream for G.711calls. G.723 and G.729 are sent out-of-band.
out-of-band	DTMF sent over trunks via control link between the CLANs.
rtp-payload	Initially, support for SIP trunks requires the entry of **rtp-payload**.

Inter-System IP DTMF Transmission Mode

Valid entries	Usage
in-band **in-band-g711** **rtp-payload** **out-of-band**	See the **DTMF over IP** field on the **Signaling Group** screen. Initially, support for SIP trunks requires the entry of **rtp-payload**.

MEDIA GATEWAY ANNOUNCEMENT SERVER PARAMETERS

Announcement Server IP Address

Identifies the IP address of the Announcement Server.

Valid entries	Usage
0 to **255**	A unique IP address is assigned to each port on any IP device that is used for a connection.

Announcement Storage Path Name

Indicates the path name on the Announcement Server where the announcements are stored.

Valid entries	Usage
Up to 40 characters or blank	Enter the directory path name where announcements are stored.

Login

Indicates the login to be used by the Media Gateway to access the Announcement Server.

Valid entries	Usage
1 to 10 characters or blank	Enter a login up to 10 characters.

Password

Indicates the password to be used by the Media Gateway to access the Announcement Server.

Valid entries	Usage
1 to 10 characters or blank	Enter a password up to 10 characters.

H.248 MEDIA GATEWAY

Link Loss Delay Timeout (minutes)

This field is to assist with the H.248 link bounce recovery mechanism of the Avaya G700 Media Gateway; specifically, to prevent the call controller from removing all boards and ports prematurely in response to a link bounce.

Valid entries	Usage
1 through **30**	Enter the number of minutes to delay the reaction of the call controller to a link bounce. Default is **5**

H.323 IP ENDPOINT

Link Loss Delay Timer (minutes)

This timer specifies how long the Communication Manager server preserves registration and any stable calls that may exist on the endpoint after it has lost the call signaling channel to the endpoint. If the endpoint does not re-establish connection within this period, MV tears down the registration and calls (if any) of the endpoint. This timer does not apply to soft IP endpoints operating in telecommuter mode.

Valid entries	Usage
1 to **60**	Enter the number of minutes to delay the reaction of the call controller to a link bounce. Default is **5**.

Primary Search Time (seconds)

While the telephone is hung-up, this is the maximum time period that the IP endpoint expends attempting to register with its current Communication Manager server. The need for this timer arises in situations where the current MV server may have a large number of C-LANS. this timer allows the customer to specify the maximum time that an IP endpoint spends on trying to connect to the C-LANS before going to an LSP.

While the IP telephone's receiver is lifted, the endpoint continues trying to re-establish connection with the current server until the call ends.

Valid entries	Usage
15 to **3600**	Enter the number of seconds an IP endpoint spends on trying to connect to the C-LANS before going to an LSP. Default is **75**.

IP Routing

Figure 380: IP Routing screen

```
change ip-route 1                                          Page 1 of 1

                                IP ROUTING

    Route Number:
Destination Node:
    Network Bits:      Subnet Mask:
         Gateway:
           Board:
          Metric:
```

Board

Enter the necessary characters.

Valid entries	Usage
01 to **44** (DEFINITY R) or **01** to **03** (DEFINITY CSI, SI) or **1** to **64** (S8700 or S8710 IP-Connect / S8700 or S8710 Multi-Connect / S8300 Media Server)	First and second characters are the cabinet number
A to **E**	Third character is the carrier
0 to **20**	Fourth and fifth character are the slot number
01 to **04** (Analog TIE trunks) **01** to **31** **1** to **80** (DEFINITY R, CSI) or **1** to **10** (DEFINITY SI) or **1** to **250** (S8700 or S8710 IP-Connect / S8700 or S8710 Multi-Connect / S8300 Media Server)	Six and seventh characters are the circuit number Gateway
V1 to **V9**	Module

Destination Node

The node name of the final destination for this connection.

Valid entries	Usage
The name previously entered on the **Node Names** screen.	Enter the name of the final destination node of the IP route for this connection.

Gateway

The node name of the first intermediate node.

Valid entries	Usage
A name previously entered on the **Node Names** screen and is either a port on the CLAN circuit pack or is identified as a Destination Node on another IP route.	If there are one or more intermediate nodes, the first intermediate node is the Gateway. If there are no intermediate nodes between the local and remote CLAN ports for this connection, the Gateway is the local CLAN port.

Metric

Valid entries	Usage
0 or **1**	Enter **1** on a server that has more than one CLAN circuit pack installed.

Network Bits

This field is a 32-bit binary number that divides the network ID and the host ID in an IP address.

Valid entries	Usage
0 to **32**	Enter the number of Network Bits that corresponds to the Subnet Mask for the IP Route. If you skip this field and enter a Subnet Mask, the system automatically sets the corresponding Network Bits value. For example, if you set this field to **24**, then the system sets the **Subnet Mask** field to **255.255.255.0**.

Route Number

Identifies the IP route.

Valid entries	Usage
1 to **400**	Enter the number of the IP route you want to add or change, or enter **n** for the next available number.

Subnet Mask

Valid entries	Usage
255, 254, 252, 248, 240, 224, 192, 128, 0	Enter the subnet mask associated with this IP route. If you enter a value in the **Network Bits** field, the system automatically completes this field with the corresponding Subnet Mask. For example, if you set this field to **255.255.255.0**, then the system sets the **Network Bits** field to **24**.

There is one-to-one mapping between the **Network Bits** and the **Subnet Mask** fields; entering a value in one field uniquely determines the other field. A list of Subnet Mask addresses and their corresponding Network Bits are shown in Table 14).

Table 14: Network Bits and Subnet Mask fields *1 of 2*

Network Bits	Subnet Mask	Number of Hosts	Network Type
0	0.0.0.0	4,294,967,294	/ 0
1	128.0.0.0	2,147,483,646	/ 1
2	192.0.0.0	1,073,741,822	/ 2
3	224.0.0.0	536,870,910	/ 3
4	240.0.0.0	268,435,454	/ 4
5	248.0.0.0	134,217,726	/ 5
6	252.0.0.0	67,108,862	/ 6
7	254.0.0.0	33,554,430	/ 7
8	255.0.0.0	16,777,214	/ 8
9	255.128.0.0	8,388,606	/ 9
10	255.192.0.0	4,194,302	/ 10
11	255.224.0.0	2,097,150	/ 11
12	255.240.0.0	1,048,574	/ 12
13	255.248.0.0	524,286	/ 13
14	255.252.0.0	262,142	/ 14
15	255.254.0.0	131,070	/ 15

1 of 2

Table 14: Network Bits and Subnet Mask fields *2 of 2*

Network Bits	Subnet Mask	Number of Hosts	Network Type
16	255.255.0.0	65,534	/ 16
17	255.255.128.0	32,766	/ 17
18	255.255.192.0	16,382	/ 18
19	255.255.224.0	8,190	/ 19
20	255.255.240.0	4,094	/ 20
21	255.255.248.0	2,046	/ 21
22	255.255.252.0	1,022	/ 22
23	255.255.254.0	510	/ 23
24	255.255.255.0	254	/ 24
25	255.255.255.128	126	/ 25
26	255.255.255.192	62	/ 26
27	255.255.255.224	30	/ 27
28	255.255.255.240	14	/ 28
29	255.255.255.248	6	/ 29
30	255.255.255.252	2	/ 30
31	255.255.255.254	1	/31
	255.255.255.255	0	/32
			2 of 2

Note:

For the **Network Bits** and **Subnet Mask** fields, if you put a value into either field and then press **Enter** or **Tab** to move the cursor to another field, the other field gets populated automatically with a value corresponding to the one you just entered.

IP Services

Figure 381: IP Services screen

```
change ip-services                                          Page 1 of X

                              IP SERVICES
Service      Enabled    Local      Local          Remote       Remote
  Type                   Node       Port           Node         Port
DLG_________    y      clan10       5678   ____________________ _____
DLG_________    y      clan2        5678   ____________________ _____
DLG_________    y      clan3        5678   ____________________ _____
DLG_________    y      clan4        5678   ____________________ _____
DLG_________    y      clan5        5678   ____________________ _____
DLG_________    y      clan6        5678   ____________________ _____
DLG_________    y      clan7        5678   ____________________ _____
DLG_________    y      clan1        5678   ____________________ _____
____________    _      __________   ____   ____________________ _____
____________    _      __________   ____   ____________________ _____
____________    _      __________   ____   ____________________ _____
____________    _      __________   ____   ____________________ _____
____________    _      __________   ____   ____________________ _____
____________    _      __________   ____   ____________________ _____
____________    _      __________   ____   ____________________ _____
____________    _      __________   ____   ____________________ _____
____________    _      __________   ____   ____________________ _____
```

Enabled

When the **Service Type** field is **DLG**, this field controls whether the co-resident DLG application listens to the interface to which it is bound.

Valid entries	Usage
y	Enter **y** to enable this IP service.
n	Does not listen over the interface.

Local Node

Specify the node name for the port. When the **Service Type** field is **DLG**, there can be only one entry on this screen with a **Service Type** field of **DLG** and the same Local Node.

Valid entries	Usage
Node names as defined on the **Node Names** screen.	If the link is administered for services over the C-LAN circuit pack, enter a node name defined on the **Node Names** screen.
processor	Processor is only available for S8300 Media Server.

Local Port

Specify the originating port number. When the **Service Type** field is **DLG**, this field becomes **5678** and is read-only.

Valid entries	Usage
5000 to **9999**	**5111** to **5117** for SAT applications **5678** for ASAI
0	For client applications, this defaults to zero.

Remote Node

Specify the server/switch at the far end of the link for SAT. The remote node should not be defined as a link on the **IP Interface** or **Data Module** screens. When the **Service Type** field is **DLG**, this field is cleared and read-only.

Valid entries	Usage
Node name as defined on the **Node Names** screen	For SAT, use a node name to provide added security.
any	Use any available node.

Remote Port

Specify the port number of the destination. When the **Service Type** field is **DLG**, this field is cleared and read-only.

Valid entries	Usage
5000 to **64500**	Use if this service is a client application, such as CDR or PMS. This must match the port administered on the adjunct, PC or terminal server that is at the remote end of this connection.
0	Default for System Management applications.

Service Type

Defines the service provided.

Valid entries	Usage
ALARM1, ALARM2	Available only on DEFINITY R. Use this to connect send alarms over a TCP/IP link.
cbc	Enter cbc to reserve the trunk for outgoing use only to enhance Network Call Redirection.
CDR1, CDR2	Use this to connect either the primary or secondary CDR device over a TCP/IP link.
DAPI	Available only on DEFINITY R, CSI, SI, S8300 Media Server, S8700 or S8710 IP-Connect, S8700 or S8710 Multi-Connect.
DLG	For S8300 Media Server only. Can be entered only if, on the **System-Parameters Customer-Options** screen, the **Co-Res DEFINITY LAN Gateway** field is **y**. Use this service type for co-resident DEFINITY LAN Gateway. For more information, see the *Avaya MultiVantage CallVisor ASAI Technical Reference*. **Note:** If **DLG** is selected, another screen displays. See DLG Administration on page 926.
PMS	
PMS_JOURNAL	Use this to connect the PMS journal printer over a TCP/IP link.

1 of 2

Valid entries	Usage
PMS_LOG	Use this to connect the PMS log printer over a TCP/IP link.
SAT	System administration terminal.
SYS_PRINT	Use this to connect the system printer over a TCP/IP link.
	2 of 2

IP Services screen - Session Layer Timers page

Use this screen to enable reliable protocol for TCP/IP links, and to establish other session-layer parameters. This screen only appears if you enter CDR1, CDR2, PMS_JOURNAL, or PMS_LOG in the **Service Type** field on page 1 or 2.

Figure 382: IP Services screen Session Layer Timer page

```
change ip-services                                          Page   3 of   x

                              SESSION LAYER TIMERS
Service   Reliable   Packet Resp    Session Connect    SPDU    Connectivity
 Type     Protocol      Timer         Message Cntr      Cntr       Timer
CDR1         y          3                  1            1           1
```

Connectivity Timer

Valid entries	Usage
1 to **255**	Indicates the amount of time (in seconds) that the link can be idle before Communication Manager sends a connectivity message to ensure the link is still up.

Packet Resp Timer

Valid entries	Usage
1 to **255**	Determines the number of seconds to wait from the time a packet is sent until a response (acknowledgement) is received from the far-end, before trying to resend the packet.

Reliable Protocol

Indicates whether you want to use reliable protocol over this link.

Valid entries	Usage
y/n	Use reliable protocol if the adjunct on the far end of the link supports it.

Service Type

A display-only field that identifies the service type for which you are establishing parameters.

Valid entries	Usage
CDR1, CDR2	Used to connect either the primary or secondary CDR device over a TCP/IP link.
PMS_JOURNAL	Used to connect the PMS journal printer over a TCP/IP link.
PMS_LOG	Used to connect the PMS log printer over a TCP/IP link.

Session Connect Message Cntr

Valid entries	Usage
1 to **5**	The Session Connect Message counter indicates the number of times Communication Manager tries to establish a connection with the far-end adjunct.

SPDU Cntr

Valid entries	Usage
1 to **5**	The Session Protocol Data Unit counter indicates the number of times Communication Manager transmits a unit of protocol data before generating an error.

ISDN Numbering Calling Party Number Conversion for Tandem Calls

Tandem calls that route to the public network cannot always provide the correct calling party information, resulting in loss of caller ID information. Communication Manager provides a way of modifying the calling party number on a tandem call that lands in the public network.

Use the **Calling Party Number Conversion for Tandem Calls** screen to administer calling party number formats for tandem calls. To generate a calling party number for the public network, the system compares the incoming calling party number to the sets of calling party lengths, calling party prefixes, and trunk groups. When a match is found, the calling party number is constructed by deleting digits identified in the **Delete** field on the **Calling Party Number Conversion for Tandem Calls** screen, and then inserting the digits specified in the **Insert** field. The numbering format specified in the **Format** field is used to determine the encoding of the NPI and TON fields for the calling party number.

Entries on this screen are only exercised if the **Modify Tandem Calling Number** field on the **ISDN Trunk Group** screen is set to **y**. To access the **Calling Party Number Conversion for Tandem Calls** screen, type **`change tandem-calling-party-number`**, and press **Enter**.

Figure 383: Calling Party Number Conversion for Tandem Calls screen

```
change tandem-calling-party-number                    Page   1 of   8
                    CALLING PARTY NUMBER CONVERSION
                           FOR TANDEM CALLS

         CPN               Trk
   Len   Prefix            Grp(s)   Delete    Insert              Format

    5    22                12-99    1         732852              natl-pub
    7    5381234           2        0         303                 lev0-pvt
    __   ________________  _____    __        ________________    ________
    __   ________________  _____    __        ________________    ________
    __   ________________  _____    __        ________________    ________
    __   ________________  _____    __        ________________    ________
    __   ________________  _____    __        ________________    ________
    __   ________________  _____    __        ________________    ________
    __   ________________  _____    __        ________________    ________
    __   ________________  _____    __        ________________    ________
    __   ________________  _____    __        ________________    ________
    __   ________________  _____    __        ________________    ________
    __   ________________  _____    __        ________________    ________
    __   ________________  _____    __        ________________    ________
    __   ________________  _____    __        ________________    ________
```

CPN Len

Use the **Calling Party Length** field to enter the total number of digits in the calling party number.

Valid entries	Usage
1 to **15**	Enter a number between 1 and 15 to indicate calling party number length.
blank	Default. Indicates that all calling party digit string lengths are valid provided the **Modify Tandem Calling Party** field on the **ISDN Trunk Group** screen is set to **y**.

CPN Prefix

Use the **Calling Party Prefix** field to enter the prefix of the tandem calling party number.

Valid entries	Usage
any combination of digits 0 to 9, up to 15 digits	Enter up to 15 digits to indicate the calling party prefix..
blank	Default. Indicates that all trunk groups are valid provided the **Modify Tandem Calling Party** field on the **ISDN Trunk Group** screen is set to **y**. A specific calling party number digit string match is not required, provided other matching criteria for tandem calling party number modification are met.

Trk Grp(s)

Use the **Trunk Groups** field to enter the ISDN trunk group number.

Valid entries	Usage
Valid trunk group or range of group numbers	Enter an ISDN tunk group number, or a range (x to y) of group numbers.
blank	Default. Indicates that all trunk groups are valid provided the **Modify Tandem Calling Numbe** field on the **ISDN Trunk Group** screen is set to **y**.

Delete

Use the **Delete Digits** field to enter the digits to delete in modifying the tandem calling party number.

Valid entries	Usage
1 to **15**	Enter a valid number of deleted digits up to 15.
all	Enter all to indicate that all digits are deleted.
blank	Default. Indicates that all trunk groups are valid provided the **Modify Tandem Calling Party** field on the **ISDN Trunk Group** screen is set to **y**. No digits are deleted from the received calling party number.

Insert

Use the **Insert Digits** field to enter the the digits to insert in modifying the tandem calling party number.

Valid entries	Usage
any combination of digits 0 to 9, up to 15 digits	Enter a valid number of between 1 and 15 to indicate the number of digits to insert.
blank	Default.Indicates that all trunk groups are valid provided the **Modify Tandem Calling Party** field on the **ISDN Trunk Group** screen is set to **y**. The received calling party number is not prefixed with any digits.

Number Format

Use the **Number Format** field to enter the numbering format to use in modifying the tandem calling party number.

Valid entries	Usage
intl-pub, lev0-pvt, lev1-pvt, lev2-pvt, locl-pub, natl-pub, pub-unk, unk-unk	Enter the appropriate format for the tandem calling number.
blank	Indicates that all trunk groups are valid provided the **Modify Tandem Calling Party** field on the **ISDN Trunk Group** screen is set to **y**. The numbering format information is not modified.

Note:

The following end validation checks should be performed for this screen:

- The length of the calling party number (the combination of CPN length, deleted digits and inserted digits) cannot exceed 15 digits.
- The number of deleted digits cannot be greater than the CPN length.
- The number of digits entered for the CPN prefix cannot be greater than the CPN length.

If any of the above are true, an error message displays.

ISDN Trunk Group

This screen assigns an Integrated Services Digital Network (ISDN) trunk group that supports the ISDN and Call-by-Call Service Selection service selection features. The trunk group provides end-to-end digital connectivity and supports a wide range of services including voice and non-voice services, to which users have access by a limited set of CCITT-defined, standard multipurpose interfaces.

The ISDN trunk group may contain ISDN-PRI or ISDN-BRI interfaces. However, it is not possible to use the two types of interfaces in the same trunk groups. The type of interface is chosen when the trunk members are assigned to the trunk group.

When ISDN-PRI interfaces are used on ISDN trunk groups, they may also be used to support the Wideband Switching feature. This is intended to work with the H0 (384 Kbps), H11 (1536 Kbps), H12 (1920 Kbps), and NXDS0 (128 to 1984 Kbps) data services, and to support high-speed video conferencing and data applications.

For more information on ISDN trunk groups, see Administering ISDN Trunk Groups on page 451.

For descriptions of the screens and fields used with non-ISDN trunks, see Trunk Group on page 1534.

Field descriptions for page 1

Many of the fields on the following screens are described in Trunk Group on page 1534. If a field on this screen is unique, it is listed and defined.

Figure 384: ISDN Trunk Group screen

```
add trunk-group next                                        Page 1 of x
                                TRUNK GROUP

 Group Number: xxx                  Group Type: isdn          CDR Reports: y
   Group Name: OUTSIDE_CALL_______________ COR: 1_       TN: 1__      TAC: ____
    Direction: two-way_       Outgoing Display? n          Carrier Medium: PRI/BRI
  Dial Access? n                Busy Threshold: 99_         Night Service: ____
 Queue Length: 0__
 Service Type: _________________      Auth Code? n             TestCall ITC: rest
Usage Alloc:_____________________________Far End Test Line No: _______________
 TestCall BCC: 4                           TestCall Service: _______________
 TRUNK PARAMETERS
          Codeset to Send Display: 6     Codeset to Send TCM,Lookahead: 6
         Max Message Size to Send: 260   Charge Advice: none_____________
   Supplementary Service Protocol: a     Digit Handling(in/out):overlap/overlap
        Digit Treatment: insertion                          Digits: 1234
        Trunk Hunt: cyclical          QSIG Value-Added? n
                                                    Digital Loss Group: __
Incoming Calling Number - Delete: ___ Insert: ________ Format: ________
               Bit Rate: 1200_        Synchronization: async    Duplex: full
  Disconnect Supervision - In? y  Out? n
  Answer Supervision Timeout: 0__
```

Carrier Medium

This field lets you to specify the type of transport medium interface used for the ISDN trunk group. Appears only when the **Group Type** field is **isdn** and, on the **System-Parameters Customer-Options** screen, either the Async. **Transfer Mode (ATM) Trunking** field or the **H.323** field is set to **y**.

Valid entries	Usage
ATM	The trunk is implemented via the ATM Interface circuit pack.
IP	The trunk is implemented via the LAN/WAN (C-LAN)/ MedPro Interface circuit pack as an H.323 trunk group. Also, enter **IP** for SBS signaling trunks.
PRI/BRI	The trunk is implemented as a standard DS1 or BRI interface.

Charge Advice

Use this field to accumulate and access charge information about a call. You already must have set the **CDR Reports** field to **y** or **r** (ring-intvl) before changing this field from its default of **none**. Remember that receiving Advice of Charge during the call (administered as "automatic" or "during-on-request") affects system performance because of the increased ISDN message activity on the signaling channel, which may reduce the maximum call capacity.

Valid entries	Usage
none	Enter **none** if you do not want the system to collect Advice of Charge information for this trunk group.
automatic	Enter **automatic** only if your public network sends Advice of Charge information automatically.
end-on-request	Enter **end-on-request** if Avaya Communication Manager must request charge information with each call, and you want to receive only the final call charge.
during-on-request	Enter **during-on-request** if Avaya Communication Manager must request charge information with each call, and you want charges to display during and at the end of a call.

Codeset to Send Display

This field defines the codeset for sending the information element for display. The value depends on the type of server/switch to which the user is connected.

Valid entries	Usage
0	CCITT (non-Communication Manager equipment)
6	Any other than CCITT or System 85 R2V4, 4E11
7	System 85 R2V4, 4E11

Codeset to Send National IEs

This field defines the codeset for sending the information element (IE) for national IEs. National IEs include all IEs previously sent only in code set 6 (such as DCS IE). Now these national IEs, including Traveling Class Marks (TCMs) and Lookahead Interflow (LAI), can be sent in code set 6 or 7. The value depends on the type of server/switch the user is connected to.

Valid entries	Usage
6	Other types.
7	System 85 R2V4, 4E11, or newer media server/switch types

Note:

A Traveling Class Mark (that is, the user's FRL or the user's trunk group FRL) is passed between tandem nodes in an ETN in the setup message only when the **Service Type** field is **tandem**. It then is used by the distant tandem switch to permit access to facilities consistent with the originating user's privileges.

Digit Handling (in/out)

This field defines whether overlap receiving and overlap sending features are enabled.

Valid entries	Usage
enbloc/enbloc	Set the field to **overlap** when you want overlap receiving or overlap sending. Set to **enbloc** when you do not want these features enabled. The first field value indicates digit receiving and the second value indicates digit sending.
enbloc/overlap	
overlap/enbloc	
overlap/overlap	

Without overlap receiving or sending enabled, the digits on incoming and outgoing calls are sent enbloc. If the **Digit Handling** field is **overlap/enbloc** or **overlap/overlap**, the following results:

- Incoming Call Handling Treatment table does not appear
- The **Digit Treatment** and **Digits** fields appear
- Warning message indicates that all **Incoming Call Handling** entries are removed when screen is submitted
- When screen is submitted with these values, all **Incoming Call Handling** entries are removed

Far End Test Line No.

Specifies the number sent to the far-end's ISDN test line extension. When the test trunk long command is issued, this exact number is sent to the far-end to establish a call that tests the integrity of the trunk member under test. The number does not pass through routing or undergo digit manipulation. The digits entered must be what the far-end expects. For example, for an ISDN tandem trunk, the far-end test number should be a 7-digit ETN (Electronic Tandem Network) number.

Valid entries	Usage
Up to 15 digits	Enter a code to test signaling channel
blank	

Incoming Calling Number Insert

Valid entries	Usage
Enter up to 15 characters (**0** to **9**), **all**, or blank	Enter the number of digits to insert in the calling party number for all incoming calls on this trunk group.

Incoming Calling Number Format

This field indicates the TON/NPI encoding applied to CPN information modified by the CLI Prefix feature. This encoding does not apply to calls originating locally. The **Numbering Format** field on page 2 of this screen applies to calls originated from this server running Communication Manager.

If this field is blank, Avaya Communication Manager passes on the encoding received in the incoming setup message. If the incoming setup message did not contain CPN information and digits are added, the outgoing message will contain these digits. If the **Format** field is blank in this case, the value defaults to pub-unk.

If the **Format** field on page 2 of this screen is also administered as **unknown**, the trunk group is modified to "unk-unk" encoding of the TON/NPI. Therefore, this field also must contain a value other than **unknown**.

Note:

The values for this field map to the **Type of Numbering (TON)** and **Numbering Plan Identifier (NPI)** values shown below.

Valid entries	Type of numbering (TON)	Numbering plan identifier (NPI)
blank	incoming TON unmodified	incoming NPI unmodified
natl-pub	national(2)	E.164(1)
intl-pub	international(1)	E.164(1)
locl-pub	local/subscriber(4)	E.164(1)
pub-unk	unknown(0)	E.164(1)
lev0-pvt	local(4)	Private Numbering Plan - PNP(9)
lev1-pvt	Regional Level 1(2)	Private Numbering Plan - PNP(9)
lev2-pvt	Regional Level 2(1)	Private Numbering Plan - PNP(9)
unk-unk	unknown(0)	unknown(0)

Max Message Size to Send

Defines Communication Manager's maximum ISDN message size. Currently, the system can receive 260-byte messages. Valid entries are **128**, **244**, **256**, and **260**.

The following table indicates the expected ISDN-PRI message size from several Lucent Technologies and Avaya Inc. products.

Products	Message Length (octets) Received
4ESS (4E11)	256
4ESS (4E13)	256
4ESS (4E14)	256
5ESS (5E4)	244
5ESS (5E5)	244
5ESS (5E6)	244
System 75 (all)	260
System 85 (R2V4)	128
System 85 (R2V5)	260
System 85 (R2V6)	260

QSIG Value-Added

Provides QSIG-VALU services. This field appears only if the **Value-Added (VALU)** field on the **System-Parameters Customer-Options** screen is **y**. This field can be set to **y** only if the **Supplementary Service Protocol** field on the **System-Parameters Customer-Options** screen is **b**. Valid entries are **y** and **n**. Blank is not a valid entry.

Service Type

Indicates the service for which this trunk group is dedicated. The following table provides a listing of predefined entries. In addition to the Services/Features listed in this table, any user-defined **Facility Type** of **0** (feature) or **1** (service) on the **Network Facilities** screen is allowed.

As many as 10 (for Avaya DEFINITY Server SI or CSI) or 200 (for DEFINITY Server R) ISDN trunk groups can have this field administered as **cbc**.

Valid entries	Usage
access	A tie trunk giving access to an Electronic Tandem Network.
accunet	ACCUNET Switched Digital Service — part of ACI (AT&T Communications ISDN) phase 2.
cbc	Call-by-Call service — provides different dial plans for different services on an ISDN trunk group. Indicates this trunk group is used by the Call-By-Call Service Selection feature.
dmi-mos	Digital multiplexed interface — message oriented signaling.
i800	International 800 Service — allows a subscriber to receive international calls without a charge to the call originating party.
inwats	INWATS — provides OUTWATS-like pricing and service for incoming calls.
lds	Long-Distance Service — part of ACI (AT&T Communications ISDN) phase 2.
megacom	MEGACOM Service — an AT&T communications service that provides unbanded long-distance services using special access (switch to 4ESS switch) from an AT&T communications node.
mega800	MEGACOM 800 Service — an AT&T communications service that provides unbanded 800 service using special access (4ESS switch to switch) from an AT&T communications node.

1 of 2

Valid entries	Usage
multiquest	AT&T MULTIQUEST Telecommunications Service — dial 700 service. A terminating-user's service that supports interactive voice service between callers at switched-access locations and service provides directly connected to the AT&T Switched Network (ASN).
operator	Network Operator — provides access to the network operator.
outwats-bnd	OUTWATS Band — WATS is a voice-grade service providing both voice and low speed data transmission capabilities from the user location to defined service areas referred to as bands; the widest band is 5.
public-ntwrk	Public network calls — It is the equivalent of CO (outgoing), DID, or DIOD trunk groups. If Service Type is public-ntwrk, Dial Access can be set to **y**.
sddn	Software Defined Data Network — provides a virtual private line connectivity via the AT&T switched network (4ESS switches). Services include voice, data, and video applications. These services complement the SDN service. Do not use for DCS with Rerouting.
sdn	Software Defined Network (SDN) — an AT&T communications offering that provides a virtual private network using the public switched network. SDN can carry voice and data between customer locations as well as off-net locations.
sub-operator	Presubscribed Common Carrier Operator — provides access to the presubscribed common carrier operator.
tandem	Tandem tie trunks integral to an ETN
tie	Tie trunks — general purpose
wats-max-bnd	Maximum Banded Wats — a WATS-like offering for which a user's calls are billed at the highest WATS band subscribed to by users.

2 of 2

Supplementary Service Protocol

Indicates which supplementary service protocol to use for services over this trunk group. Supplementary Service protocols are mutually exclusive.

Valid entries	Usage
a	National
b	ISO/ETSI QSIG Private Network. Also used for SBS signaling trunks.
c	ETSI public network
d	European Computer Manufacturer's Association (ECMA) QSIG private network (supports only Name Identification and Additional Network Feature Transit Counter (ANF-TC))
e	DCS with Rerouting - Do not use the **Service Type** field entry of **dmi-mos** or **sddn** with this option. - Set the Used for DCS field (on page 2) to **y**.
f	ISDN Feature Plus Public network feature plus signaling.
g	ANSI. Available only if, on the **System-Parameters Customer-Options** screen, the **ISDN-PRI** or **ISDN-BRI** field is **y** or the **Used for DCS** field is **y**.

Test Call BCC

Indicates the Bearer Capability Code (BCC) used for the ISDN test call.

Valid entries	Usage
0	Voice
1	Mode 1
2	Mode 2 Asynchronous
3	Mode 3 Circuit
4	Mode 0

Testcall ITC

Controls the encoding of the Information Transfer Capability (ITC) codepoint of the bearer capability Information Element (IE) in the SETUP message when generating an ISDN test call. Allowed values are **rest** (restricted) and **unre** (unrestricted).

Note:

ISDN Testcall feature has no routing, so a testcall is never blocked due to an incompatible ITC.

Testcall Service

Specifies the call-by-call selection for an ISDN test call. Only appears if the **Service Type** field is **cbc**. Valid entries are all of the services listed in Service Type on page 1176, excluding **sddn** or any new **Facility Type** of **0** (feature), **1** (service), or **3** (outgoing) that is defined by users on the **Network Specific Facility Encoding** screen.

Trunk Hunt

Avaya Communication Manager performs a trunk hunt when searching for available channels within a facility in an ISDN trunk group. With both **ascend** and **descend**, all trunks within an ISDN trunk group are selected based on this field and without regard to the order in which trunks are administered within the trunk group. When using ISDN-BRI interfaces, only **cyclical** is allowed.

Valid entries	Usage
ascend	Enter to enable a linear trunk hunt search from the lowest to highest numbered channels.
cyclical	Enter to enable a circular trunk hunt based on the sequence the trunks were administered within the trunk group.
descend	Enter for a linear trunk hunt search from the highest to lowest numbered channels.

Note:

The cyclical option cannot be set if the trunk group using ISDN-PRI interfaces is to be used for Wideband operations (the **Wideband Support** field set to **y**).

The search can be administered per ISDN-PRI trunk group, but it infers the direction of search within all ISDN-PRI facilities (or portions of those facilities) administered within the trunk group.

Usage Alloc

Appears when the **Service Type** field is **cbc**.

Valid entries	Usage
y/n	Enter **y** to allocate service provided by the trunk group. Use **y** to enhance Network Call Redirection. When you enter **y**, the **CBC Trunk Group Usage Allocation Plans** screen and the **CBC Trunk Group Usage Allocation Plan Assignment Schedule** appear.

Field descriptions for page 2

Figure 385: ISDN Trunk Group screen

```
add trunk-group next                                          Page 2 of x
                              TRUNK FEATURES
           ACA Assignment? n              Measured: none____
                                                    Maintenance Tests? y
                                Data Restriction? n


    Abandoned Call Search? n
    Suppress # Outpulsing? n

        Charge Conversion: 1
            Decimal Point: none
          Currency Symbol:
              Charge Type: units
                                     Per Call CPN Blocking Code:
                                   Per Call CPN Unblocking Code:

              Outgoing ANI:                        Ds1 Echo Cancellation? n
```

Figure 386: ISDN Trunk Group screen

```
add trunk-group next                                          Page 2 of x
                              TRUNK FEATURES
           ACA Assignment? n Measured: none____  Wideband Support? n
  Long Holding Time(hours: _      Internal Alert? _      Maintenance Tests? y
 Short Holding Time (sec): _   Data Restriction? n    NCA-TSC Trunk Member: 7
  Short Holding Threshold: __          Send Name: n    Send Calling Number: n
            Used for DCS? n             Hop Dgt: _  Send Connected Number: n
    Suppress # Outpulsing? n    Numbering Format: ______    DCS Signaling:n
Outgoing Channel ID Encoding: _________
        Charge Conversion: 1____                  UUI IE Treatment: shared
            Decimal Point: none__   Maximum Size of UUI IE Contents: 128
          Currency Symbol: ___
              Charge Type: units__      Send Called/Busy/Connected Number: n
                                            Modify Tandem Calling Number?: n

  Send UCID? n           BSR Reply-best DISC Cause Value: 31
  Send Codeset 6/7 LAI IE? y

Path Replacement with Retention? n
Path Replacement Method: better-route
                    SBS? y
```

These fields are unique to the **ISDN Trunk Group** screen.

BSR Reply-best DISC Cause Value

Servers running Avaya Communication Manager that are polled as resources in a Best Service Routing application return data to the polling media server or switch in the ISDN DISC message. Since some cause values do not work over some networks, this field sets the cause value that your server will return in response to a BSR status poll. If this field is set incorrectly, incoming status poll calls over this trunk group will be dropped before any data is returned to the polling server or switch. This field only appears if the **UUI IE Treatment** field is set to **shared**.

Valid entries	Usage
31 (normal-unspecified)	Enter **31** unless otherwise instructed by Avaya or your network service provider.
17 (user-busy)	
16 (normal-call-clearing)	

CAUTION:

In most cases, this field will be set to the appropriate value during installation. If you need to change it, your network service provider should be able to help you choose an appropriate value. Don’t change this field without the assistance of Avaya or your network service provider.

DCS Signaling

Specifies the means used to send the DCS message. This field only appears if the **Used for DCS** field entry is **y** and the **Service Type** field is anything except **dmi-mos** or **sddn**.

Valid entries	Usage
bx.25	Enter for the traditional DCS feature.
d-chan	Enter for the DCS over ISDN-PRI D-channel feature.

DCS over D-channel is not supported on trunk groups containing ISDN-BRI interfaces.

- Hop Dgt — The Tandem Hop Limitation and QSIG Additional Network Feature Transit Counter (ANF-TC) features provide a counter that reflects the number of switches (that is, the number of hops) that a call has gone through. The counter increments as a call leaves Avaya Communication Manager using tandem facilities. Valid values are **y** and **n**. One or both of the features can be applied to the trunk group depending on the following:
  - If you enter **y** and the **Group Type** field is **tandem** or the **Group Type** field is **isdn** and the **Service Type** field is **tandem**, the Tandem Hop Limitation feature is applied to the trunk group.
  - If you enter **y** and you set the **Group Type** field to **isdn**, set the **Service Type** field to **access**, **dmi-mos**, **public-ntwrk**, **tandem**, **tie**, or any of the craft-defined services allowed in the field. Set the **Supplementary Service Protocol** field to **b** or **d**, then the ANF-TC feature is applied to calls on the trunk group.

  Note:

  The above conditions overlap. If the **Group Type** field is **isdn**, the **Service Type** field is **tandem**, and the **Supplementary Service Protocol** field is **b** or **d**, then both the Tandem Hop Limitation and ANF-TC features are applied to calls on the trunk group.

  - If both features are applied to calls on the trunk group, ANF-TC takes precedence. In situations where Avaya Communication Manager serves as an Incoming or Outgoing Gateway, either feature uses the hop count and transit information provided by the other.

Maximum Size of UUI IE Contents

This field appears when the **UUI IE Treatment** field is **shared**.

Valid entries	Usage
32 to **128**	Enter the maximum number of bytes of user information that the network supports.

Modify Tandem Calling Number

This field appears when **Trunk Group Type** is **ISDN**, Direction is either **Outgoing** or **Two-way**, **Carrier Medium** is **PRI/BRI** or **IP**, and **Send Calling Number** is either **y** or **r**. It is used to control whether call processing processes the entries on the **ISDN -Tandem Calling Party Number** screen.

Valid entries	Usage
y/n	Enter **y** to modify the calling party number IE in the format specified on the **ISDN-Tandem Calling Party Number** screen. Default is **n**.

NCA-TSC Trunk Member

Identifies the trunk member whose D-channel will be used to route tandem NCA-TSCs or QSIG CISCs. Value range for this field is from 1 to the maximum number of members per trunk group supported on the server/switch, or blank.

Network (Japan) Needs Connect Before Disconnect

Sends an ISDN Connect message just prior to the Disconnect message.

Numbering Format

This field appears if the **Send Calling Number** field is **y** or **r** or the **Send Connected Number** field is **y** or **r**. This specifies the encoding of Numbering Plan Indicator for identification purposes in the Calling Number and/or Connected Number IEs, and in the QSIG Party Number. Valid entries are **public**, **unknown**, **private**, and **unk-pvt**. **Public** indicates that the number plan according to CCITT Recommendation E.164 is used and that the Type of Number is national. **Unknown** indicates the Numbering Plan Indicator is unknown and that the Type of Number is unknown. **Private** indicates the Numbering Plan Indicator is PNP and the Type of Number is determined from the **ISDN Private-Numbering** screen. An entry of **unk-pvt** also determines the **Type of Number** from the **ISDN Private-Numbering** screen, but the **Numbering Plan Indicator** is unknown.

Outgoing Channel ID Encoding

Appears only if the **Group Type** field is **isdn** and the **Service Type** field is anything except dmi-mos or **sddn**. Determines whether to encode the Channel ID IE as preferred or exclusive. Blank is not a valid entry. Defaults are determined as follows:

If the **Group Type** field is **isdn** and the **Used for DCS** field is **y**, default is **exclusive**.

If the **Group Type** field is **isdn** and the **Used for DCS** field is **n**, default is **preferred**.

If the **Group Type** field is not **isdn** or it is **isdn**, but the **Used for DCS** field does not appear, default is **preferred**.

Path Replacement Method

Appears when either the **ISDN-PRI trunk** or the **ISDN-BRI trunk** fields and the **Basic Call Setup** and **Supplementary Services with Rerouting** fields are set to **y** on the **System-Parameters Customer-Options** screen and when the **Supplementary Service Protocol** is either **b** or **e** and the **Group Type** field is **isdn** on the **ISDN trunk group** screen.

Valid entries	Usage
better-route	Uses the most economical route, for example, the reconfigured call does not use the same trunk group as the original call.
always	Always reconfigures the call regardless of the trunk group used.

Replace Restricted Numbers

Appears when the **Group Type** field is **isdn**. Indicates whether to replace restricted numbers with administrable strings for incoming and outgoing calls assigned to the specified trunk group. This field applies to BRI and PRI trunks.

Valid entries	Usage
y/n	Enter **y** for the display to be replaced regardless of the service type of the trunk.

Replace Unavailable Numbers

Appears when the **Group Type** field is **isdn**. Indicates whether to replace unavailable numbers with administrable strings for incoming and outgoing calls assigned to the specified trunk group. This field applies to BRI and PRI trunks.

Valid entries	Usage
y/n	Enter **y** for the display to be replaced regardless of the service type of the trunk.

SBS

Appears when the **Local Country Code and International Access Code** fields are administered on the **Feature-Related System-Parameters** screen and when the **Supplementary Service Protocol** is **b** and the **Group Type** field is **isdn** and **Carrier Medium** is **IP** and **Dial Access** is **n** on page 1 of the **ISDN trunk group** screen.

Valid entries	Usage
y/n	Enter **y** to enable Separation of Bearer and Signaling (SBS) for the trunk group. The default is **n** (SBS is not enabled).

Send Called/Busy/Connected Number

Appears if the **QSIG Value-Added** field on the **Trunk Group** screen is **y**. Specifies if the dialed number, whether called (ringing), busy (busy tone), or connected (answered) is sent on incoming or tandemed ISDN calls. Valid entries are **y**, **n**, or **r** (restricted). If **y** is entered, the **ISDN Numbering - Public/Unknown Format** screen is accessed to construct the actual number sent, or the **ISDN Numbering-Private** screen (based on the **Numbering Format** field) is used. If the value is **r**, the connected number is sent "presentation restricted." The **Send Called/Busy/Connected Number** field must be set to **y** in order for the Calling Party Number of an incoming ISDN call to display at the transferred-to station after a QSIG transfer operation.

Send Calling Number

Specifies whether the calling party's number is sent on outgoing or tandemed ISDN calls. Valid entries are y, **n**, or **r** (restricted). If **y** is entered, the **ISDN Numbering - Public/Unknown Format** screen is accessed to construct the actual number to be sent, or the **ISDN Numbering-Private** screen (based on the **Numbering Format** field) is used. If the value is **r**, the calling number is sent "presentation restricted."

When **Send Calling Number** is **n**, an incoming number is not tandemed out again. Similarly, when **Send Calling Number** is **r** (restricted), an incoming number is marked restricted when it is tandemed out again. This applies to all Supplementary Service Protocols.

Note:

The **ISDN Numbering - Public/Unknown Format** screen can override the **Send Calling Number** field entry for any administrable block of extensions.

Send Codeset 6/7 LAI IE

Specifies whether the ISDN trunk should transmit information in Codeset 6/7. If the UUI IE Treatment field is **shared**, then this field should be **n**. Otherwise, the same information will be sent twice and may exceed the message size. Default is **y** for pre-DEFINITY 6.3 compatibility.

Send Connected Number

Appears if the **QSIG Value-Added** field on the **Trunk Group** screen is **n**. Specifies if the connected party's number is sent on incoming or tandemed ISDN calls. Valid entries are **y**, **n**, or **r** (restricted). If **y** is entered, the **ISDN Numbering - Public/Unknown Format** screen is accessed to construct the actual number sent, or the **ISDN Numbering-Private** screen (based on the **Numbering Format** field) is used. If the value is **r**, the connected number is sent "presentation restricted." The **Send Connected Number** field must be set to **y** in order for the Calling Party Number of an incoming ISDN call to display at the transferred-to station after a QSIG transfer operation.

When **Send Connected Number** is **n**, an incoming number is not tandemed out again. Similarly, when **Send Connected Number** is **r** (restricted), an incoming number is marked restricted when it is tandemed out again. This applies to all Supplementary Service Protocols.

Note:

The AT&T Switched Network Protocol does not support restricted displays of connected numbers. Therefore, if you administer the 1a country-protocol/ protocol-version combination on the **DS1** screen, you should not administer the **Send Connected Number** field to **r** (restricted) on the **ISDN Trunk Group** screen, as this causes display problems.

Note:

The **ISDN Numbering - Public/Unknown Format** screen overrides the **Send Connected Number** field entry for any administrable block of extensions.

Network Call Redirection

Administrable if, on the **System-Parameters Customer-Options** screen, the **ISDN-PRI** or **ISDN-BRI Trunks** field is **y**, the **ISDN Network Call Redirection** field is **y**, and on the **ISDN Trunk Group** screen, the **Supplementary Service Protocol** field is **c** or **g**. Whenever the **Supplementary Service Protocol** field is changed, this field resets to **none** to prevent an inadvertent incorrect value.

Valid entries	Usage
ANSI-transfer	Indicates standard ANSI encoding functionality. Allowed only if the **Supplementary Service Protocol** field is **g**.
Nortel-transfer	Indicates non-standard ENCT functionality. Allowed only if the **Supplementary Service Protocol** field is **g**.
deflect	Indicates network call redirection. Allowed only if the **Supplementary Service Protocol** field is **c**.
none	No redirection (default).

Send Name

Specifies whether the calling/connected/called/busy party's administered name is sent to the network on outgoing/incoming calls. Valid entries are **y**, **n**, or **r** (restricted). The value **r** indicates that the calling/connected name will be sent by Avaya Communication Manager, but will be marked "presentation restricted." This value is valid only if the **Supplementary Service Protocol** field is **a** (national supplementary service), **b** (for called/busy only) or **d** for the QSIG Global Networking Supplementary Service Protocol. When the **Supplementary Service Protocol** field is **e** (DCS with Rerouting), only values of **y** and **n** are permitted.

When the **Send Name** field in **n**, an incoming name is not tandemed out again if the **Supplementary Service Protocol** field is any value other than **b** (QSIG). Similarly, when **Send Name** is **r** (restricted), an incoming name is marked restricted when it is tandemed out again. However, if the **Supplementary Service Protocol** field is **b** (QSIG), then an incoming name is passed on unchanged and the **Send Name** field is ignored.

Note:

If name information is not administered for the calling station or the connected/called/busy station, the system sends the extension number in place of the name.

Send UCID

Specifies whether or not the trunk should transmit Universal Call IDs. The valid entries are **y** and **n**.

Send UUI IE

Specifies whether to block sending UUI information on a per trunk group basis. The valid entries are y and **n**.

US NI Delayed Calling Name Update

Administrable if, on the **System-Parameters Customer-Options** screen, the **ISDN-PRI** field is **y**, and on the **Trunk Group** screen, the **Carrier Medium** field is either **PRI/BRI** or **ATM**, and the **Supplementary Service Protocol** field is **a**. This field provides display updates to the terminating telephone for delayed calling party name provided by the network.

Valid entries	Usage
y	If calling name information is received after the incoming call has been delivered to the terminating telephone, there is a display update. **Note:** BRI trunks do not support this value.
n	If calling name information is received after the incoming call has been delivered to the called telephone, there is no display update to the terminating telephone.

UUI IE Treatment

Specifies whether the user Information Element (IE) is shared.

Valid entries	Usage
shared	If the trunk is connected to an Avaya DEFINITY 6.3 (or later) switch or media server.
service-provider	If the trunk is connected to a pre-DEFINITY 6.3 switch, or service provider functionality is desired.

Wideband Support

Specifies whether Wideband Switching is supported by this trunk group. Valid entries are **y** or **n**. For ISDN trunk groups containing ISDN-BRI interfaces, the only valid entry is **n**. Otherwise you can administer this field only if the **Wideband Switching** field is **y** on the **System-Parameters Customer-Options** screen. If set to **y**, the **Wideband Support Options** page appears. All trunk members must be from TN464C (or later) circuit packs.

Note:

Wideband trunk calls are treated as a single trunk call when Automatic Circuit Assurance (ACA) measurements are taken. This way, if an ACA referral call is generated (for short or long holding time), the wideband call only triggers a single referral call using the lowest B-channel trunk member associated with the wideband channel.

Field descriptions for the Shared UUI Feature Priorities page

The fields in this page show the priorities for each type of information to be forwarded in the Shared UUI. This page appears only on the **ISDN trunk group** screen when all of the following conditions are met:

- The **UUI IE Treatment** field is **shared**.
- The **Supplementary Service Protocol** field is set to anything except **b**.

Figure 387: Shared UUI Feature Priorities screen

```
add trunk-group next                                          Page y of x
                        SHARED UUI FEATURE PRIORITIES

                      ASAI: 1

        Universal Call ID: 2

MULTI SITE ROUTING (MSR)

               In-VDN Time: 3
                  VDN Name: 4
          Collected Digits: 5
    Other LAI Information: 6
```

Changing the priorities in this screen may affect whether certain information will be sent. For more information about setting priorities, see "Information Forwarding" in *Avaya Communication Manager Contact Center Guide to ACD Contact Centers*. These fields are unique to the **ISDN trunk group** screen.

ASAI

User information from ASAI. Valid entries are **1** to **6** (**1** is high) and blank. If blank, that field's information is not forwarded.

Collected Digits

Digits collected from caller (not including dial-ahead digits). Valid entries are **1** to **6** (**1** is high) and blank. If blank, that field's information is not forwarded.

In-VDN TIme

Number of seconds the call has spent in vector processing. Valid entries are **1** to **6** (**1** is high) and blank. If blank, that field's information is not forwarded.

Other LAI Information

Includes the time stamp of when the call entered the current queue, the call's priority level in its current queue, and the type of interflow. Valid entries are **1** to **6** (**1** is high) and blank. If blank, that field's information is not forwarded.

Universal Call ID

Unique tag to identify each call. Valid entries are **1** to **6** (**1** is high) and blank. If blank, that field's information is not forwarded.

VDN Name

Name of the active VDN (also called LAI DNIS). Valid entries are **1** to **6** (**1** is high) and blank. If blank, that field's information is not forwarded.

Field descriptions for the Incoming Call Handling Treatment page

Figure 388: Incoming Call Handling Treatment screen

```
add trunk-group next                                          Page y of x
                        INCOMING CALL HANDLING TREATMENT
     Service/    Called     Called      Del  Insert            Per Call     Night
     Feature      Len       Number                            CPN/BN       Serv
   _____________  ___   _____________   ___   _____________ __________   _______
   _____________  ___   _____________   ___   _____________ __________   _______
   _____________  ___   _____________   ___   _____________ __________   _______
   _____________  ___   _____________   ___   _____________ __________   _______
   _____________  ___   _____________   ___   _____________ __________   _______
   _____________  ___   _____________   ___   _____________ __________   _______
   _____________  ___   _____________   ___   _____________ __________   _______
   _____________  ___   _____________   ___   _____________ __________   _______
   _____________  ___   _____________   ___   _____________ __________   _______
   _____________  ___   _____________   ___   _____________ __________   _______
   _____________  ___   _____________   ___   _____________ __________   _______
   _____________  ___   _____________   ___   _____________ __________   _______
   _____________  ___   _____________   ___   _____________ __________   _______
   _____________  ___   _____________   ___   _____________ __________   _______
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   _____________  ___   _____________   ___   _____________ __________   _______
   _____________  ___   _____________   ___   _____________ __________   _______
   _____________  ___   _____________   ___   _____________ __________   _______
   _____________  ___   _____________   ___   _____________ __________   _______
   _____________  ___   _____________   ___   _____________ __________   _______
```

Note:

This page does not appear if the **Digit Handling (in/out)** field is **overlap** on the "in" side or if the **Direction** field is **outgoing**.

The **Incoming Call Handling Treatment** screen can (optionally) provide unique call treatment for different incoming calls on any ISDN trunk group. The number of unique treatments that can be specified in this table and the number of pages vary depending on whether the Usage Allocation Enhancements feature is enabled and on the amount of available system memory.

Unique digit manipulation capabilities, CPN/BN requests, and night service destinations are possible for different types of incoming calls. The unique digit manipulation capabilities can be particularly useful to accommodate different dial plans for different services on an ISDN trunk type with a **Service Type** (field entry) of **cbc** (Call-by-Call). The table can also be used for ISDN trunk groups that are not Call-by-Call. For example, an ISDN group with **Service Type** set to **mega800** could use the Incoming Call Handling Treatment table to distinguish treatment of calls to different 800 numbers based on the Dialed Number Identification Service (DNIS) number that is incoming to Avaya Communication Manager.

Each row in the table consists of seven columns. The first three columns (**Service/Feature**, **Called Len**, and **Called Number**) constitute a key that together select which row or unique treatment should apply for an incoming call on the group. The remaining four columns (**Del**, **Insert**, and so on) specify the treatment to be provided for a call that matches the key.

If an incoming call is for a service listed in a row on the table, then that row may specify the treatment for the call, depending on the other two columns of the key. The **Called Len** field is used to continue the row determination. If the number of digits received with the incoming call matches the number of digits in the **Called Len** field for calls to the matched service, then this row may apply. If no other row also contains a matching service and called length, then this row does apply. If another row does exist with the same service and number length, then the **Called Number** field will be used to continue the row determination.

If the leading digits received with the incoming call match the digits specified in the **Called Number** field, then this row applies to the call. Therefore, with this table, a unique treatment can be given to any incoming call, even if these calls are to the same service or have the same length of digits. The remaining four fields specify the unique treatment for the call once the row has been determined. Together, the **Del** and **Insert** fields can be used to manipulate the incoming number that will be used to route the call. The **Per Call CPN/BN** field can be used to request CPN/BN from AT&T networks for specific calls incoming on the group. The **Night Serv** field is used to have calls of different types routed to different night destinations when night service is in effect.

The **Incoming Call Handling Treatment Table** always automatically rearranges to show the precedence order the system uses to pick an entry. Thus, you can easily predict the behavior of the **Incoming Call Handling Treatment Table** by noting the order in which the entries display. (The entries rearrange after submitting the **Trunk Group** screen. A subsequent change trunk-group or display trunk-group command then shows the rearranged order.)

Avaya Communication Manager traverses the table from top to bottom and picks the first entry that matches all the following criteria:

- The Service /Feature, if applicable, matches
- The Called/Length matches
- The Called Number matches

If the administered **Called Length** or **Called Number** is blank, that criterion is considered successful.

Incoming Call Handling Treatment Table entries with a predefined service/feature always appear before entries with a user-defined service/feature. To control the order in which certain entries appear, you must use user-defined services/features for those entries. For example, you can redefine the predefined mega800 service/feature as an identical user-defined entry with the name m800.

User-defined entries are always listed in the reverse order compared to the way they appear on the **ISDN Network-Facilities** screen. See Packet Gateway Board on page 1333. Thus, given two user-defined services/features ABC and XYZ, you can force XYZ to appear before ABC in an **Incoming Call Handling Treatment Table** by putting **XYZ** after **ABC** on the **Network-Facilities** screen.

Note:

DCS features that use the **remote-tgs** button (on the remote server/switch) do not work when the local trunk group deletes or inserts digits on the incoming call. These buttons try to dial a local TAC. Adding or deleting digits defeats this operation and renders the remote feature inoperable. If digit manipulation is needed, use it on the outgoing side, based on the routing pattern. One reason for digit manipulation is insertion of the AAR feature access code (FAC).

These fields are located on the **Incoming Call Handling Treatment Table** screen.

Called Len

Specifies the number of digits received for an incoming call. A blank entry is used as a "wild card" entry and, when used, means that any length of digits associated with the specified Service/Feature can match in this field. Valid entries are **0** to **21**, or leave blank.

Called Number

Specifies the leading digits received for an incoming call. A blank entry is used as a "wild card" entry and, when used, means that any number associated with the specified Service/Feature can match in this field. Valid entries are up to 16 digits, or leave blank.

Del

Specifies the number of leading digits to be deleted from the incoming Called Party Number. Calls of a particular type may be administered to be routed to a single destination by deleting all incoming digits and then administering the **Insert** field with the desired extension. Valid entries are **1** to **21**, **all**, or leave blank.

Insert

Specifies the digits to be prepended to the front of the remaining digits after any (optional) digit deletion has been performed. The resultant number formed from digit deletion/insertion is used to route the call, provided night service is not in effect. Valid entries are up to 16 characters consisting of a combination from the following: **0** to **9**, *****, #, or leave blank.

Per Call CPN/BN

Specifies when and how to request Calling Party Number (CPN) or Billing Number (BN) for calls of this type. Leave blank when connected to another media server or switch, or when connected to a public network outside North America. Within North America, leave blank when connected to a public network that does not permit customer equipment to request CPN or BN for individual incoming calls. The AT&T Switched Network offers this service under the titles "CPN/BN to Terminating End on a Per-Call Basis" and "ANI (BN) on Request." An entry of **none** indicates Communication Manager will not request either CPN or BN for any incoming calls of this type. Valid entries are **cpn-only**, **bn-only**, **bn-pref** (prefer BN, but accepts CPN), **cpn-pref** (prefer CPN, but accepts BN), **none**, or leave blank. Leave blank when connected to another server/switch or to a network other than the AT&T Switched Network.

Note:

A 4-second delay occurs in terminating the call to the far-end station if the connecting server or switch does not respond to the request.

Night Serv

Specifies a night service extension (can be a VDN extension) per Service/Feature. An entry other than blank overrides **Night Service** entry on page 1 of the screen. This entry can be overridden by the **Trunk/Member Night Service** entry when provided. Valid entries are an assigned extension, the attendant group access code (**attd**), or leave blank.

Service/Feature

Specifies the ISDN Services/Features for an incoming call type. See the **Service Type** field description for a list of predefined Services/Features that can be received. Or, use Type 0, Type 1, and Type 2 user defined services. The identifier other can be used for any Services/Features not explicitly specified.

Field Descriptions for the CBC Trunk Group Usage Allocation page

Appears when the **Service Type** field is **cbc** and the **Usage Alloc** field is **y**.

Figure 389: CBC Trunk Group Usage Allocation screen

```
add trunk-group next                                              Page y of x
                          CBC TRUNK GROUP USAGE ALLOCATION
Usage Allocation Plan 1    Usage Allocation Plan 2    Usage Allocation Plan 3
                 Min# Max#                  Min# Max#                  Min# Max#
Service/Feature  Chan Chan  Service/Feature  Chan Chan Service/Feature  Chan Chan
_______________  __   __    _______________  __   __   _______________  __   __
_______________  __   __    _______________  __   __   _______________  __   __
_______________  __   __    _______________  __   __   _______________  __   __
_______________  __   __    _______________  __   __   _______________  __   __
_______________  __   __    _______________  __   __   _______________  __   __
_______________  __   __    _______________  __   __   _______________  __   __
_______________  __   __    _______________  __   __   _______________  __   __
_______________  __   __    _______________  __   __   _______________  __   __
_______________  __   __    _______________  __   __   _______________  __   __
_______________  __   __    _______________  __   __   _______________  __   __
_______________  __   __    _______________  __   __   _______________  __   __
_______________  __   __    _______________  __   __   _______________  __   __
_______________  __   __    _______________  __   __   _______________  __   __
_______________  __   __    _______________  __   __   _______________  __   __
_______________  __   __    _______________  __   __   _______________  __   __
```

The **CBC Trunk Group Usage Allocation** screen sets a minimum and maximum number of members for up to ten different Services/Features for up to three different Usage Allocation Plans (1 to 3). See "Call-by-call Service Selection" in *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205, for a detailed description of Usage Allocation Plans.

Max# Chan

Indicates the maximum number of members of a ISDN trunk group with a **Service Type** field of **cbc** that a particular Service/Feature can use at any given time. This field must be completed if a Service/Feature has been entered in the **Incoming Call Handling Treatment Table** screen. Valid values are **0** to **99** or blank.

Min# Chan

Indicates the minimum number of members of an ISDN trunk group with a **Service Type** field of **cbc** that a particular Service/Feature can use at any given time. The sum of the minimum number of members for all Service/Features must not exceed the total number of members of the trunk group. Valid values are **0** to **99** or blank.

Service/Feature

Specifies the ISDN Services/Features that can be requested at call setup time when using this trunk group. See Service Type description for a list of predefined Services/Features that can be received on a call by call basis. In addition, the user defined service types can also be used. The identifier **other** is used for all Services/Features not explicitly specified.

Field descriptions for the CBC Service Trunk Group Allocation Plan Assignment Schedule page

Appears when the **Service Type** field is **cbc** and the **Usage Alloc** field is **y**.

Figure 390: CBC Service Trunk Group Allocation Plan Assignment Schedule screen

```
add trunk-group next                                        Page y of x
            CBC SERVICE TRUNK GROUP ALLOCATION PLAN ASSIGNMENT SCHEDULE

  Usage Method:

         Fixed? y      Allocation Plan Number: 1
     Scheduled? n


Usage Allocation Plan Activation Schedule:

       Act  Plan   Act  Plan   Act  Plan   Act  Plan   Act  Plan   Act  Plan        T
ime  #     Time  #     Time  #     Time  #     Time  #     Time  #
Sun     __:__ __    __:__ __    __:__ __    __:__ __    __:__ __    __:__ __
Mon     __:__ __    __:__ __    __:__ __    __:__ __    __:__ __    __:__ __
Tue     __:__ __    __:__ __    __:__ __    __:__ __    __:__ __    __:__ __
Wed     __:__ __    __:__ __    __:__ __    __:__ __    __:__ __    __:__ __
Thu     __:__ __    __:__ __    __:__ __    __:__ __    __:__ __    __:__ __
Fri     __:__ __    __:__ __    __:__ __    __:__ __    __:__ __    __:__ __
Sat     __:__ __    __:__ __    __:__ __    __:__ __    __:__ __    __:__ __
```

The **CBC Service Trunk Group Allocation Plan Assignment Schedule** screen provides for administering a fixed schedule or administering a schedule that can change up to 6 times a day for each day of the week. This screen determines which CBC Service Trunk Group Allocation Plan will be in use at any given time.

Act Time

Indicates the time the usage allocation plan administered in the next field (**Plan #**) will become effective. Enter the time in military time. There must be at least one entry per day. Valid entries are **00:00** through **23:59**.

Allocation Plan Number

Specifies the CBC Trunk Allocation Plan (1 through 3) that is in effect if a fixed usage method has been selected. This field must be assigned if the **Fixed** field is **y**. Valid entries are **1** to **3** or blank.

Fixed

Indicates whether the allocation plan will be fixed. If **y** is entered, the plan number entered in the **Allocation Plan Number** field will be enabled.

Plan

Specifies the number of the usage allocation plan that will be in effect from the activation time until the activation time of the next scheduled plan change. Valid entries are **1** to **3** or blank.

Scheduled

Indicates whether or not the allocation plans will be in effect according to the schedule found on this page. If **y** is entered in this field then there must be at least one entry in the schedule.

Field descriptions for the Wideband Support Options page

Figure 391: Wideband Support Options screen

```
add trunk-group next                                        Page y of x
                      Wideband Support Options

                              H0? n
                             H11? n
                             H12? n
                           NxDS0? y     Contiguous? n
```

The **Wideband Support Options** screen appears immediately before the trunk group member pages. The actual page number will vary.

Note:

All B-channels that comprise the wideband call must reside on the same ISDN-PRI facility. Also, all trunk members in an ISDN trunk group with the **Wideband Support** field set to **y** must be from a TN464C (or later) circuit pack.

H0

Enter **y** to specify the ISDN information transfer rate for 384-kbps of data, which is comprised of six B-channels. When a trunk group is administered to support H0, the trunk/hunt algorithm to satisfy a call requiring 384-kbps of bandwidth uses a fixed allocation scheme.

H11

Enter **y** to specify the ISDN information transfer rate for 1536-kbps of data, which is comprised of 24 B-channels. When a trunk group is administered to support H11, the trunk/hunt algorithm to satisfy a call requiring 1536-kbps bandwidth uses a fixed allocation scheme.

H12

Enter **y** to specify the ISDN information transfer rate for 1920-kbps of data, which is comprised of 30 B-channels. When a trunk group is administered to support H12, the trunk/hunt algorithm to satisfy a call requiring 1920-kbps bandwidth uses a fixed allocation scheme.

Contiguous

Specifies whether or not to hunt contiguous NXDS0 channels. This field only appears if the **NxDS0** field is **y**.

The trunk/hunt algorithm to satisfy an NXDS0 call is as follows:

- Enter **y** to specify the "floating" scheme. NXDS0 calls are placed on a contiguous group of B-channels large enough to satisfy the requested bandwidth without constraint on the starting channel (no fixed starting point trunk).

Note:

H0 and NXDS0 "floating" scheme cannot both be **y**.

- Enter **n** to specify the "flexible" scheme. NXDS0 calls are placed on any set of B-channels on the same facility as long as the requested bandwidth is satisfied. There are no constraints such as contiguity of B-channels or fixed starting points.

NxDS0

Enter **y** to specify the "N by DS-zero" multi-rate service.

Field Descriptions for the Group Member Assignments page

Figure 392: ISDN Group Member Assignments screen

```
add trunk-group next                                        Page y of x
                                TRUNK GROUP
                                Administered Members(min/max): xxx/yyy
GROUP MEMBER ASSIGNMENTS           Total Administered Members: xxx

       Port     Code     Sfx  Name        Night   Sig Grp
   1: _______  _____     _   __________  _____     ___
   2: _______  _____     _   __________  _____     ___
   3: _______  _____     _   __________  _____     ___
   4: _______  _____     _   __________  _____     ___
   5: _______  _____     _   __________  _____     ___
   6: _______  _____     _   __________  _____     ___
   7: _______  _____     _   __________  _____     ___
   8: _______  _____     _   __________  _____     ___
   9: _______  _____     _   __________  _____     ___
  10: _______  _____     _   __________  _____     ___
  11: _______  _____     _   __________  _____     ___
  12: _______  _____     _   __________  _____     ___
  13: _______  _____     _   __________  _____     ___
  14: _______  _____     _   __________  _____     ___
  15: _______  _____     _   __________  _____     ___
```

The total number of pages, and the first page of Group Member Assignments, will vary depending on whether the CBC and Wideband Support pages display.

Note:

When supporting DCS, Member Number Assignments must be the same between nodes (Member #1 must be Member #1 at the far-end trunk group).

Port

When using ISDN-BRI interfaces, B-channel 1 is the port number while B channel 2 is the port number plus 16. For example, if B channel 1's port number is 01A1002, then B channel 2's port number is 01A1018.

When using ISDN-PRI interfaces, the port number will be the one allied with the B-channel. For example, if the DS1 is located in 01A10, then B channel 1 will be 01A1001, B channel 2 will be 01A1002 and so forth.

Sig Grp

Thsi field appears when the **Group Type** field is **isdn-pri**. Enter the signaling group of this trunk group member. Valid entries are from **1** to **650**, and must be configured for IP group members. If you administer a Port that resides on a DS1 board and that DS1 board belongs to one and only one signaling group, you can leave the **Signaling Group** column blank. Then, when you submit the screen, the appropriate default signaling group number is inserted by Communication Manager. If a DS1 board is assigned to more than one Signaling Group, then you must enter a Signaling Group number. You must enter a signaling group if the port is entered as **IP**. A trunk group may contain members from different signaling groups.

Related topics

See "ISDN Service" in *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205, for more information Integrated Services Digital Network trunks.

See "DS1 Trunk Service" in *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205, for more information.

ISDN Numbering — Private

This screen supports Private Numbering Plans (PNP). It allows you to specify the digits to be put in the Calling Number information element (IE), the Connected Number IE, and the QSIG Party Number for extensions in the Private Numbering Plan.

Avaya Communication Manager supports private-network numbers up to 15 digits long. If the total number — including the level 1 and 2 prefixes, the PBX identifier, and the extension — is more than 15 digits long, neither QSIG Party Numbers nor the information elements are created or sent.

Figure 393: ISDN Number — Private Format screen

```
change isdn private-numbering                                  Page 1 of 1

                     ISDN NUMBERING - PRIVATE FORMAT



          Network Level: _                    PBX Identifier: ____
           Level 2 Code: _____                Deleted Digits: _
           Level 1 Code: _____
```

Deleted Digits

In some networks, the leading digits of the number you dial internally to reach any extension are not part of its longer private network number. For example, extension 35581 might have a private network number of 221-5581. To send this network number correctly, you must set the **Deleted Digits** field to **1** (that is, you must tell Avaya Communication Manager to delete 1 digit before inserting the PBX Identifier of 221).

Valid entries	Usage
0	Enter the number of digits to be deleted from the most significant digits of a local extension prior to sending it in the Calling or Connected Number IEs or in the QSIG PartyNumber. This allows UDP network numbers to be converted to PNP Complete Numbers.
1	
2	

Level 1 Code

Enter the first level regional code of the media server or switch in the network. Administer this field carefully. Communication Manager will not check to ensure you have entered a code that supports your entry in the **Network Level** field. You cannot enter anything in this field unless the **Network Level** field is set to **1** or **2**.

Valid entries	Usage
0 to **9**	Enter up to 5 digits.
blank	Because blank regional codes are valid, an entry is not required if the **Network Level** field is **1** or **2**.

In QSIG standards, this level 1 code is called the Level 0 Regional Code.

Level 2 Code

Enter the second level regional code of the media server or switch in the network. Administer this field carefully. The system will not check to ensure you have entered a code that supports your entry in the **Network Level** field. You cannot enter anything in this field unless the **Network Level** field is set to **2**.

Valid entries	Usage
0 to **9**	Enter up to 5 digits.
blank	Because blank regional codes are valid, an entry is not required if the **Network Level** field is 2.

In QSIG standards, this level 2 code is called the Level 1 Regional Code.

Network Level

Enter the value of the highest regional level employed by the PNP network. Use the following table to find the relationship between the network level and the Numbering Plan Identification/ Type of Number (NPI/TON) encoding used in the QSIG PartyNumber or the Calling Number and Connected Number IEs.

Valid entries	Usage
0	NPI - PNP TON - local
1	NPI - PNP TON - Regional Level 1
2	NPI - PNP TON - Regional Level 2
blank	If this field is blank and the **Send Calling Number** and/or **Send Connected Number** field is **y** or **r** with **private** specified for the **Numbering Format** field on the **ISDN Trunk Group** screen, the Calling Number and/or Connected Number IEs will not be sent. If the field is left blank but **private** has been specified in the **Numbering Format** field on the **ISDN Trunk Group** screen, the Identification Number (PartyNumber data type) is sent for QSIG PartyNumbers encoded in ASN.1-defined APDUs. In this case, the ASN.1 data type containing the PartyNumber (PresentedAddressScreened, PresentedAddressUnscreened, PresentedNumberScreened, or PresentedNumberUnscreened) is sent marked as **PresentationRestricted** with **NULL** for the associated digits.

PBX Identifier

Valid entries	Usage
0 to **9** or blank	Enter up to 5 digits. Digits entered in this field are added to the beginning of local extensions. The PBX identifier is added first, before Level 1 or Level 2 codes. For example, in private networks with a 7-digit numbering plan, the first 3 digits of the private-network number are often considered the PBX identifier.

Note:

The PBX identifier does not depend on the network level. Therefore, in a level 0 private network you can use this field to create a level 0 number longer than 5 digits.

ISDN Numbering — Public/ Unknown

This screen supports the ISDN Call Identification Display feature. The feature provides a name/number display for display-equipped stations within an ISDN network. The system uses the caller's name and number and displays it on the called party's display. Likewise, the called party's name and number can be displayed on the caller's display.

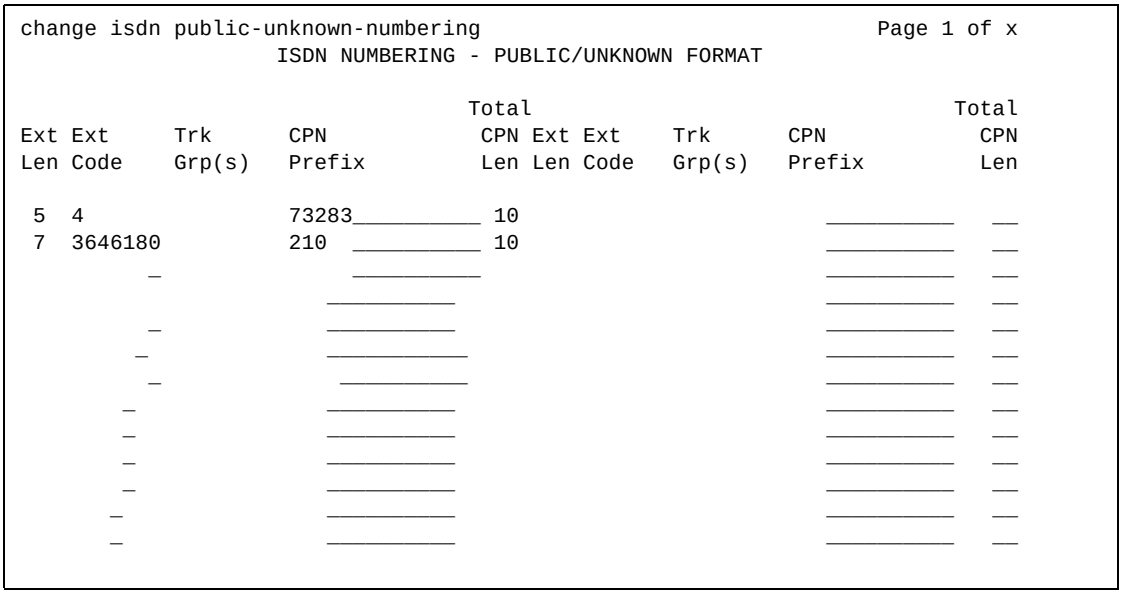
The screen allows you to specify the desired digits for the Calling Number IE and the Connected Number IE (in addition to the QSIG Party Number) for any extension in the Public and/or Unknown Number Plans.

Administer these screens if either the **Send Calling Number**, **Send Connected Number** field is specified, or the **Supplementary Service Protocol** field is **b** on the **Trunk Group** screen.

Note:

If the table is not properly administered and the **Send Calling Number** or **Send Connected Number** field is **y** or **r** and the **Numbering Format** field on the **ISDN Trunk Group** screen is **public** or **unknown**, the Calling Number and Connected Number IE are not sent. If the table is not administered, but the **Send Calling Number** or **Send Connected Number** field is **public** or **unknown**, the Identification Number (PartyNumber data type) is not sent for QSIG PartyNumbers. In this case, the ASN.1 data type containing the PartyNumber (PresentedAddressScreened, PresentedAddressUnscreened, PresentedNumberScreened, or PresentedNumberUnscreened) will be sent marked as **PresentationRestricted** with **NULL** for the associated digits.

Figure 394: ISDN Numbering Public/Unknown screen



CPN Prefix

Use this field to specify the number that is added to the beginning of the extension to form a Calling or Connected Number.

Valid entries	Usage
1 to 15 digits	Only digits are allowed in the CPN Prefix column. Leading spaces, or spaces in between the digits, are not allowed. • If the length of the CPN Prefix matches the Total CPN Length, the extension number is not used to formulate the CPN number. • If the number of digits in the CPN Prefix plus the extension length exceeds the administered Total CPN Length, excess leading digits of the extension are deleted when formulating the CPN number. • If the number of CPN Prefix digits plus the extension length is less than the Total CPN Length, the entry is not allowed. • If the Total CPN Length is 0, no calling party number information is provided to the called party and no connected party number information is provided to the calling party.
blank	If this field is blank, the extension is sent unchanged. This is useful in countries where the public network is able to insert the appropriate CPN Prefix to form an external DID number.

Ext Code

Allows for groups of extensions to be administered.

Note:

When **0** alone is entered, the **Ext Len** field must be 1 and the DDD number must be 10-digits.

Valid entries	Usage
0 to **9** or blank	The **Ext Code** can be up to 7-digits long depending on the **Ext Len** field entry. The entry cannot be greater than the **Ext Len** field entry. For example, in the case of a 4-digit **Ext Len** field entry, an Ext Code of **12** is the equivalent of all extensions of the screen 12xx, excluding any explicitly listed longer codes. If a code of 123 is also listed, the 12 code is equivalent of all extensions of the screen 12xx except extensions of the screen 123x. The coding precludes having to list all the applicable 12xx extensions.
attd	For attendant

Ext Len

Specifies the number of digits the extension can have.

Valid entries	Usage
0 to **7** or blank	Corresponds to the extension lengths allowed by the dial plan.

Total CPN Len

Valid entries	Usage
0 to **15** or blank	Enter the total number of digits to send.

Trk Grp(s)

Communication Manager generates the station's identification number if there is an entry in the **Ext Code** field, and this field is administered with the trunk group number carrying the call.

Valid entries	Usage
1 to 7 digits	Enter the valid administered ISDN trunk-group number or a range of group numbers. For example, if trunk groups 10 through 24 use the same CPN Prefix, enter **10** to **24**.
blank	The identification numbers are not dependent on which trunk group the call is carried.

ISDN-BRI Trunk Circuit Pack

This screen administers an ISDN-BRI circuit pack. See *Hardware Guide for Avaya Communication Manager* (555-245-207) for information on the maximum number of ISDN-BRI circuit packs that you can administer.

Figure 395: BRI Trunk screen (using a TN2185 circuit pack)

```
change bri-trunk-board                                          Page 1 of 1
                         ISDN-BRI TRUNK CIRCUIT PACK

              Location: 01A09                          Name:  _______________
  Interface Companding: a-law_  DCP/Analog Bearer Capability: 3.1kHz
 T3 Timer Length (sec): 15_


Port  Interface  Side  Cntry/Peer TEI  Time  Invert Synch   Layer 1  Detect
                       Protocol       Fill  Bits?  Source? Stable?  Slips?
  1:  user_______        12__     0__  flags   n       n       n       n
  2:  network____        etsi     0__  ones_   y       y       y       y
  3:  user_______        2___     auto flags   y       n       y       n
  4:  peer-slave_  b     QSIG     0__  flags   n       y       y       n
  5:  peer-master  a     QSIG     auto ones_   n       n       n       n
  6:  ___________        ____     0__  ones_   n       n       y       n
  7.  ___________        ____     0__  ones_   n       n       y       n
  8:  ___________        ____     0__  ones_   n       n       y       n
```

Cntry/Peer Protocol

Tells call processing software which ISDN protocol standard is applied.

Valid entries	Usage
1 to **25**	When this field is **10**, **12**, **13**, or **etsi**, the **Protocol Version** field is equivalent to **b** on the **DS1 Circuit Pack** screen. When the **Cntry/Peer Protocol** field is set **10**, **12**, **13**, or **etsi**, set the **Provol Version** field to **b**. For all other administered values, the **Protocol Version** sets to **a**.
etsi	
QSIG	When the **Interface** field is **peer-slave** or **peer-master**, this field must be **QSIG**. The choice **QSIG** is valid only when the Interface field is **peer-slave**.
blank	You cannot leave this field blank if the **Interface** field is set to a valid, non-blank value

DCP/Analog Bearer Capability

Valid entries	Usage
3.1kHz	Indicates how to encode the Bearer Capability IE for an outgoing call originated by a DCP or analog endpoint.
speech	

Detect Slips

Valid entries	Usage
y/n	Tells maintenance software whether slips reported by the BRI port should be logged.

Interface

Valid entries	Usage
network	Tells call processing software whether a particular port is connected to a user/network or a peer interface. These entries are valid for the TN2185. You can enter **peer-slave** only if the QSIG Basic Call Setup feature is enabled
user	
peer-master	
peer-slave	

Interface Companding

Valid entries	Usage
a-law	Indicates the companding algorithm expected by the system at the far end.
mu-law	

Invert Bits

Valid entries	Usage
y/n	This parameter is needed by the TN2185 board. Blank is not a valid entry for this field. Default is **n**.

Layer 1 Stable

Valid entries	Usage
y	Tells call processing and maintenance software whether to expect the network to drop Layer 1 when the BRI port is idle. Only the TN2185 may be set to **n**.
n	

Location

This is a display-only field. It shows the TN2185 circuit pack location (PPCSS)

Name

Valid entries	Usage
1 to 15 alpha-numeric characters	This name is used to identify the circuit pack.

Port

This is a display-only field. It shows the port number to which parameters administered on the row apply.

Side

Valid entries	Usage
a	Determines how glare conditions are handled when Interface field is **peer-slave**.
b	

Synch Source

The **Synch Source** field applies only for TN2185 boards.

Note:

For MM720 and MM722 bri media modules, the **Synch Source** field does not appear. For the MM720 and MM722, this parameter is configured using the gateway CLI.

Valid entries	Usage
y	When set to **y**, allows a TN2185 board displayed on the **Synchronization Plan** screen to be entered as the Primary or Secondary synchronization source, if at least one of the ports on that board has **Synch Source?** enabled. Only those ports marked **y** are candidates for clock synchronization with the far-end network.
n	

T3 Timer Length (sec)

Valid entries	Usage
1 to **127**	Tells the TE side how long to wait, in seconds, for an inactive Layer 1 to become active.

TEI

Valid entries	Usage
auto	TEI is assigned automatically by the network
0	TEI is fixed.

Time Fill

Valid entries	Usage
flags, ones	Enter the bit pattern sent on the D-channel between valid LAPD packets. Blank is not a valid entry. Default is **ones**.

Field descriptions for page 1

Figure 396: BRI Trunk screen (with a TN556B or TN2198 circuit pack)

```
change bri-trunk-board                                      Page 1 of 1
                        ISDN-BRI TRUNK CIRCUIT PACK

               Location: 01A09                    Name:  _______________
   Interface Companding: a-law_  DCP/Analog Bearer Capability: 3.1kHz


Port  Interface   Side  Cntry/Peer TEI  Time Invert
                        Protocol       Fill  Bits?
  1:  network__           12__     0__  flags   n
  2:  network__           etsi     0__  ones_   y
  3:  network__           2___     auto flags   y
  4:  peer-master  b      QSIG     0__  flags   n
  5:  peer-master  a      QSIG     auto ones_   n
  6:  __________          ____     0__  ones_   n
  7.  __________          ____     0__  ones_   n
  8:  __________          ____     0__  ones_   n
  9:  __________          ____     0__  ones_   n
 10:  __________          ____     0__  ones_   n
 11:  __________          ____     0__  ones_   n
 12:  __________          ____     0__  ones_   n
```

The following field descriptions are unique to the **ISDN-BRI Circuit Pack** screen with a TN556B or TN2198 circuit pack. The following fields do not display with a TN556B or TN2198 circuit pack:

- **T3 Timer Length (sec)**
- **Synch Source**
- **Layer 1 Stable**
- **Detect Slips**

Cntry/Peer Protocol

Tells call processing software which ISDN protocol standard is applied.

Valid entries	Usage
1 to **25**	When this field is **10**, **12**, or **13**, the **Protocol Version** field is equivalent to **b** on the **DS1 Circuit Pack** screen.
etsi	When this field is **etsi**, the **Protocol Version** field is equivalent to **b** on the **DS1 Circuit Pack** screen.
QSIG	When the **Interface** field is **peer-master**, this field must be **QSIG**.
blank	You cannot leave this field blank if the **Interface** field is set to a valid, non-blank value.

Interface

Valid entries	Usage
network	Tells call processing software whether a particular port is connected to a user/network or a peer interface. These entries are valid for the TN556B. You can enter **peer-master** only if the QSIG Basic Call Setup feature is enabled
peer-master	

Side

Valid entries	Usage
a	Determines how glare conditions are handled when **Interface** field is peer-slave. This field is not administrable when the Interface field is **network**.
b	

Field descriptions for page 2

Note:

If administering a TN2185 circuit pack, 8 ports appear; otherwise, 12 ports appear.

Figure 397: BRI Trunk screen - Page 2 (using a TN2185 circuit pack)

```
change bri-trunk-board                                    Page 2 of 2
                        ISDN-BRI TRUNK CIRCUIT PACK

 Port  Interwork  XID  Endpt    SPID      Endpt    SPID      Endpt  Max
       Message   Test? Init?               ID                ID   NCA TSC
   1:  PROGress   y      n   ____________  __   ____________  __     0___
   2:  ALERTing   y      y   908957200000  __   ____________  __     0___
   3:  PROGress   y      y   0001________  __   ____________  __     0___
   4:  PROGress   n      n   ____________  __   ____________  __     0___
   5:  PROGress   n      y   625761449___  01   ____________  __     0___
   6:  PROGress   n      n   ____________  __   ____________  __     0___
   7.  PROGress   n      n   ____________  __   ____________  __     0___
   8:  PROGress   n      n   ____________  __   ____________  __     0___

   Port Directory   Directory          Port Directory   Directory
          Number      Number                 Number      Number
     1:                                 5:
     2:                                 6:
     3:                                 7:
     4:                                 8:
```

Figure 398: BRI Trunk screen - Page 2 (using a TN2198/TN556B circuit pack)

```
change bri-trunk-board                                    Page 2 of 2
                        ISDN-BRI TRUNK CIRCUIT PACK

 Port  Interwork  XID  Endpt    SPID      Endpt                    Max
       Message   Test? Init?               ID                   NCA TSC
   1:  PROGress   n      y   _____________  __                     0___
   2:  ALERTing   n      y   _____________  __                     0___
   3:  PROGress   n      y   _____________  __                     0___
   4:  PROGress   n      y   _____________  __                     0___
   5:  PROGress   n      y   _____________  __                     0___
   6:  PROGress   n      y   _____________  __                     0___
   7.  PROGress   n      y   _____________  __                     0___
   8:  PROGress   n      y   _____________  __                     0___
   9:  PROGress   n      y   _____________  __                     0___
  10:  PROGress   n      y   _____________  __                     0___
  11:  PROGress   n      y   _____________  __                     0___
  12:  PROGress   n      y   _____________  __                     0___
```

Note:

You cannot change the **Endpt Init**, **SPID**, or **Endpt ID** port parameters unless that port is busied out or unadministered. It is possible to change all other fields on this page even if the corresponding port is active.

Note:

If the **Interface** field on page 1 contains a valid value when the screen is submitted, the contents of the fields on page 2 for that port are validated. If the **Interface** field is blank when the screen is submitted, the fields on this page for that port reset to their default values.

Directory Number

These 10-digit fields contain the directory numbers assigned to the interface, which it views as being allocated to 2 separate endpoints.

Valid entries	Usage
Any string of 1 to 10 digits	These fields must be administered in pairs. That is, if you enter a value in one field, you must enter a value in the other.

Endpt ID

A 2-digit field containing the Endpoint Identifier expected by the far end. Avaya Communication Manager blocks you from changing this field unless the port is busied out or unadministered.

Valid entries	Usage
00 to **62**	Leading zeroes considered significant and not ignored.

Endpt Init

Indicates whether the far end supports endpoint initialization. Avaya Communication Manager blocks you from changing this field unless the port is busied out or unadministered.

Valid entries	Usage
y	If set to **y**, the **SPID** field must *not* be blank. Avaya Communication Manager blocks you from changing this field and the **SPID** field unless that port is busied out or unadministered.
n	If set to **n**, the **SPID** and **Endpt ID** fields must be blank.

Interworking Message

This field determines what message Communication Manager sends when an incoming ISDN trunk call interworks (is routed over a non-ISDN trunk group).

Valid entries	Usage
PROGress	Normally select this value. PROGress asks the public network to cut through the B-channel and let the caller hear tones such as ringback or busy tone provided over the non-ISDN trunk.
ALERTing	ALERTing causes the public network in many countries to play ringback tone to the caller. Select this value only if the DS1 is connected to the public network, and it is determined that callers hear silence (rather than ringback or busy tone) when a call incoming over the DS1 interworks to a non-ISDN trunk.

Max NCA TSC

Valid entries	Usage
0 to **63**	This 2-digit field gives the maximum number of Non-Call-Associated Temporary Signaling Connections allowed on this BRI D-channel. This field's function is the same as the field with the same name on the **Signaling Group** screen.

Port

This is a display-only field. It shows the port number to which parameters administered on the row apply.

SPID

A 12-digit field containing the SPID expected by the far end. Avaya Communication Manager blocks you from changing this field unless the port is busied out or unadministered. The only protocol supported for SPID initialization is Bellcore (Country Code 1). Trunks will not be put in service if SPID installation is not successful.

Valid entries	Usage
Any string of 1 to 12 digits	Leading zeroes considered significant and not ignored

XID Test

Valid entries	Usage
y/n	Indicates whether the far end supports the Layer 2 XID test.

Trunk Member Administration

Administer BRI trunk members using the following scheme to address the individual B-channels:

- B-channel 1 uses the port address of the BRI Trunk Port.
- B-channel 2 uses the port address of B-channel 1 incremented by 16.

When adding a BRI trunk to an ISDN trunk-group, Avaya Communication Manager blocks you from administering a Signaling Group for that trunk member.

Avaya Communication Manager blocks you from administering a BRI trunk member if the port has not yet been administered on the **BRI Trunk** screen.

For example, administer the B-channels on a TN2185 circuit pack inserted in slot 01A10 as follows:

Port	B-channel 1	B-channel 2
1	01A1001	01A1017
2	01A1002	01A1018
3	01A1003	01A1019
4	01A1004	01A1020
5	01A1005	01A1021
6	01A1006	01A1022
7	01A1007	01A1023
8	01A1008	01A1024

Interactions

The **`add bri-trunk board PPCSS`** command is rejected if PPCSS identifies a TN556B circuit pack, and a port on that circuit pack has already been assigned to a station or data-module. If a TN556B circuit pack has been administered as a BRI trunk circuit pack, any port on that circuit pack is prevented from being assigned to a station or data-module.

Language Translations

Pre-translated messages are available for English, French, Italian, and Spanish to display messages on your display phones When you want to use a user-defined language, use the **Language Translation** screens to define the messages. Set up your messages using Roman letters, punctuation, digits and blank spaces. Diacritical marks are not supported.

All button names can be assigned a user-defined name. For more information, see Telephone feature Buttons on page 123.

Note:

If "user-defined" is entered for the display language on the **Station** screen or **Attendant** screen, and no messages are defined on these screens, a string of asterisks appears on all display messages. For information on administering Unicode languages, see Administering Unicode display on page 181.

In this section, the field descriptions are listed before the screens

English

This is a display-only field. It contains the English version of the message on the display.

Meaning of English term

This is a display-only field. It explains the abbreviated English message.

Translation

Enter the message you want to appear on the telephone display in place of the English message. Remember that a long message may be shortened on phones that display fewer than 32 characters.

Figure 399: Language Translations screen — AD programming

```
change display-messages ad-programming                          Page  1 of  x
                           LANGUAGE TRANSLATIONS

     English                              Translation

  1. Press button to program.         1.
**************************************
  2. Change program?                  2. **********************
  3. Yes = 1 No = 2                   3. ****************
  4. Enter number:                    4.
**************************************
  5. Press # to save.                 5. ****************
  6. Number saved.                    6.
**************************************
  7. Change label?                    7. **********************
  8. Enter label:                     8.
**************************************
  9. Press * to advance; # to save.   9. ******************************
 10. Press * to reenter; # to save.  10. ******************************
 11. Label saved. Hang up to update. 11.
**************************************
```

Figure 400: Language Translations screen — Auto-Wakeup-Dn-Dst (Page 1)

```
change display-messages auto-wakeup-dn-dst                      Page  1 of  x
                           LANGUAGE TRANSLATIONS

   1.     English:    AUTO WAKEUP - Ext:
      Translation: ********************:

   2.     English: WAKEUP ENTRY DENIED
      Translation: ************************

   3.     English: WAKEUP REQUEST CANCELED
      Translation: ************************************

   4.     English: WAKEUP REQUEST CONFIRMED
      Translation: ************************************

   5.     English: Wakeup Call
      Translation: ************************************

   6.     English:  Time:
      Translation: *****:
```

Figure 401: Language Translations screen — Auto-Wakeup-Dn-Dst (Page 2)

```
change display-messages auto-wakeup-dn-dst                    Page  2 of  x
                          LANGUAGE TRANSLATIONS

  7.     English:    DO NOT DIST - Ext:
    Translation: ********************:

  8.     English:     DO NOT DIST - Group:
    Translation: ***********************:

  9.     English: DO NOT DIST ENTRY DENIED
    Translation: ************************

 10.     English: THANK YOU - DO NOT DIST ENTRY CONFIRMED
    Translation: ****************************************

 11.     English: THANK YOU - DO NOT DIST REQUEST CANCELED
    Translation: ****************************************
```

Figure 402: Language Translations screen — Auto-Wakeup-Dn-Dst (Page 3)

```
change display-messages auto-wakeup-dn-dst                    Page  3 of  x
                          LANGUAGE TRANSLATIONS

 12.     English: INTERVAL FULL
    Translation: *************

 13.     English: NO PERMISSION
    Translation: *************

 14.     English: SYSTEM FULL
    Translation: *************

 15.     English: TOO SOON
    Translation: *************

 16.     English: INVALID EXTENSION - TRY AGAIN
    Translation: ****************************************

 17.     English: INVALID GROUP - TRY AGAIN
    Translation: ****************************************
```

Figure 403: Language Translations screen — Button Labels (page 1)

```
change display-messages button-labels                          Page 1 of x
                            LANGUAGE TRANSLATIONS

 English                            Translation

  1. Abr Mark                    1. **************
  2. Abr Pause                   2. **************
  3. Abr Program                 3. **************
  4. Abr Spec Char               4. **************
  5. Abr Stop                    5. **************
  6. Abr Suppress                6. **************
  7. AbRing                      7. *****
  8. Abr Wait                    8. **************
  9. Account                     9. **************
 10. AD                         10. ***********
 11. AddBusyInd                 11. **************
 12. AdmConnAlarm               12. **************
 13. AfterCall                  13. ***********
 14. Alert Agent                14. **************
```

Figure 404: Language Translations screen — Button Labels (page 2)

```
change display-messages button-labels                          Page 2 of x
                            LANGUAGE TRANSLATIONS

 English                            Translation

  1. Alternate FRL               1. **************
  2. ANI Request                 2. **************
  3. Assist                      3. ***********
  4. ASVN Halt                   4. **************
  5. AttQueueCall                5. **************
  6. AttQueueTime                6. **************
  7. Audix Record                7. **************
  8. Auto Callback               8. **************
  9. Auto Ckt Halt               9. **************
 10. AutoIC                     10. *******
 11. Auto In                    11. ***********
 12. AutoWakeAlarm              12. **************
 13. Auto Wakeup                13. **************
 14. AuxWork                    14. ********
 15. Busy                       15. *****
```

Figure 405: Language Translations screen — Button Labels (page 3)

```
change display-messages button-labels                          Page 3 of x
                            LANGUAGE TRANSLATIONS

 English                            Translation

  1. Button Ring                 1. **************
  2. Button View                 2. **************
  3. Caller Info                 3. **************
  4. CFrwd                       4. *****
  5. Call Park                   5. **************
  6. Call Pickup                 6. **************
  7. Call Time                   7. **************
  8. CAS Backup                  8. **************
  9. Cancel LWC                  9. **************
 10. CDR1 Fail                  10. **************
 11. CDR2 Fail                  11. **************
 12. CFBDA                      12. *****
 13. Check In                   13. **************
 14. Check Out                  14. **************
```

Figure 406: Language Translations screen — Button Labels (page 4)

```
change display-messages button-labels                          Page 4 of x
                            LANGUAGE TRANSLATIONS

 English                            Translation

  1. ClkOverride                 1. ************
  2. CO Line                     2. ***********
  3. Conf Display                3. **************
  4. Consult                     4. **************
  5. Cover Msg Ret               5. **************
  6. CovrCallBack                6. **************
  7. CPN Block                   7. **************
  8. CPN Unblock                 8. **************
  9. Crisis Alert                9. **************
 10. Data                       10. *****
 11. Delete Msg                 11. **************
 12. Dial Icom                  12. **********
 13. DID Remove                 13. **************
 14. DID View                   14. **************
```

Figure 407: Language Translations screen — Button Labels (page 5)

```
change display-messages button-labels                          Page 5 of x
                            LANGUAGE TRANSLATIONS

 English                            Translation

  1. Directory                    1. **************
  2. Dir Pickup                   2. **************
  3. Disp Charges                 3. **************
  4. DoNotDisturb                 4. **************
  5. Exclusion                    5. **************
  6. ExtDoNotDstrb                6. **************
  7. Far End Mute                 7. **************
  8. Flash                        8. **************
  9. FTC Alarm                    9. **************
 10. Goto Cover                  10. **************
 11. GrpPg                       11. *****
 12. GrpDoNotDstrb               12. **************
 13. Hunt NS                     13. ***********
 14. InCalID                     14. ********
```

Figure 408: Language Translation screen - Button Labels (page 6)

```
change display-messages button-labels                          Page 6 of x
                            LANGUAGE TRANSLATIONS

 English                            Translation

  1. Inspect                      1. **************
  2. IntAutoAnswer                2. **************
  3. License Error                3. **************
  4. Link Fail                    4. ************
  5. Lock LWC                     5. **************
  6. LSVN Halt                    6. **************
  7. Major Alarm                  7. **************
  8. Make Call                    8. **************
  9. ManOverid                    9. *************
 10. Manual In                   10. ***********
 11. MCT Activate                11. **************
 12. MCT Control                 12. **************
 13. Mj/Mn Alarm                 13. **************
```

Figure 409: Language Translations screen — Button Labels (page 7)

```
change display-messages button-labels                          Page 7 of x
                            LANGUAGE TRANSLATIONS

 English                            Translation

  1. MM Basic                    1. **************
  2. MM Call                     2. **************
  3. MM Call Fwd                 3. **************
  4. MM Data Conf                4. **************
  5. MM Mult Nbr                 5. **************
  6. MM PC Audio                 6. **************
  7. Msg                         7. *****
  8. Msg Retrieve                8. **************
  9. MsgW                        9. *****
 10. MsgWaitAct                 10. **************
 11. MsgWaitDeact               11. **************
 12. MST Debug                  12. **************
 13. Next                       13. **************
 14. Night Service              14. **************
```

Figure 410: Language Translations screen — Button Labels (page 8)

```
change display-messages button-labels                          Page 8 of x
                            LANGUAGE TRANSLATIONS

 English                            Translation

  1. NoAnsAltr                   1. **********
  2. OffBoardAlarm               2. **************
  3. PAGE1 Alarm                 3. **************
  4. PAGE2 Alarm                 4. **************
  5. PMS Failure                 5. **************
  6. PMS Ptr Alarm               6. **************
  7. Posted MSGs                 7. **************
  8. Priority Call               8. **************
  9. QueueCall                   9. **********
 10. QueueTime                  10. **********
 11. Release                    11. **************
 12. RemBusyInd                 12. **************
 13. ResetAlert                 13. **************
 14. Ringer Off                 14. **************
 15. Ring Stat                  15. **************
```

Figure 411: Language Translations screen — Button Labels (page 9)

```
change display-messages button-labels                          Page 9 of x
                            LANGUAGE TRANSLATIONS

 English                            Translation

  1. RSVN Halt                    1. *************
  2. SD                           2. **
  3. SendAllCalls                 3. *************
  4. Send TEG                     4. *************
  5. Service Obsrv                5. *****
  6. Sgnl                         6. *************
  7. SSVN Halt                    7. *************
  8. Start Bill                   8. *************
  9. Station Lock                 9. *************
 10. Stored Number               10. *************
 11. Store LWC                   11. *************
 12. Stroke Count                12. ************
 13. System Alarm                13. *************
 14. TermGroup                   14. **********
 15. Time/Date                   15. *************
 16. Timer                       16. *************
```

Figure 412: Language Translations screen — Button Labels (page 10)

```
change display-messages button-labels                         Page 10 of x
                            LANGUAGE TRANSLATIONS

 English                            Translation

  1. Toggle Swap  :               1. ************
  2. Trunk ID                     2. ***********
  3. Trunk Name                   3. *************
  4. Trunk NS                     4. *************
  5. UUI Info                     5. *************
  6. Verify                       6. *************
  7. VIP Check In                 7. *************
  8. VIP retry                    8. *************
  9. VIP Wakeup                   9. *************
 10. VOA Repeat                  10. *****
 11. VU Display                  11. *************
 12. WhisperAct                  12. **********
 13. WhisperAnbk                 13. *************
 14. WhisperOff                  14. *************
 15. Work Code                   15. *************
```

Figure 413: Language Translations screen — Call-Identifiers (Page 1)

```
change display-messages call identifiers                        Page  1 of  x
                            LANGUAGE TRANSLATIONS

    English              Meaning of English term                 Translated
    Term                                                         Term

 1. sa            ACD Supervisor Assistance                     1: **
 2. ac            Attendant Assistance Call                     2: **
 3. tc            Attendant Control Of A Trunk Group            3: **
 4. an            Attendant No Answer                           4: **
 5. pc            Attendant Personal Call                       5: **
 6. rc            Attendant Recall Call                         6: **
 7. rt            Attendant Return Call                         7: **
 8. sc            Attendant Serial Call                         8: **
 9. co            Controlled Outward Restriction                9: **
10. cs            Controlled Station To Station Restriction    10: **
11. ct            Controlled Termination Restriction           11: **
12. db            DID Find Busy Station With CO Tones          12: **
13. da            DID Recall Go To Attendant                   13: **
14. qf            Emergency Queue Full Redirection             14: **
15. hc            Held Call Timed Reminder                     15: **
```

Figure 414: Language Translations screen — Call-Identifiers (Page 2)

```
change display-messages call identifiers                        Page  2 of  x
                            LANGUAGE TRANSLATIONS

    English              Meaning of English term                 Translated
    Term                                                         Term

16. ic            Intercept                                    16: **
17. ip            Interposition Call                           17: **
18. ld            LDN Calls On DID Trunks                      18: **
19. so            Service Observing                            19: **
20. na            Unanswered Or Incomplete DID Call            20: **
21. ACB           Automatic Callback                           21: ********
22. callback      Callback Call                                22: ********
23. park          Call Park                                    23: ********
24. control       Control                                      24: ********
25. ICOM          Intercom Call                                25: ********
26. OTQ           Outgoing Trunk Queuing                       26: ********
27. priority      Priority Call                                27: ********
28. recall        Recall Call                                  28: ********
29. return        Return Call                                  29: ********
30. ARS           Automatic Route Selection                    30: ********
```

Figure 415: Language Translations screen — Call-Identifiers (Page 3)

```
change display-messages call identifiers                        Page  3 of  x
                            LANGUAGE TRANSLATIONS

    English              Meaning of English term                 Translated
    Term                                                         Term

 31. forward       Call Forwarding                               31: ********
 32. cover         Cover                                         32: ********
 33. DND           Do Not Disturb                                33: ********
 34. p             Call Pickup                                   34: *
 35. c             Cover All Calls                               35: *
 36. n             Night Station Service, Including No Answer    36: *
 37. B             All Calls Busy                                37: *
 38. f             Call Forwarding                               38: *
 39. b             Cover Busy                                    39: *
 40. d             Cover Don't Answer                            40: *
 41. s             Send All Calls                                41: *
 42. to            <calling party> to <called party>             42: **
 43. VDN           Vector Directory Number                       43: ***
 44. hunt          Station Hunting, Origination                  44: ********
 45. h             Station Hunting, Termination                  45: *
```

Figure 416: Language Translations screen — Call-Identifiers (Page 4)

```
change display-messages call identifiers                        Page  4 of  x
                            LANGUAGE TRANSLATIONS

    English         Meaning of English term                      Translated
    Term                                                         Term

 46. OPERATOR        Operator                                   46: ***************
 47. EXT             Extension                                  47: ***************
 48. OUTSIDE CALL    Outside Call                               48: ***************
 49. UNKNOWN NAME    Unknown Name                               49: ***************
 50. CONFERENCE      Conference                                 50: ************
 51. ringing         Ringing                                    51: ********
 52. busy            Busy                                       52: ********
 53. busy(I)         Busy With Intrusion Allowed                53: ********
 54. wait            Wait                                       54: *****
 55. (I)             Intrusion                                  55: ***
 56. Sta             Station                                    56: *********
 57. Trk             Trunk                                      57: *********
 58: offered         QSIG call offered to remote endpoint       58: ********
 59: cl              Controlled Toll Restriction                59: **
 60: vm              Call to Attendant Out of Voicemail         60: **
```

Figure 417: Language Translations screen — Date-Time (Page 1)

```
change display-messages date-time                              Page  1 of  x
                        LANGUAGE TRANSLATIONS

     English           Translation            English           Translation

  1. SUNDAY          1: ***********        11. APRIL          11: ***********
  2. MONDAY          2: ***********        12. MAY            12: ***********
  3. TUESDAY         3: ***********        13. JUNE           13: ***********
  4. WEDNESDAY       4: ***********        14. JULY           14: ***********
  5. THURSDAY        5: ***********        15. AUGUST         15: ***********
  6. FRIDAY          6: ***********        16. SEPTEMBER      16: ***********
  7. SATURDAY        7: ***********        17. OCTOBER        17: ***********
  8. JANUARY         8: ***********        18. NOVEMBER       18: ***********
  9. FEBRUARY        9: ***********        19. DECEMBER       19: ***********
 10. MARCH          10: ***********

 20.     English: SORRY, TIME UNAVAILABLE NOW
     Translation: ************************************
```

Figure 418: Language Translations screen — Leave-Word-Calling (Page 2)

```
change display-messages leave-word-calling                     Page  1 of  x
                          LANGUAGE TRANSLATIONS

   1.     English:              MESSAGES FOR
      Translation: ************************

   2.     English: WHOSE MESSAGES? (DIAL EXTENSION NUMBER)
      Translation: ************************************

   3.     English: END OF MESSAGES (NEXT TO REPEAT)
      Translation: ************************************

   4.     English: MESSAGES UNAVAILABLE - TRY LATER
      Translation: ************************************

   5.     English: MESSAGE RETRIEVAL DENIED
      Translation: ************************************

   6.     English: MESSAGE RETRIEVAL LOCKED
      Translation: ************************************
```

Figure 419: Language Translations screen — Leave-Word-Calling (Page 2)

```
change display-messages leave-word-calling                    Page  2 of  x
                          LANGUAGE TRANSLATIONS

   7.     English: NO MESSAGES
      Translation: ****************************************

   8.     English: IN PROGRESS
      Translation: ****************************************

   9.     English: DELETED
      Translation: ****************************************

  10.     English: GET DIAL TONE, PUSH Cover Msg Retrieval
      Translation: ****************************************

  11.     English: Message Center (AUDIX) CALL
      Translation: ****************************************

  12.     English: CANNOT BE DELETED - CALL MESSAGE CENTER
      Translation: ****************************************
```

Figure 420: Language Translations screen — Malicious-Call-Trace (Page 1)

```
change display-messages mailcious-call-trace                  Page  1 of  x
                          LANGUAGE TRANSLATIONS

   1.     English: MALICIOUS CALL TRACE REQUEST
      Translation: ****************************************

   2.     English: END OF TRACE INFORMATION
      Translation: ****************************************

  3.     English: original call redirected from:
     Translation: ****************************:

  4.     English:           voice recorder port:
     Translation: ****************************:

  5.     English:  MCT activated by:        for:
     Translation: *****************:       ****:
```

Figure 421: Language Translations screen — Malicious-Call-Trace (Page 2)

```
change display-messages mailcious-call-trace                    Page  2 of  x
                          LANGUAGE TRANSLATIONS


  6.     English:    party :                    (EXTENSION)
    Translation: ******** :       ************************

  7.     English:    party :                 (ISDN SID/CNI)
    Translation: ******** :              ****************

  8.     English:    party :                      (PORT ID)
    Translation: ******** :           *******************

  9.     English:    party :                 (ISDN PORT ID)
    Translation: ******** :           *******************
```

Figure 422: Language Translations screen — Miscellaneous-Features (Page 1)

```
change display-messages miscellaneous-features                  Page  1 of x
                          LANGUAGE TRANSLATIONS

    English                                  Translation

 1. ALL MADE BUSY                         1: ***************
 2. BRIDGED                               2: ***************
 3. DENIED                                3: ***************
 4. INVALID                               4: ***************
 5. NO MEMBER                             5: ***************
 6. OUT OF SERVICE                        6: ***************
 7. RESTRICTED                            7: ***************
 8. TERMINATED                            8: ***************
 9. TRUNK SEIZED                          9: ***************
10. VERIFIED                             10: ***************
11. CDR OVERLOAD                         11: ***************
12. ANSWERED BY                          12: ***************
13. CALL FROM                            13: ***************
14. Skills                               14: ***************
```

Figure 423: Language Translations screen — Miscellaneous Features (Page 2)

```
change display-messages miscellaneous-features                  Page  2 of x
                              LANGUAGE TRANSLATIONS
    English              Meaning of English term              Translated
    Term                                                      Term

 15. TOLL          Toll                                    15: ****
 16. FULL          Full                                    16: ****
 17. NONE          None                                    17: ****
 18. ORIG          Origination                             18: ****
 19. OTWD          Outward                                 19: ****
 20. CALL          <call> This Number                      20: ****
 21. INTL          International                           21: ****
 22. Info          Information                             22: *****
 23. p             Primary                                 23: *
 24. s             Secondary                               24: *
 25. m             Mark                                    25: *
 26. p             Pause                                   26: *
 27. s             Suppress                                27: *
 28. w             Wait For A Specified Time               28: *
 29. W             Wait For Off-Premise Dial Tone          29: *
```

Figure 424: Language Translations screen — Miscellaneous-Features (Page 3)

```
change display-messages miscellaneous-features                  Page 3 of x

                              LANGUAGE TRANSLATIONS

  30.     English: You have adjunct messages
      Translation: ****************************************

  31.     English: Login Violation
      Translation: ****************************************

  32.     English: Barrier Code Violation
      Translation: ****************************************

  33.     English: Authorization Code Violation
      Translation: ****************************************

  34.     English: DIRECTORY - PLEASE ENTER NAME
      Translation: ****************************************

  35.     English: DIRECTORY UNAVAILABLE - TRY LATER
      Translation: ****************************************
```

Figure 425: Language Translations screen — Miscellaneous-Features (Page 4)

```
change display-messages miscellaneous-features                  Page 4 of x

                            LANGUAGE TRANSLATIONS

  36.     English: NO MATCH - TRY AGAIN
      Translation: ************************************

  37.     English: NO NUMBER STORED
      Translation: ************************************

  38.     English: TRY AGAIN
      Translation: ************************************

  39.     English: Ext          in EMRG Q
      Translation: ***          **********

  40.     English:       HUNT GROUP   NOT ADMINISTERED
      Translation: ***************   ********************

  41.     English: Q-time       calls
      Translation: ******       *****
```

Figure 426: Language Translations screen — Miscellaneous-Features (Page 5)

```
change display-messages miscellaneous-features                  Page 5 of x
                             LANGUAGE TRANSLATIONS

   42.     English: Add Skill: Enter number, then # sign
       Translation: ************************************

   43.     English: Remove Skill: Enter number, then # sign
       Translation: ************************************

   44.     English: Enter Skill Level, then # sign
       Translation: ************************************

   45.     English: Enter Agent LoginID
       Translation: ************************************

   46.     English: Call Type
       Translation: ***************

   47.     English: Call Charge
       Translation: ********************
```

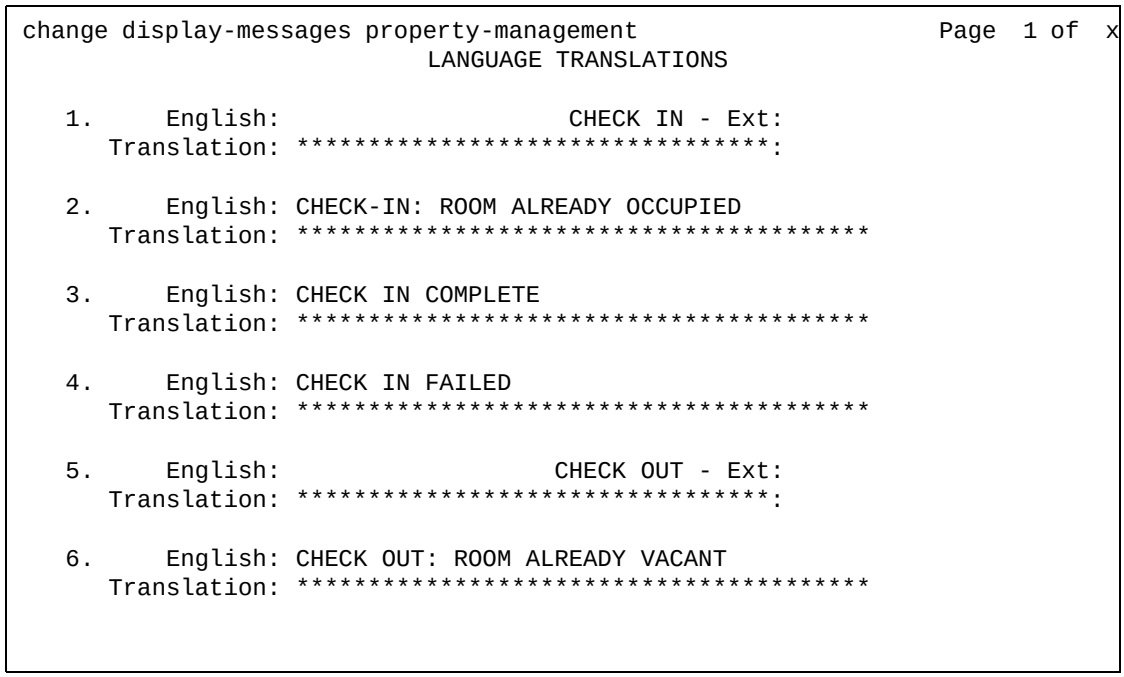
Figure 427: Language Translations screen — Miscellaneous-Features (Page 6)

```
change display-messages miscellaneous-features                Page 6 of x
                              LANGUAGE TRANSLATIONS

   48.     English: Station Security Code Violation
       Translation: ****************************************
   49.     English: ENTER REASON CODE
       Translation: ****************************************
   50.     English: Whisper From
       Translation: ******************
   51.     English: Whisper To
       Translation: ******************
   52.     English: Press button to remove.
       Translation: ****************************************
   53.     English: Press # to remove.
       Translation: ******************
```

Figure 428: Language Translations screen — Miscellaneous-Features (Page 7)

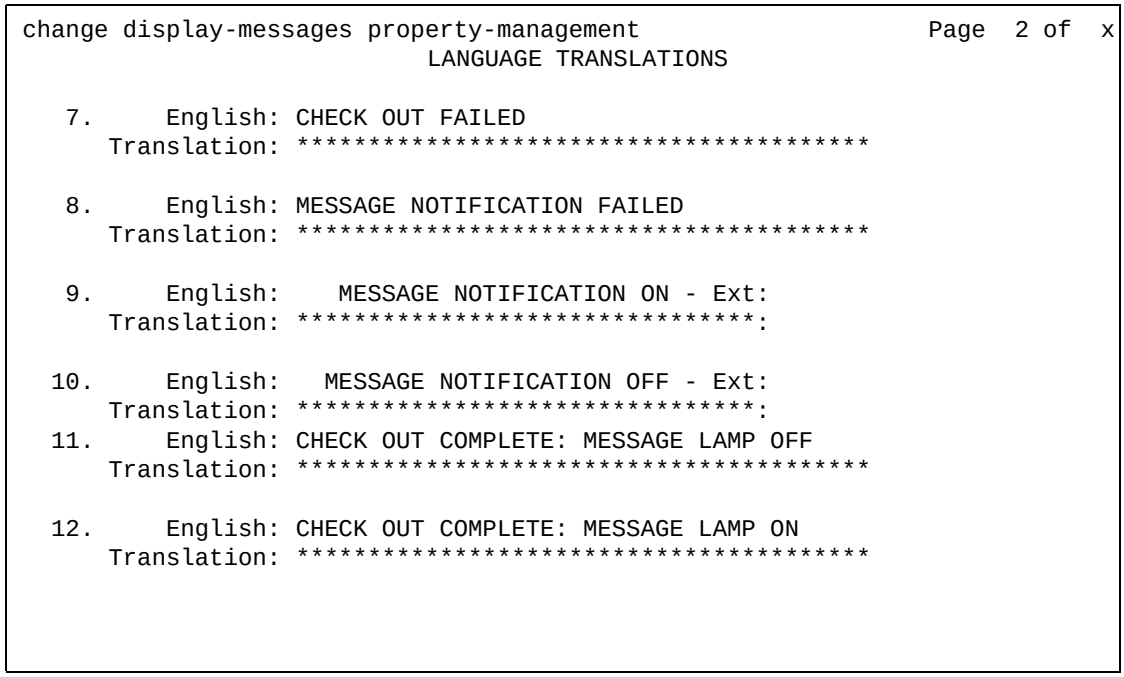
```
change display-messages miscellaneous-features                Page 7 of x
                              LANGUAGE TRANSLATIONS

   54.     English: Button removed.
       Translation: ****************************************
   55.     English: Ringer On
       Translation: ****************************************
   56.     English: Ringer Off
       Translation: ****************************************
   57.     English: Ringer Abbreviated
       Translation: ****************************************
   58.     English: Ringer Delayed
       Translation: ****************************************
   59.     English: Select a held party's line to talk.
       Translation: ****************************************
```

Figure 429: Language Translations screen — Property-Management (Page 1)



```
change display-messages property-management                    Page  1 of  x
                          LANGUAGE TRANSLATIONS

   1.     English:                    CHECK IN - Ext:
      Translation: ********************************:

   2.     English: CHECK-IN: ROOM ALREADY OCCUPIED
      Translation: ****************************************

   3.     English: CHECK IN COMPLETE
      Translation: ****************************************

   4.     English: CHECK IN FAILED
      Translation: ****************************************

   5.     English:                   CHECK OUT - Ext:
      Translation: ********************************:

   6.     English: CHECK OUT: ROOM ALREADY VACANT
      Translation: ****************************************
```

Figure 430: Language Translations screen — Property-Management (Page 2)



```
change display-messages property-management                    Page  2 of  x
                          LANGUAGE TRANSLATIONS

   7.     English: CHECK OUT FAILED
      Translation: ****************************************

   8.     English: MESSAGE NOTIFICATION FAILED
      Translation: ****************************************

   9.     English:    MESSAGE NOTIFICATION ON - Ext:
      Translation: ********************************:

  10.     English:   MESSAGE NOTIFICATION OFF - Ext:
      Translation: ********************************:
  11.     English: CHECK OUT COMPLETE: MESSAGE LAMP OFF
      Translation: ****************************************

  12.     English: CHECK OUT COMPLETE: MESSAGE LAMP ON
      Translation: ****************************************
```

Figure 431: Language Translations screen — Property-Management (Page 3)

```
change display-messages property-management                    Page  3 of  x
                           LANGUAGE TRANSLATIONS


 13.     English: MESSAGE LAMP ON
    Translation: ************************************

 14.     English: MESSAGE LAMP OFF
    Translation: ************************************
 15.     English: Occupied Rooms
    Translation: ************************************
 16.     English: Enter Room Status (1-6)
    Translation: ************************************
 17.     English: State, Enter number from 1 - 6
    Translation: ************************************
 18.     English: DID
    Translation: ********************
```

Figure 432: Language Translations screen — Property-Management (Page 4)

```
change display-messages property-management                    Page 4 of x
                           LANGUAGE TRANSLATIONS
   19.     English: DID VIEW: EXT?
       Translation: ************************************
   20.     English: DID=                   CHANGE?
       Translation: ****                    **********************
   21.     English: DID VIEW DONE
       Translation: ************************************
   22.     English: NO DID AVAILABLE
       Translation: ************************************
   23.     English: CHECK IN COMPLETE, DID=
       Translation: ********************************
```

Figure 433: Language Translations screen — Property-Management (Page 5)

```
change display-messages property-management                    Page 5 of x

                              LANGUAGE TRANSLATIONS
   24.     English: REMOVE(1), REPLACE(2)?
       Translation: ****************************************
   25.     English: DID REMOVED
       Translation: ****************************************
   26.     English: VIP CHECK IN - Ext:
       Translation: ****************************************
   27.     English: SPECIFY VID DID:
       Translation: ****************************************
   28.     English: CHECK IN COMPLETE, INVALID DID
       Translation: ****************************************
   29.     English: CHECK IN COMPLETE, DID UNAVAILABLE
       Translation: ****************************************
```

Figure 434: Language Translations screen — Property-Management (Page 6)

```
change display-messages property-management                    Page 6 of x

                              LANGUAGE TRANSLATIONS
   30.     English: DID REMOVE - Ext:
       Translation: ****************************************
   31.     English: DID CHANGED
       Translation: ****************************************
   32.     English: AUTOMATIC ASSIGN(1), SELECT(2)?
       Translation: ****************************************
   33.     English: DID UNAVAILABLE
       Translation: ****************************************
```

Figure 435: Language Translations screen — Self-Administration (page 1)

```
hange display-messages self-administration                    Page 1 of x

                              LANGUAGE TRANSLATIONS

 English                            Translation

  1. SECURITY CODE:                  1. **************
  2. INCORRECT SECURITY CODE         2. ************************
  3. SELECT FEATURE                  3. ************************
  4. EXTENSION:                      4. ******************
  5. OPTIONAL EXTENSION:             5. ******************
  6. TEL NUM:                        6. ********
  7. PRESS BUTTON TO PROGRAM         7. ************************
  8. BUTTON PROGRAMMED!              8. ************************
  9. INCORRECT BUTTON                9. ************************
 10. BUTTON LIMIT REACHED           10. ************************
 11. BUSY, TRY AGAIN LATER          11. ************************
 12. YOUR PHONE IS BUSY             12. ************************
 13. SECURITY CODE NEEDED           13. ************************
 14. BUTTON NOT CHANGED             14. ************************
```

Figure 436: Language Translations screen — Self-Administration (page 2)

```
change display-messages self-administration                   Page 2 of x

                              LANGUAGE TRANSLATIONS

 English     Translation    English                 Translation

  1. Acct    *****          CDR Account Code        ************************
  2. AutoD   *****          Automatic Dialing       ************************
  3. CFrwd   *****          Call Forwarding         ************************
  4. CPark   *****          Call Park               ************************
  5. CPkUp   *****          Call Pickup             ************************
  6. DPkUp   *****          Directed Call Pickup    ************************
  7. GrpPg   *****          Group Paging            ************************
  8. SAC     *****          Send All Calls          ************************
  9. Swap    *****          Conf/Trans Toggle-Swap  ************************
 10. WspPg   *****          Activate whisper Page   ************************
 11. WspAn   *****          Answerback for Whisper  ************************
 12. WsOff   *****          Whisper Page Off        ************************
 13. Blank   *****          Blank Button            ************************
```

Figure 437: Language Translations screen — Self-Administration (page 3)

```
change display-messages self-administration                    Page 3 of x

                              LANGUAGE TRANSLATIONS

 English                            Translation

  1. Whisper Page Off 1. *****
  2. Blank Button     2. *****
  3. Done             3. *****
  4. Cont             4. **********
  5. Expl?            5. *****
  6. ShortMode?       6. *****
  7. Next             7. *****
  8. Selct            8. *****
  9. Clear            9. *****
 10. Cancel          10. **********
 11. Delete          11. *****
 12. Replace         12. *****
 13. Bksp            13. *****
```

Figure 438: Language Translations screen — View-buttons (page 1)

```
change display-messages view-buttons                           Page 1 of x
                              LANGUAGE TRANSLATIONS

 English                            Translation

  1. Analog Bridge Appearance        1. ************************
  2. Abbreviated Dial Program        2. ************************
  3. Abbrev Dial Character           3. ************************
  4. Abbreviated Dial                4. ************************
  5. Abrv/Delayed Ring Change        5. ************************
  6. Auto Circuit Assurance          6. ************************
  7. Admin Connection Alarm          7. ************************
  8. CDR Account Code                8. ************************
  9. TSA Administration Mode         9. ************************
 10. ACD After Call Work            10. ************************
 11. ACD Change Alert               11. ************************
 12. Alternate FRL                  12. ************************
 13. Supervisor Assist              13. ************************
 14. SVN Auth Code Halt             14. ************************
 15. Attendant Queue Calls          15. ************************
 16. Attendant Queue Time           16. ************************
```

Figure 439: Language Translations screen — View-buttons (page 2)

```
change display-messages view-buttons                          Page 2 of x
                            LANGUAGE TRANSLATIONS

 English                          Translation

  1. Auto Message Waiting          1. ************************
  2. Automatic Call Back           2. ************************
  3. Automatic Dialing             3. ************************
  4. Automatic Intercom            4. ************************
  5. Auto-In Work Mode             5. ************************
  6. Automatic Wakeup              6. ************************
  7. Auxiliary Work Mode           7. ************************
  8. Bridged Appearance            8. ************************
  9. Busy Indicator                9. ************************
 10. Call Appearance              10. ************************
 11. Call Displayed Number        11. ************************
 12. Call Forwarding              12. ************************
 13. Call Park                    13. ************************
 14. Call Pickup                  14. ************************
 15. Caller Information           15. ************************
 16. CAS (Branch) Backup          16. ************************
```

Figure 440: Language Translations screen — View-buttons (page 3)

```
change display-messages view-buttons                          Page 3 of x
                            LANGUAGE TRANSLATIONS

 English                          Translation

  1. CDR 1st Printer Alarm         1. ************************
  2. CDR 2nd Printer Alarm         2. ************************
  3. Check In Hotel/Motel          3. ************************
  4. Check Out Hotel/Motel         4. ************************
  5. Call Forwarding Busy/DA       5. ************************
  6. Clocked Override              6. ************************
  7. Consult/Return                7. ************************
  8. Coverage Callback             8. ************************
  9. Cover Message Retrieve        9. ************************
 10. Data Extension               10. ************************
 11. Time of Day/Date Display     11. ************************
 12. Delete LWC Message           12. ************************
 13. Dial Intercom                13. ************************
 14. Integrated Directory         14. ************************
 15. Directed Call Pickup         15. ************************
 16. Normal/Local Mode Toggle     16. ************************
```

Figure 441: Language Translations screen — View-buttons (page 4)

```
change display-messages view-buttons                          Page 4 of x
                            LANGUAGE TRANSLATIONS

 English                          Translation

  1. Do Not Disturb                1. ************************
  2. Drop                          2. ************************
  3. Off-Board DS1 Alarm           3. ************************
  4. Manual Exclusion              4. ************************
  5. Do Not Disturb Extension      5. ************************
  6. Flash                         6. ************************
  7. Go To Cover                   7. ************************
  8. Do Not Disturb Group          8. ************************
  9. Group Paging                  9. ************************
 10. Headset On/Off               10. ************************
 11. Hunt Night Service           11. ************************
 12. Coverage Call Identify       12. ************************
 13. Inspect Call Appearance      13. ************************
 14. Internal Auto Answer         14. ************************
 15. Last Number Dialed           15. ************************
 16. Link Failure Alarm           16. ************************
```

Figure 442: Language Translations screen — View-buttons (page 5)

```
change display-messages view-buttons                          Page 5 of x
                            LANGUAGE TRANSLATIONS

 English                          Translation

  1. Login Security Violation      1. ************************
  2. Cancel Leave Word Call        2. ************************
  3. Lockout Leave Word Call       3. ************************
  4. Leave Word Call Message       4. ************************
  5. Major Hardware Alarm          5. ************************
  6. Manual Message Waiting        6. ************************
  7. Manual Override               7. ************************
  8. ACD Manual-In                 8. ************************
  9. MCT Call Trace Activate       9. ************************
 10. MCT Call Trace Control       10. ************************
 11. Major/Minor Alarm            11. ************************
 12. Message Retrieve             12. ************************
 13. Message Waiting On           13. ************************
 14. Message Waiting Off          14. ************************
 15. Next                         15. ************************
 16. Night Service Activate       16. ************************
```

Figure 443: Language Translations screen — View-buttons (page 6)

```
change display-messages view-buttons                          Page 6 of x
                            LANGUAGE TRANSLATIONS

 English                          Translation

  1. Redirect On No Answer         1. ************************
  2. Normal Display Mode           2. ************************
  3. Personal CO Line              3. ************************
  4. Property Management Fail      4. ************************
  5. Wakeup Printer Alarm          5. ************************
  6. Print Messages                6. ************************
  7. Priority Calling              7. ************************
  8. PMS Printer Alarm             8. ************************
  9. System Printer Alarm          9. ************************
 10. Number of Queued Calls       10. ************************
 11. Oldest Queued Time           11. ************************
 12. ACD Release                  12. ************************
 13. Ringer Cut-Off               13. ************************
 14. System Reset Alert           14. ************************
 15. Remote Access Violation      15. ************************
 16. Scroll Mode                  16. ************************
```

Figure 444: Language Translations screen — View-buttons (page 7)

```
change display-messages view-buttons                          Page 7 of x
                            LANGUAGE TRANSLATIONS

 English                          Translation

  1. Send All Calls                1. ************************
  2. TEG Send All Calls            2. ************************
  3. Service Observing             3. ************************
  4. Manual Signalling             4. ************************
  5. Station Security Call         5. ************************
  6. Stored Number Display         6. ************************
  7. Stroke Counts                 7. ************************
  8. Term Extension Group          8. ************************
  9. Timer                         9. ************************
 10. Facility Test Call Alarm     10. ************************
 11. Trunk ID                     11. ************************
 12. Trunk Name                   12. ************************
 13. Trunk Night Service          13. ************************
 14. UUI Info                     14. ************************
 15. Busy Verification            15. ************************
 16. VDN of Origin Announce       16. ************************
```

Figure 445: Language Translations screen — View-buttons (page 8)

```
change display-messages view-buttons                          Page 8 of x
                            LANGUAGE TRANSLATIONS

 English                           Translation

  1. Vu-Stats Displays              1. ************************
  2. Activate Whisper Page          2. ************************
  3. Whisper Page Answerback        3. ************************
  4. Whisper Page Off               4. ************************
  5. Call Work Code                 5. ************************
  6. Unassigned Button              6. ************************
  7. View Button                    7. ************************
  8. Call Timer                     8. ************************
  9. Add Busy Indicator             9. ************************
 10. Remove Busy Indicator         10. ************************
 11. VIP Wakeup                    11. ************************
 12. VIP Retry                     12. ************************
 13. Crisis Alert                  13. ************************
 14. DID View                      14. ************************
 15. DID Remove                    15. ************************
 16. VIP Check In                  16. ************************
```

Figure 446: Language Translations screen — View-buttons (page 9)

```
change display-messages view-buttons                          Page   9 of x
                            LANGUAGE TRANSLATIONS

 English                           Translation

  1.  Station Lock                  1. ************************
  2.  License Error                 2. ************************
  3.  Conference Display            3. ************************
  4.  Conf/Trans Toggle-Swap        4. ************************
  5.  Far End Mute                  5. ************************
  6.  Paging 1st Link Alarm         6. ************************
  7.  Paging 2nd Link Alarm         7. ************************
  8.  EC500                         8. ************************
  9.  No Hold Conference            9. ************************
 10.  Posted Messages              10. ************************
 11.  Audix Recording              11. ************************
```

Figure 447: Language Translations screen — Vustats

```
change display-messages vustats                                  Page   1 of 1

                              LANGUAGE TRANSLATIONS

 English                             Translation

  1.  FORMAT                          1. *******

  2.  NOT DEFINED                     2. ****************************
  3.  DOES NOT ALLOW OR REQUIRE ID    3. ****************************

  4.  AGENT                           4. ************
  5.  SPLIT/SKILL                     5. ************
  6.  TRUNK GROUP                     6. ************
  7.  VDN                             7. ************

  8. NOT ADMINISTERED                 8. *****************
  9. NOT MEASURED                     9. *****************

 10. AGENT NOT LOGGED IN             10.
**************************************
```

Figure 448: Language Translations screen — Softkey-Labels

```
change display-messages softkey-labels                Page   1 of   1
                          LANGUAGE TRANSLATIONS
English      Translation   English      Translation   English      Translation
 1. Acct       1. *****     17. Drop     17. *****     33. RmBsy    33. *****
 2.  AD        2. *****     18. Excl     18. *****     34. RngOf    34. *****
 3. AdBsy      3. *****     19. FMute    19. *****     35.  SAC     35. *****
 4. Admin      4. *****     20. GrpPg    20. *****     36. SFunc    36. *****
 5. AutCB      5. *****     21. HFAns    21. *****     37. Spres    37. *****
 6. BtnVu      6. *****     22. IAuto    22. *****     38. Stats    38. *****
 7. CFrwd      7. *****     23. IDial    23. *****     39. Stop     39. *****
 8. CnfDs      8. *****     24. Inspt    24. *****     40. Swap     40. *****
 9. CnLWC      9. *****     25. Last     25. *****     41. Timer    41. *****
10. Cnslt     10. *****     26.  LWC     26. *****     42. TmDay    42. *****
11. Count     11. *****     27. Mark     27. *****     43. View     43. *****
12. CPark     12. *****     28. NHCnf    28. *****     44. Wait     44. *****
13. CPkUp     13. *****     29. Pause    29. *****     45. WspAn    45. *****
14. CTime     14. *****     30. PCall    30. *****     46. WspPg    46. *****
15.  Dir      15. *****     31. PoMSG    31. *****
16. DPkUp     16. *****     32. Prog     32. *****
```

In order to provide unique labeling for abbreviated dialing button types for softkey-labels, Communication Manager replaces the last two characters with digits for the 12-key 8400 and 15-key 8434D phones.

On the softkey label **Language Translation** screen, the digits following the "**AD**" are derived from the button position. If the first button is an AD button, then it is **AD1** and the fifteenth button is **AD15**. All the AD buttons between 1 and 15 have the position number appended to "**AD**."

Figure 449: Language Translations screen — Time-Of-Day-Routing

```
change display-messages time-of-day-routing                    Page  1 of  x
                          LANGUAGE TRANSLATIONS

   1.     English: ENTER ACTIVATION ROUTE PLAN, DAY & TIME
      Translation: ****************************************

   2.     English: ENTER DEACTIVATION DAY AND TIME
      Translation: ****************************************

   3.     English:    OLD ROUTE PLAN:      ENTER NEW PLAN:
      Translation: ****************:   ****************:

   4.     English:    OLD ROUTE PLAN:         NEW PLAN:
      Translation: ****************:   **************:

   5.     English: ROUTE PLAN:    FOR       ACT-TIME:
      Translation: **********:   ****     **********:

   6.     English: ROUTE PLAN:    FOR     DEACT-TIME:
      Translation: **********:   ****     **********:
```

Figure 450: Language Translations Transfer-conference screen (page 1)

```
change display-messages transfer-conference                    Page 1 of x
                        LANGUAGE TRANSLATIONS

 1.     English: Transfer completed.
   Translation: ******************************

 2.     English: Call next party.
   Translation: ************************************

 3.     English: Press conference to add party.
   Translation: ************************************

 4.     English: ^-party conference in progress.
   Translation: ************************************

 5.     English: Conference canceled.
   Translation: ************************************

 6.     English: Select line ^ to cancel or another line.
   Translation: ************************************
```

Note:

For Messages 4, 6, 12, you manually must change "~" to "^" in your user-defined language. The software will not update automatically.

Message 4

The character "^ "is a place holder.

English Text	Replacement Info
^-party conference in progress	"^" is replaced with the number of parties currently on the conference call.

Message 6

The character "^ "is a place holder.

English Text	Replacement Info
Select line ^ to cancel or another line.	" ^ " is replaced with the letter of the line that is on soft hold.

Figure 451: Language Translations Transfer-conference screen (page 2)

```
change display-messages transfer-conference           Page 2 of 4
                         LANGUAGE TRANSLATIONS

 7.     English: Dial number.
    Translation: ************************************

 8.     English: Press transfer to complete.
    Translation: ************************************

 9.     English: Hang-up to complete transfer.
    Translation: ************************************

10.     English: Dial number or select held party.
    Translation: ************************************

11.     English: Select held party to conference.
    Translation: ************************************

12.     English: Select line ^ to add party.
    Translation: ************************************
```

Figure 452: Language Translations Transfer-conference screen (page 3)

```
change display-messages transfer-conference           Page 3 of 4
                         LANGUAGE TRANSLATIONS

13.     English: Select alerting line to answer call.
    Translation: ************************************

14.     English: Transfer canceled.
    Translation: ************************************

15.     English: Connecting to ^.
    Translation: ************************************

16.     English: Called party ^is busy.
    Translation: ************************************

17.     English: Invalid number dialed -
    Translation: **************************

18.     English: Party ^ is not available.
    Translation: **************************
```

Message 15, 16, 18

The character "^ "is a place holder.

English Text	Replacement Info
Select line ^ to add party.	" ^ " is replaced with the letter of the line that is on soft hold.

Figure 453: Language Translations Transfer-conference screen (page 4)

```
change display-messages transfer-conference           Page 4 of 4
                         LANGUAGE TRANSLATIONS

19.     English: Mute.
    Translation: ****
```

Message 12

The character "^ "is a place holder.

English Text	Replacement Info
Select line ^ to add party.	" ^ " is replaced with the letter of the line that is on soft hold.

Figure 454: Language Translations screen — VuStats

```
change display-messages vustats                              Page  1 of  1
                           LANGUAGE TRANSLATIONS

       English                           Translations

  1.   FORMAT                        1.   *******
  2.   NOT DEFINED                   2.   *******
  3.   DOES NOT ALLOW OR REQUIRE ID   3.   *******

  4.   AGENT                         4.   ****************************
  5.   SPLIT/SKILL                   5.   ****************************
  6.   TRUNK GROUP                   6.   ****************************
  7.   VND                           7.   ****************************

  8.   NOT ADMINISTERED              8.   ****************
  9.   NOT MEASURED                  9.   ****************

 10.   AGENT NOT LOGGED IN          10.    ************************
```

Listed Directory Numbers

Allows Direct Inward Dialing (DID) numbers to be treated as public Listed Directory Numbers (LDNs). When one of these numbers is direct inward dialed, the calling party is routed to the attendant. The attendant display indicates a Listed Directory Number call and the name associated with the dialed extension.

The number of Listed Directory Numbers that can be assigned varies depending on system configuration. See the *Hardware Guide for Avaya Communication Manager* for maximum values.

Figure 455: Listed Directory Numbers screen

```
change listed-directory-number                              Page   1 of   2
                        LISTED DIRECTORY NUMBERS

                         Night Destination:

         Ext     Name                          TN
      1:
      2:                                       1
      3:                                       1
      4:                                       1
      5:                                       1
      6:                                       1
      7:                                       1
      8:                                       1
```

Ext

Valid entries	Usage
1 to **8** digits	Enter the extension number.

Name

Valid entries	Usage
Up to 27 alphanumeric characters	Enter a name used to identify the Listed Directory Number

Night Destination

Enter the valid assigned extension number that will receive calls to these numbers when Night Service is active.

Valid entries	Usage
0 to **9** (1 to 8 digits)	For DEFINITY CSI, SI, S8700 or S8710 IP-Connect. Enter a night service extension, a recorded announcement extension, a Vector Directory Number, an individual attendant extension, or a hunt group extension.

TN

Valid entries	Usage
1 to **100**	Enter the Tenant Partition number.

Local Survivable Processors

If the primary server has Local Survivable Processors (LSPs), you must enter the LSP node names on the **Local Survivable Processor** screen to enable the LSPs to get translations updates from the primary controller. Once the LSPs are successfully entered on the **Local Survivable Processor** screen, their status can be viewed with the **`display lsp`** command.

Figure 456: Local Survivable Processor screen

```
change lsp                                                  Page   1 of  16

                          LOCAL SURVIVABLE PROCESSOR
 Number    Name            IP Address         Service          Translations
                                                State            Updated
1         LSPNYC_________  128.256.173.101   in-service/idle   03:21 02/19/03
2         LSPCHI_________  128.256.113.213   out-of-service     04:32 02/20/03
3         LSPLAX_________ 156.256.114.214   in-service/idle    05:43 03/19/03
4         LSPPHL_________ 128.213.234.34    in-service/idle    03:54 12/19/03
5         LSPPHL2________ 128.256.213.12    in-service/idle    06:24 11/23/03
6         LSPPHL3________ 128.256.173.14    in-service/idle    07:13 05/19/02
7         LSPORD_________ 128.256.173.244   in-service/idle    13:25 02/19/03
8         LSPORD1________ 128.256.173.232   in-service/idle    23:39 12/19/03
9         LSPORD2________ 128.256.173.222   out-of-service     21:33 12/19/03
10        LSPORD3________ 128.256.173.111   out-of-service      12:27 05/19/03
11        LSPMIA_________ 128.123.173.333    out-of-service     22:18 10/19/03
12        LSPDET_________ 128.222.173.211   out-of-service     23:31 12/19/03
13        LSPEWJ_________ 128.134.173.233    in-service/busy    23:45 06/19/03
14        LSPBOS_________ 128.174.173.111   in-service/busy    13:11 08/19/03
15        LSPDCA_________ 128.237.173.222   in-service/busy    13:46 12/19/03
16        LSPDEN________ 128.256.173.313    in-service/busy    04:15 11/
```

Number

Valid entries	Usage
1 to **250**	The number of the LSP.

Name

Valid entries	Usage
values from **Node Names** screen.	Enter the node name for each LSP supported by the primary controller. Default is blank.

IP Address

Valid entries	Usage
up to 12 digits	Enter the IP address of the LSP.

Service State

Valid entries	Usage
Display only	Refers to whether the LSP is communicating with the primary controller. Default is **out of service**. The keywords used in this column and their meaning are as follows: **in-service/idle**: The LSP is registered and connected, but not in use **in-service/busy**: The LSP is registered, connected, and is in use. **out-of-service**: The LSP is not registered. **busied-out**: This value is not currently used.

Translations Updated

Valid entries	Usage
Display only	Date and time of the last translation update.

Locations

Use the **Locations** screen to provide daylight savings time displays to users, to set the area code for each location, and to administer different location information for each location. If the Multiple Locations feature is enabled, you can administer up to 250 location specifications, depending on the configuration of the server that is running Communication Manager. Otherwise, information for Location No.1 applies to all your locations.

Figure 457: Locations screen

```
change locations                                          Page   1 of   1
                                  LOCATIONS

                ARS Prefix 1 Required For 10-Digit NANP Calls? y

 Loc. Name             Timezone  Rule      NPA    ARS Attd Loc.   Pre- Proxy Sel.
 No                     Offset                  FAC FAC  Parms. fix  Rte. Pat.
 1.  Main             + 00:00    1         312
 2.  Denver-01______  - 01:00    1         303    ____ ____ __    ____ ________
 3.  Lincroft-01____  + 01:00    1         953    ____ ____ __    ____ ________
 xxx ________________ _ __:__   __        ___    ____ ____ __    ____ ________
 xxx ________________ _ __:__   __        ___    ____ ____ __    ____ ________
```

ARS FAC

This field is controlled by the **Multiple Locations** field on the **Optional Features** screen (use the **`system-parameters customer-options`** command). Administration of this field must follow the same rules that exist for administering an ARS code on the **Feature Access Code (FAC)** screen.

Valid entries	Usage
0 to **9**	Any valid FAC format is acceptable, up to four digits. Characters ***** or # are permitted, but only in the first position. Many locations are expected to share the same access code.

ARS Prefix 1 Required for 10-Digit NANP Calls?

Valid entries	Usage
y/n	Enter **y** when a 1 must be dialed before all 10-digit NANP calls.

Attd FAC

The **Attd FAC** field is controlled by the **Multiple Locations** field on the **Optional Features** screen (use the **`system-parameters customer-options`** command).

A user cannot adminster an Attendant FAC unless an Attendant Access Code has first been administered on either the **Dial Plan Analysis Table** screen or the **Feature Access Code (FAC)** screen.

Note:

Within a dial plan, **FAC/DAC** codes and extensions cannot both start with the same first digits.Either the **FAC/DAC** entries or the block of extensions must be changed to have a different first digit..

Valid entries	Usage
0 to **9**	.Values up to two digits are permitted. Characters ***** or **#** are not permitted. Many locations are expected to share the same access code.

Loc Number

Displays the location number (**1** to **250**). Corresponding entries in each row define the numbering plan and daylight savings rule for each location number.

Loc. Parms.

This field is an index to the corresponding **Location Parameters n** screens for a specific location. If Multinational Locations is activated, and you enter information into any other field on a location row, you must make an entry in the **Loc. Parms** field. If you don’t, an error message displays, and your IP telephones may not be usable.

Valid entries	Usage
1 to **25** or blank	Enter the number of the corresponding Location Parameter set for this location. Default is blank.

Name

Identifies the server running Communication Manager associated with each location number.

Valid entries	Usage
up to 15 alphanumeric characters	A name you use for the location. Names are easier to remember than location numbers.

NPA

Valid entries	Usage
0 to **9**	Enter the 3-digit numbering plan area code for each location.

Prefix

This field is used for prepending the leading digits for **Uniform Dial Plan Table** screen entries for calls that have, in the Insert Digits field, an **L*x*** value, where ***x*** is the number of leading digits of the Prefix digits to prepend for the location of an originating call. This field is controlled by the **Multiple Locations** field on the **Optional Features** screen (use the **`system-parameters customer-options`** command).

Valid entries	Usage
0 to **9**	Values from one to five digits (0 to 99999) are permitted.

Proxy Sel Rte Pat

The **Proxy Selection Route Pattern** field identifies the routing pattern that is used to get to the proxy server. This is the route pattern assigned on the **Route Pattern** screen.

Valid entries	Usage
1 to **999** or blank	Enter the number of the routing pattern to be used to get to the proxy server.

Rule

This field must be filled in for each administered location.

Valid entries	Usage
0	No Daylight Savings
1 to **15** or blank	Specifies the number for each Daylight-Savings Rule (set up on the **Daylight Savings Rule** screen) that is applied to this location.

Timezone Offset

Timezone Offset is actually 3 fields (+/-, hour, and minute) that specify how much time each location differs from the system time. This field must be completed for each administered location. Use +00:00 for the time zone offset for a single location media server.

Valid entries	Usage
+	Shows that the time set on this location is a certain amount of time ahead of the system time.
-	Shows that the time set on this location is a certain amount of time behind the system time.

Valid entries	Usage
0 to **23**	Shows the number of hours difference between this location and system time.

Valid entries	Usage
0 to **59**	Shows the number of minutes difference between this location and system time.

Location Parameters

The **Location Parameters** screen allows you to set or change certain administerable characteristics that determine part of a location's behavior. These include recall timing intervals and loss plans for 2-party and conference calls.

Figure 458: Location Parameters screen

```
change location-parameters 1                                  Page   1 of  4

                    LOCATION PARAMETERS 1

               Tone Generation Plan: 1
             Analog Ringing Cadence: 1
          Analog Line Transmission: 1
       DCP Terminal-parameters Plan: 1
               Country code for CDR: 1
                    Companding Mode: Mu-law

RECALL TIMING
                Flashhook Interval? _              Upper Bound (msec): 1000
          Disconnect Timing (msec): 150             Lower Bound (msec): 200


           Forward Disconnect Timer (msec): 600
                 MF Interdigit Timer (sec): 10
Outgoing Shuttle Exchange Cycle Timer (sec): 4
```

Field descriptions for page 1

Analog Ringing Cadence

The country code identifies the ringing cadence to be used by analog phones in the system

Valid entries	Usage
1 to **25**	See the Country code table at the beginning of the **System-Parameters Country-Options** screen description. For more information, see "Audible Ringing Patterns" in *Hardware Guide for Avaya Communication Manager,* 555-245-207.

Analog Line Transmission

The country code identifies the transmission and signaling parameters.

Valid entries	Usage
1 to **25**	See the Country code table at the beginning of the **System-Parameters Country-Options** screen description.

Companding Mode

Identifies the companding algorithm to be used by system hardware.

Valid entries	Usage
A-Law	Generally used outside the U.S.
Mu-law	Generally used in the U.S.

Country code for CDR

Appears only when the Multinational Locations feature is turned on in the license file.

Valid entries	Usage
1 to **999**	The value in this field corresponds to the country code to be used for Call Detail Recording information for a location. Default is 1

DCP Terminal-parameters Plan

Appears only when the Multinational Locations feature is turned on in the license file. The value in this field corresponds to the DCP terminal transmission parameters administered for location ***n*** on the **Terminal Parameters n** screens.

Valid entries	Usage
1 to **25**	Enter terminal-parameters plan number 1 to 25. Default is **1**.

Tone Generation Plan

Appears only when the Multinational Locations feature is turned on in the license file. The value in this field corresponds to the tone generation characteristics administered for location ***n*** on the **Tone Generation n** screens.

Valid entries	Usage
1 to **25**	Enter tone-generation plan number 1 to 25. Default is **1**.

RECALL TIMING

Disconnect Timing (msec)

Appears when the **Flashhook Interval** field is **n**.

Valid entries	Usage
80 to **1250** (in increments of 10).	An on-hook that lasts for a period of time less than this value will be ignored; greater than or equal to this value will be regarded as a disconnect.

Flashhook Interval

Valid entries	Usage
y	Enter **y** to indicate that a flashhook interval (recall window) is required. If a **y** is entered, **Upper Bound** and **Lower Bound** appear.
n	If **n** is entered, **Disconnect Timing** appears.

Forward Disconnect Timer (msec)

Valid entries	Usage
25 to **1500** (in increments of 25).	Specify the duration of a momentary disconnect sent by the server/switch to an analog station user when that user is the last party still off-hook on a call.

Lower Bound (msec)

The lower bound of the station-to-switch recall signal timing interval in milliseconds. Specifies the lower bound of the flashhook interval. Appears when the **Flashhook Interval** field is **y**.

Valid entries	Usage
80 to **1250** (in increments of 10).	Specify the lower bound of the flashhook interval.

MF Interdigit Timer (sec)

Applies only to multifrequency signaling trunks. Specify the maximum number of seconds Communication Manager will wait for the first forward signal (digit) to arrive, and for subsequent digits to arrive. Intercept returns to the calling party if this timer expires.

Valid entries	Usage
1 to **255**	This number must be less than the number of seconds entered in the short interdigit timer.

Outgoing Shuttle Exchange Cycle Timer (sec)

Appears when the **Incoming Call Type** field is **group-ii-mfc** or **non-group-ii-mfc** and the **Outgoing Call Type** field is **group-ii-mfc** or **none** on the **Multifrequency-Signaling-Related System Parameters** screen. This field applies only to multifrequency signaling calls made from Avaya Communication Manager.

Valid entries	Usage
1 to **25**	Enter the number of seconds to time an exchange cycle (starts when the far end requests a digit until Avaya Communication Manager sends the requested digit).

Upper Bound (msec)

Specifies the upper bound of the flashhook interval. Specifies the upper bound of the station-to-switch recall signal timing interval in milliseconds. Appears when the **Flashhook Interval** field is **y**.

Valid entries	Usage
80 to **1250** (in increments of 10).	A flash of 50 msec to 130 msec is always acceptable from a 2500-type set regardless of the setting of the Upper and Lower bounds and will be treated as the digit one.

Field descriptions for page 2

Figure 459: Loss Plans screen

```
change location-parameters N                                    Page   2 of  4

LOSS PLANS

           Inter-location Loss Group: 18

                  2 Party Loss Plan: 1                  Customize? n

                     Tone Loss Plan: 1                  Customize? n

End-to-End total loss (dB) in a n-party conference:
3: 15   4: 21   5: 26   6: 29                            Customize? n
```

2-Party Loss Plan/Tone Loss Plan

Provides the default values for digital loss plan and for n-party conference loss.

Valid entries	Usage
1 to **25**	See the Country code table at the beginning of the **System Parameters Country-Options** screen description. Note that different codes may have similar plans.

Customize

This field appears when the **Digital Loss Plan Modification** field is enabled on the **System-Parameters Customer-Options** screen. It enables customization on the corresponding loss plan table. For the **End-to-End total loss (dB) in a n-party conference** field, when **Customize** set to **y** (yes), the fields can be changed by the administrator. When set to **n**, the **End-to-End total loss (dB) in a n-party conference** fields are reset to the values that they would have had under the **2 Party Loss Plan** administered on page 3 of this screen. They also become display only.

Valid entries	Usage
y/n	Enables customization on the corresponding loss plan table.

End-to-End total loss (dB) in a n-party conference

Provides total loss for a conference call with the designated number of parties.

Note:

The End-to-End total loss for multi-party conference calls that is administered on this screen is not always applied to a specific call. For more information on how loss is applied to a multi-party conference call, see the "Loss Plans" feature description in the *Feature Description and Implementation* document.

Valid entries	Usage
0 to **99**	The higher the number listed for a call with a fixed display number of parties, the more loss Communication Manager adds into a conference call with that number of parties; therefore, the conference call is quieter.

Inter-location Loss Group

Appears only when the Multinational Locations feature is turned on in the license file. When inserting loss for a call, the server treats parties on the call who are in separate locations as if the location with the most parties were connected by an equal number of IP tie trunks as there are parties at other locations. The **Inter-location Loss Group** field specifies the digital loss group number that is used by these "virtual" IP tie trunks.

Valid entries	Usage
1 to **19**	Enter the digital loss group number to use on inter-location calls involving this location. Default is 18.

Field descriptions for page 3

Figure 460: 2 Party Loss Plan screen

change location-parameters Page 3 of 4

2 PARTY LOSS PLAN

	TO:	1	2	3	4	5	6	7	8	9	10	11	12	13	14	15	16	17	18	19
	1:	0	0	0	0	3	0	0	0	0	3	0	6	6	6	0	3	3	0	0
	2:	0	0	0	0	0	3	3	2	2	3	0	6	6	6	2	3	3	0	0
	3:	0	0	0	0	0	3	3	3	2	3	0	6	6	6	0	3	3	6	6
	4:	15	0	0	0	6	0	0	0	0	3	0	6	6	6	0	3	3	0	0
	5:	0	-3	-3	0	0	-3	-3	-3	-3	0	-3	3	0	0	-3	3	3	0	0
	6:	0	3	3	0	0	6	8	6	5	5	5	9	9	9	5	3	3	0	0
F	7:	0	3	3	0	0	8	8	6	5	5	5	9	9	9	5	3	3	0	0
R	8:	0	3	3	0	0	6	6	6	3	5	3	9	6	6	3	3	3	0	0
O	9:	0	2	2	0	0	5	5	3	0	0	2	3	3	3	9	3	3	0	0
M	10:	3	3	3	3	3	5	5	5	0	0	3	3	3	3	3	3	3	3	3
	11:	0	0	0	0	0	5	5	3	2	3	0	6	6	3	0	3	3	0	0
	12:	0	0	0	0	0	3	3	3	-3	-3	0	0	0	0	0	3	3	6	6
	13:	0	0	0	0	0	3	3	3	-3	-3	0	0	0	0	0	3	3	6	6
	14:	0	0	0	0	0	3	3	3	-3	-3	-3	0	0	0	0	3	3	6	6
	15:	0	2	0	0	0	5	5	3	0	3	0	6	6	6	0	3	3	0	0
	16:	3	3	3	3	3	3	3	3	3	3	3	3	3	3	3	3	3	3	3
	17:	3	3	3	3	3	3	3	3	3	3	3	3	3	3	3	3	3	3	3
	18:	0	0	0	0	3	0	0	0	0	3	0	6	6	6	0	3	3	0	0
	19:	0	0	0	0	0	3	3	2	2	3	0	6	6	6	2	3	3	0	0

FROM / TO

Display-only fields that identify the variable digital loss values.

Valid entries	Usage
-3 through **15**	An unsigned number is a decibel loss, while a number preceded with a minus sign is a decibel gain.

Field descriptions for page 4

Figure 461: Tone Loss Plan screen

```
change location-parameters                                        Page 4 of 4


                                TONE LOSS PLAN

                                   TO
            1   2   3   4   5   6   7   8   9  10  11  12  13  14  15  16  17  18  19
      Dial: 0   3   3   0   0   6   6   6   5   0   6   5   5   5   5   0   0   0   0
   Confirm: 0   0   0   0   0   0   0   0   0   0   0   0   0   0   0   0   0   0   0
   Reorder: 0   0   0   0   0   0   0   0   0   0   0   0   0   0   0   0   0   0   0
      Busy: 0   0   0   0   0   0   0   0   0   0   0   0   0   0   0   0   0   0   0
   Ringing: 0   0   0   0   0   0   0   0   0   0   0   0   0   0   0   0   0   0   0
 Spec Ring: 0   0   0   0   0   0   0   0   0   0   0   0   0   0   0   0   0   0   0
 Intercept: 0   0   0   0   0   0   0   0   0   0   0   0   0   0   0   0   0   0   0
   Waiting: 0   0   0   0   0   0   0   0   0   0   0   0   0   0   0   0   0   0   0
    Verify: 0   0   0   0   0   0   0   0   0   0   0   0   0   0   0   0   0   0   0
   Intrude: 0   0   0   0   0   0   0   0   0   0   0   0   0   0   0   0   0   0   0
       Zip: 3  -3  -3   3   3  -3  -3  -3  -3  -3   3  -3 -3  -3  -3   0   0  -3 -3
     Music: 0   3   3   0   0   6   6   6   3   0   6   3   3   3   3   0   0   0   0
```

FROM / TO

Display-only fields that identify the variable digital tone values.

Valid entries	Usage
-3 through **15**	An unsigned number is a decibel loss, while a number preceded with a minus sign is a decibel gain.

Login Administration

When Avaya Communication Manager is delivered to you, one customer *super-user* login and password combination is already defined for you. You must administer additional logins and passwords, if needed. If you are the super-user, you have full customer permissions and can customize any login you create. The maximum number of customer logins is 11.

As super-user, you can establish permissions for the other logins in your system. You can block access to any object that may compromise Communication Manager security. Once the user login is established, set permissions using the **Command Permission Categories** screen.

Field descriptions for page 1

Figure 462: Login Administration screen

```
 add login                                                  Page 1 of 2
                              LOGIN ADMINISTRATION

               Password of Login Making Change:

           LOGIN BEING ADMINISTERED
                             Login's Name:xxxxxxx
                               Login Type:
                            Service Level:
    Disable Following a Security Violation?
                                                  Access to INADS Port? _

          LOGIN'S PASSWORD INFORMATION
                         Login's Password:
                  Reenter Login's Password:
        Password Aging Cycle Length (Days):

        LOGOFF NOTIFICATION
          Facility Test Call Notification?     Acknowledgment Required?
               Remote Access Notification?     Acknowledgment Required?

ACCESS SECURITY GATEWAY PARAMETERS
Access Security Gateway?
```

Password of Login Making Change

You can make changes to any login with permissions less than your own. You must enter your password to save any changes you make to this screen.

Valid entries	Usage
User password	Enter your password.

LOGIN BEING ADMINISTERED

Access to INADS Port

This field only appears when the **Customer Access to INADS Port** field is set to **y** on the **Maintenance-Related System-Parameters** screen.

Valid entries	Usage
y/n	Enter **y** to allow a user to access the remote administration port.

Disable Following a Security Violation

This field only appears when the **SVN Login Violation Notification** field is set to **y** on the **Security-Related System-Parameters** screen.

Valid entries	Usage
y/n	Enter **y** to have the login disabled following a security violation.

Login’s Name

This display-only field shows the login name specified with the add command.

Login Type

Valid entries	Usage
Customer	Enter **customer** to indicate the login belongs to a customer. This is the only valid value for customer logins.

Service Level

Valid entries	Usage
super-user	Enter super-user to indicate the user is a super-user. Super-user logins can use the add, change, display, list, and remove commands for all customer logins and passwords.
non-super-user	Enter non-super-user to indicate the user is a non-super-user. Non-super-user logins can change their own password, but are otherwise limited to permissions set by super-user.
remote	Customer logins with a Service Level of **remote** can log in to the media server via the analog line and modem.

LOGIN'S PASSWORD INFORMATION

Login's Password

Passwords must be 4 to 11 characters in length and contain at least 1 alphabetic and 1 numeric character. Passwords do not display as you type.

Valid entries	Usage
A to **Z**, **a** to **z**, **0** to **9**, and **! & * ? ; ' ^ ( ) , . : -**	Enter the initial password for login, assigned by the super-user.

Password Aging Cycle Length (Days)

For security, you should assign password aging to all passwords. The system tracks a password from the day the login was created, or the day the user last changed the password.

Valid entries	Usage
1 to **99**	Enter the number of days the password remains valid.

Re-enter Login's Password

Passwords do not display as you type.

Valid entries	Usage
A to **Z**, **a** to **z**, **0** to **9**, and **! & * ?** ; ' **^ ()** , **. : -**	Re-enter the initial password for the login, assigned by the super-user.

LOGOFF NOTIFICATION

Facility Test Call Notification

For security, this field should be set to y for all logins.

Valid entries	Usage
y/n	Enter **y** to have the user receive notification at logoff if Facility Test Notification is still administered.

Facility Test Call Notification — Acknowledgment Required

Valid entries	Usage
y/n	Enter **y** to require the user to acknowledge they want to logoff while Facility Test Notification is still administered.

Remote Access Notification

For security, this field should be set to **y** for all logins.

Valid entries	Usage
y/n	Enter **y** to have the user receive notification at logoff if remote access is still administered.

Remote Access Notification — Acknowledgment Required

Valid entries	Usage
y/n	Enter **y** to require the user to acknowledge they want to logoff while remote access is still administered.

ACCESS SECURITY GATEWAY

Access Security Gateway

This field appears only if the **Access Security Gateway (ASG)** field on the **System-Parameters Customer-Options** screen is set to **y**.

Valid entries	Usage
y/n	Enter **y** to have ASG administered for the login ID.

Field descriptions for page 2

This page only appears if the **Access Security Gateway** field is set to **y** on page 1.

Figure 463: Access Security Gateway Login Administration screen

```
add login                                                  Page 2 of 2
                    ACCESS SECURITY GATEWAY LOGIN ADMINISTRATION

                   Blocked?: n
System Generated Secret Key?: n       Secret Key: _____________________



EXPIRATION CRITERIA
   Expiration Date: __/__/____
Number of Sessions: __

RESTRICTION CRITERIA
Restrict Days of Week
   Monday? n     Tuesday? n     Wednesday? n     Thursday? n     Friday? n
 Saturday? n      Sunday? n

 Restrict From Time: __:__      Restrict To Time: __:__
```

Blocked

Setting the **Blocked** field to **y** does not remove the login ID from the system.

Valid entries	Usage
y/n	Enter **y** to temporarily disable the login ID from accessing the system through the ASG interface.

Expiration Date

Expiration of the login ID occurs at 23:59:59 of the entered date. This field consists of three fields separated by forward slashes (for example, 01/01/1999). Expiration of a login ID does not remove the login ID from the system. The system accepts administration of year 2000 dates.

If an login ID requiring ASG authentication has expired, the login ID for the standard login is not available. If an login ID not requiring ASG login has expired, the login ID still may be active for the standard login as long as expiration criteria associated with that component of the login has not been satisfied.

Valid entries	Usage
mm = month, **dd = day**, and **yyyy = year** (later than the current date)	Enter the date when the associated login ID expires.

Number of Sessions

Expiration of a login ID administered with both Expiration Date and Number of Sessions expiration criteria is based on whichever criteria is satisfied first.

Valid entries	Usage
1 to **999**	Enter the number of times the login ID can be used to access the system via the ASG interface.

Restrict Days of Week

This field consists of seven subfields that correspond to the seven days of the week. Each subfield specifies whether the corresponding login ID is restricted from accessing the system via ASG on the day indicated. Access restrictions imposed by this field apply to the entire day unless limited by time restrictions.

Valid entries	Usage
y/n	Enter **y** to restrict the login from accessing the system on the associated day of the week.

Restrict From Time and Restrict to Time

These two fields are separated by colons, for example, hh:mm where hh = hour and mm = minute. Periods that span an interval that crosses a day boundary are specified by setting the **Restrict From Time** field greater than the **Restrict To Time** field. For example, a **Restriction From Time** of **17:00** and a **Restrict To Time** of **08:00** limits access to the traditional working hours, 8 to 5.

When used with the **Restrict Days of Week** field, overnight periods start on the days where access is not restricted by the **Restrict Day of Week** field.

If the **Restrict Days of Week** field is not specified, the restricted time interval specified by the **Restrict From Time** to **Restrict To Time** fields applies to every day of the week.

Valid entries	Usage
00:00 to **23:59**	Enter the time interval each day the login ID is blocked from accessing the system via the ASG interface.

Secret Key

The value of this field is only displayed during the initial add or change login command to allow it to be programmed into any ASG Site Manager, ASG Mobile, or ASG Pass-Key used to interact with ASG for user authentication.

If the **System Generated Secret Key** field is set to **y**, the system generates the secret key. If the **System Generated Secret Key** field is set to **n**, the administrator is required to enter a 20-digit octal value. It is important to note the value of the secret key so you can enter it into response generation devices.

Valid entries	Usage
20-digit octal string using **0** to **7** as possible digits	Enter the 20-digit octal value used by both the lock and the key to verify user authenticity. The last digit must be **0** and the second to last digit must be either **0**, **2**, **4**, or **6**.

System Generated Secret Key

Valid entries	Usage
y/n	Enter **y** to generate the secret key by the system and display it to the administrator.

Loudspeaker Paging

The **Loudspeaker Paging** screen administers voice paging, deluxe voice paging, and chime paging.

Figure 464: Loudspeaker Paging screen

```
change paging loudspeaker                                    Page   1 of   1
                            LOUDSPEAKER PAGING

                                      CDR? _
              Voice Paging Timeout (sec): ___
             Code Calling Playing Cycles: _

PAGING PORT ASSIGNMENTS
                   Voice Paging     Code Calling
 Zone  Port        TAC  COR  TN     TAC  COR  TN     Location:
   1: ________     ____ __   _      ____ __   _      ______________________________
   2: ________     ____ __   _      ____ __   _      ______________________________
   3: ________     ____ __   _      ____ __   _      ______________________________
   4: ________     ____ __   _      ____ __   _      ______________________________
   5: ________     ____ __   _      ____ __   _      ______________________________
   6: ________     ____ __   _      ____ __   _      ______________________________
   7: ________     ____ __   _      ____ __   _      ______________________________
   8: ________     ____ __   _      ____ __   _      ______________________________
   9: ________     ____ __   _      ____ __   _      ______________________________
 ALL: ________     ____ __   _      ____ __   _
```

CDR

This field determines whether CDR data is collected for the paging ports.

Valid entries	Usage
y/n	Enter **y** if you want the server running Communication Manager to collect CDR data on the paging ports.

Code Calling — COR

This field assigns a Class of Restriction to a paging zone.

Valid entries	Usage
0 to **95**	You can assign different classes of restriction to different zones.
blank	Leave this field blank for unused paging zones.

Code Calling Playing Cycles

This field sets the number of times a chime code will play when a user makes a chime page. To determine the best setting, consider who your code calling users are and whether they are likely to hear the code chime the first time.

Valid entries	Usage
1 to **3**	Enter the number of times you want the chime code to play when a user makes a page.
blank	The field may not be blank when you administer chime paging (code calling).

Code Calling — TAC

This field assigns a Trunk Access Code (TAC) to a paging zone. Users dial this code to make a page to a specific zone. One TAC must be assigned to each zone you want to use. Two zones cannot have the same TAC. If you enter a TAC in the **ALL** field, users may activate speakers in all the zones by dialing that code.

Valid entries	Usage
1 to 4 digits	Enter a Trunk Access Code (TAC) allowed by your dial plan.
*****	May be used as first digit.
#	May be used as first digit.
blank	Leave this field blank for unused paging zones.

Code Calling — TN

Valid entries	Usage
1 to **20** (DEFINITY SI, CSI)	If your system uses Tenant Partitioning, you may use this field to assign a paging zone to a specific tenant partition.
1 to **100** (DEFINITY R, S8300 Media Server, S8700 or S8710 IP-Connect, S8700 or S8710 Multi-Connect)	

Location

Valid entries	Usage
1 to 27 characters	Assign a descriptive name for the physical location corresponding to each zone. Typical entries might be "conference room A," "warehouse," or "storeroom."

Port

This field assigns a port on an auxiliary trunk circuit pack to a paging zone. Enter the necessary characters.

Valid entries	Usage
01 to **44** (DEFINITY R) or **01** to **03** (DEFINITY CSI, SI) or **1** to **64** (S8700 or S8710 IP-Connect / S8700 or S8710 Multi-Connect / S8300 Media Server)	First and second characters are the cabinet number
A to **E**	Third character is the carrier

1 of 2

Valid entries	Usage
0 to **20**	Fourth and fifth character are the slot number
01 to **04** (Analog TIE trunks) **01** to **31**	Six and seventh characters are the circuit number
1 to **80** (DEFINITY R, CSI) or **1** to **10** (DEFINITY SI) or **1** to **250** (S8700 or S8710 IP-Connect / S8700 or S8710 Multi-Connect / S8300 Media Server)	Gateway
V1 to **V9**	Module
01 to **31**	Circuit
blank	Leave this field blank for unused paging zones.

2 of 2

Voice Paging — COR

This field assigns a Class of Restriction to a paging zone.

Valid entries	Usage
0 to **95**	You can assign different classes of restriction to different zones.
blank	Leave this field blank for unused paging zones.

Voice Paging — TAC

This field assigns a Trunk Access Code (TAC) to a paging zone. Users dial this code to make a page to a specific zone. One TAC must be assigned to each zone you want to use. Two zones cannot have the same TAC. If you enter a TAC in the **ALL** field, users may activate speakers in all the zones by dialing that code.

Valid entries	Usage
1 to 4 digits	Enter a Trunk Access Code (TAC) allowed by your dial plan.
*	May be used as first digit.
#	May be used as first digit.
blank	Leave this field blank for unused paging zones.

Voice Paging Timeout (sec)

This field limits the duration of voice pages. When this interval ends, calls are disconnected. To determine the best setting, time the typical pages you expect to broadcast and then add another 4 to 5 seconds.

Valid entries	Usage
10 to **600** seconds	Enter the maximum number of seconds you want any page to last.
blank	The field may not be blank when you administer voice paging.

Note:

To use a port that has no hardware associated with it, place an **x** in this field.

Voice Paging — TN

Valid entries	Usage
1 to **20** (DEFINITY SI, CSI)	If your system uses Tenant Partitioning, you may use this field to assign a paging zone to a specific tenant partition.
1 to **100** (DEFINITY R, S8300 Media Server, S8700 or S8710 IP-Connect, S8700 or S8710 Multi-Connect)	

Related topics

See Setting up Voice Paging Over Loudspeakers on page 491 for instructions.

See "Loudspeaker Paging" in *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205, for more information.

Media-Gateway

This screen administers the media gateways and their corresponding ports. It is required to exist on all Communications Controllers, both external and internal, that support Media Gateways. This screen does not apply to DEFINITY R.

Field descriptions for page 1

Figure 465: Media-Gateway Administration screen

```
add media-gateway x                                          Page   1 of 1
                                MEDIA-GATEWAY

        Number: ___
          Name: ____________________ Identifier: ____________________
    IP Address: _______________     MAC Address: _________________
Network Region: ____                   Location: __
     Site Data: ____________________ Registered? _
                ____________________

                  Slot Module Type
                  V1: # _________
                  V2: ___________
                  V3: ___________
                  V4: ___________



                  V8: messaging-analog
                  V9: gateway-announcements
```

Identifier

This field is for the controller to identify the gateway.

Valid entries	Usage
1 to 20 characters or blank	Enter the serial number or other alpha-numeric characters for identification.

IP Address

A display-only field containing the IP address of the media gateway. This field displays a blank until the media gateway registers for the first time.

Location

This field refers to a time-zone offset, daylight savings rule and number plan area code

Valid entries	Usage
1 to **10**	Enter a number between **1** to **x** or blank where **x** is the maximum value for the different communications controllers.

MAC Address

A display-only field containing the MAC address of the media gateway.

Module Type

Fields **V1** through **V4** list the Media Gateway media modules.

Valid entries	Usage
analog **bri** **dcp** **icc** **voip** **ds1**	Enter the actual hardware media module.

Field **V8** lists a "virtual" media modules (available in August).

Valid entries	Usage
messaging-analog or blank	Messaging will be administered in this slot.

Field V9 lists a "virtual" media modules (available in December).

Valid entries	Usage
gateway-announcements or blank	Announcement will be administered in this slot.

Name

This field is for customer naming of the gateway.

Valid entries	Usage
1 to 20-alphanumeric character string or blank	Enter the alpha-numeric characters for identification.

Network Region

This field is used by the communications controller to allocate resources from the nearest media gateway.

Valid entries	Usage
1 to **80** (DEFINITY CSI, SI) **1** to **250** (S8300 Media Server, S8700 or S8710 IP-Connect, S8700 or S8710 Multi-Connect)	Enter a number between **1** to **x** or blank where **x** is the maximum value for the different communications controllers.

Number

A display-only field showing the assigned media-gateway number.

Registered

A display-only field indicating whether a Media Gateway is registered with a communications controller.

Valid entries	Usage
1 character	Enter a number between **1** to **x** or blank where **x** is the maximum value for the different communications controllers.

Site Data

This field is for general site information.

Valid entries	Usage
2 lines of 20 characters or blank	Enter the alpha-numeric characters for identification.

Meet-me Vector Directory Numbers

Field descriptions for page 1

Figure 466: Meet-me Vector Directory Numbers screen

```
list meet-me-vdn

                  MEET-ME VECTOR DIRECTORY NUMBER

                                    Access          Vec Control
Name                           Ext  Code  COR  TN  Num   Ext

Secure Meet-me Conference    4000   *     1    1   1
Nonsecure Meet-me Conference 4006         1    1   2    84590
```

If the **Access Code** field shows an asterisk (*), an access code is assigned. If the **Access Code** field is blank, no access code is assigned. The access code displays for administrators with super-user permissions (such as *init*).

Note:

If the vector for Meet-Me conferencing allows a new party to join a conference immediately, and that party is joining as an H.323 ip trunk user, the caller may not have talkpath with the others in the conference. To prevent this, include in the vector a short delay before a new party joins the Meet-Me conference, such as a step to collect digits, a 1-second delay, or play an announcement. Since Meet-Me vectors are always configured with announcements and digit collections, this should rarely be an issue.

Mode Code Related System Parameters

This screen establishes parameters associated with the Mode Code Voice Mail System Interface.

Note:

You can only administer this screen if the **Mode Code Interface** field on the **Feature-Related System Parameters** screen is set to **y**.

Field descriptions for page 1

Figure 467: Mode Code Related System Parameters screen

```
change system-parameters mode-code                              Page   1 of x

                   MODE CODE RELATED SYSTEM PARAMETERS

MODE CODES  (FROM SWITCH TO VMS)
      Direct Inside Access: __
Direct Dial Access - Trunk: __
         Internal Coverage: __
         External Coverage: __

           Refresh MW Lamp: __

     System In Day Service: __
   System In Night Service: __

                     OTHER RELATED PARAMETERS
DTMF Duration On (msec): __ Off (msec): __ Sending Delay (msec):__
 VMS Hunt Group Extension : ____
Remote VMS Extensions - First:     Second:
```

MODE CODES (FROM SWITCH TO VMS)

Direct Dial Access - Trunk

This value defines a mode code that the media server or switch sends when an external caller dials the VMS access number.

Valid entries	Usage
0 to **9**, #, *****, **#00**	Up to six digits that may include these characters

Direct Inside Access

This value defines a mode code that the media server or switch sends when a caller at an internal extension dials the Voice Mail System (VMS) access number.

Valid entries	Usage
0 to **9**, #, *****, **#00**	Up to six digits that may include these characters

External Coverage

This value defines a mode code that the media server or switch sends when an external caller tries to reach a user at another extension and the call goes to the user’s voice-mail coverage.

Valid entries	Usage
0 to **9**, #, *****, **#00**	Up to six digits that may include these characters

Internal Coverage

This value defines a mode code that Communication Manager sends when an internal caller tries to reach a user at another extension and the call goes to the user’s voice mail coverage.

Valid entries	Usage
0 to **9**, #, *****, **#00**	Up to six digits that may include these characters

Refresh MW Lamp

This value defines a mode code that Communication Manager sends during a system level 3 or higher reset that requests the VMS to refresh the Message Waiting (MW) lamps.

Valid entries	Usage
0 to **9**, #, *****, **#00**	Up to six digits that may include these characters

System In Day Service

This value indicates to the VMS that the Avaya Communication Manager has changed from Night to Day Service.

Valid entries	Usage
0 to **9**, #, *****, **#11**	Up to six digits that may include these characters

System In Night Service

This value indicates to the VMS that the Avaya Communication Manager has changed from Day to Night Service.

Valid entries	Usage
0 to **9**, #, *****, **#12**	Up to six digits that may include these characters

OTHER RELATED PARAMETERS

DTMF Duration On

Valid entries	Usage
Between **75** and **500** in multiples of 25	Define in milliseconds the length of mode code digits sent to the VMS. This field cannot be blank

Off

Valid entries	Usage
Between **75** and **200** in multiples of 25	Define in milliseconds the pause between mode code digits as they are sent to the VMS. This field cannot be blank.

Remote VMS Extensions - First

You can administer this field if the **Mode Code for Centralized Voice Mail** field on the **System-Parameters Customer-Options** screen is set to **y**. Specifies the first remote UDP VMS hunt group extension.

Valid entries	Usage
Remote assigned hunt group extension	Enter the first UDP VMS hunt group extension.

Remote VMS Extensions - Second

You can administer this field if the **Mode Code for Centralized Voice Mail** field on the **System-Parameters Customer-Options** screen is set to **y**. Specifies the second remote UDP VMS hunt group extension.

Valid entries	Usage
Remote assigned hunt group extension	Enter the second UDP VMS hunt group extension. This extension cannot be the same as the first Remote VMS Extension.

Sending Delay

Valid entries	Usage
75 to **1000** in multiples of 25	Define in milliseconds the delay between the time answer supervision is received from the VMS and the time the first mode code digit is sent. This field cannot be blank.

VMS Hunt Group Extension

A check is made to verify that a valid hunt group extension is entered, but a check is not made to verify that the hunt group members are VMI extensions.

Valid entries	Usage
Valid extension.	Enter the extension of a hunt group containing VMI extensions.

Modem Pool Group

There are two types of conversion resources for Modem Pooling. The first type, an *integrated conversion resource*, is a circuit pack that emulates a Trunk Data Module connected to a 212A-type modem. Two conversion resources are on each circuit pack.

The second type, a *combined conversion resource*, is a separate Trunk Data Module and modem administered as a unit. The Trunk Data Module component of the conversion resource may be either a Modular Trunk Data Module (MTDM) or 7400A Data Module and connects to a digital port using Digital Communications Protocol (DCP); the modem connects to an analog port.

Field descriptions for page 1

Figure 468: Modem Pool Group screen — if Group Type is integrated

```
change modem-pool num                                      Page 1 of x
                                MODEM POOL GROUP
                    Group Number: 1                 Group Type: integrated
Receiver Responds to Remote Loop? n              Hold Time (min): 5
          Send Space Disconnect? y     Receive Space Disconnect? y
                    CF-CB Common? y   Loss of Carrier Disconnect? y

Speed: LOW/300/1200      Duplex: full     Synchronization: a/sync

CIRCUIT PACK ASSIGNMENTS
               Circuit Pack   Circuit Pack
               Location       Location
               1: ___         9: ___
               2: ___        10: ___
               3: ___        11: ___
               4: ___        12: ___
               5: ___        13: ___
               6: ___        14: ___
               7: ___        15: ___
               8: ___        16: ___
```

Figure 469: Modem Pool Group screen — if Group Type is combined

```
change modem-pool num                                        Page 1 of x
                                MODEM POOL GROUP
                   Group Number: _           Group Type: combined
                     Modem Name: ______ Hold Time (min): 5_
                     Time Delay: 0_           Direction: two-way
Answer Supervision Timeout(sec): _

     Speed: LOW/300/1200___     Duplex: full     Synchronization: async

PORT PAIR ASSIGNMENTS
       Analog Digital    Analog Digital     Analog Digital      Analog Digital
     1: _____ _____    9: _____ _____    17: _____ ______    25: _____ _____
     2: _____ _____   10: _____ _____    18: _____ ______    26: _____ _____
     3: _____ _____   11: _____ _____    19: _____ ______    27: _____ _____
     4: _____ _____   12: _____ _____    20: _____ ______    28: _____ _____
     5: _____ _____   13: _____ _____    21: _____ ______    29: _____ _____
     6: _____ _____   14: _____ _____    22: _____ ______    30: _____ _____
     7: _____ _____   15: _____ _____    23: _____ ______    31: _____ _____
     8: _____ _____   16: _____ _____    24: _____ ______    32: _____ _____
```

Note:

The **Speed**, **Duplex**, and **Synchronization** fields cannot be filled out for the "integrated" pooled modem screens but can be assigned on the "combined" pooled modem screen. The integrated conversion resource automatically will adjust its speed and synchronization to the endpoint it is connected to. In synchronous mode, the integrated modem pool can operate at 1200 baud. In asynchronous mode, it can operate at 300 or 1200 baud. Full-duplex operation is always used.

Answer Supervision Timeout (sec)

This field appears only when the **Group Type** field is **combined**.

Valid entries	Usage
1 to **255**	Enter the number of seconds to wait before the far-end answers.
0	No answer supervision

CF-CB Common

This field appears only when the **Group Type** field is **integrated**.

Valid entries	Usage
y/n	Enter **y** to indicate that the CF and CB leads on the conversion resource are logically connected.

Direction

Enter the direction of the call for which modem pool will operate. This field appears only when the **Group Type** field is **combined**.

Valid entries	Usage
incoming	Converts an analog signal to digital for the data endpoint.
outgoing	Converts analog to digital (or digital to analog) for data calls.
two-way	Allows incoming and outgoing data communication.

Duplex

Display-only when the **Group Type** field is **integrated**. When the **Group Type** field is **combined**, enter the duplex mode of the conversion resources in the group.

Valid entries	Usage
full	Can talk and listen at the same time.
half	Cannot talk and listen at the same time.

Group Number

A display-only field when the screen is accessed using an administration command such as **add** or **change**.

Group Type

This field designates what physical model pool you are going to.

Valid entries	Usage
integrated	Maps to the Pooled Modem circuit pack.
combined	Maps to an external modem pool (when you have a data module and a modem).

Hold Time (min)

Valid entries	Usage
1 to **99**	Enter the maximum number of minutes that a conversion resource in the group may be held while a call waits in a queue or reserved after Data Call Preindication.

Loss of Carrier Disconnect

This field appears only when the **Group Type** field is **integrated**.

Valid entries	Usage
y/n	Enter **y** to permit conversion resource to disconnect if it detects a dropped carrier.

Modem Name

Indicates the name of the modem pool. This field appears only when the **Group Type** field is **combined**.

Valid entries	Usage
1 to 6 alphanumeric character string	Enter the name of the modem pool.

Receive Space Disconnect

This field appears only when the **Group Type** field is **integrated**.

Valid entries	Usage
y/n	Enter **y** to allow the conversion resource to disconnect after receiving 1.6 seconds of space.

Receiver Responds to Remote Loop

This field appears only when the **Group Type** field is **integrated**.

Valid entries	Usage
y/n	Enter **y** to allow the far-end modem to put conversion resource into loop back mode.

Send Space Disconnect

This field appears only when the **Group Type** field is **integrated**.

Valid entries	Usage
y/n	Enter **y** to allow the conversion resource to send 4 seconds of space before disconnecting.

Speed

Display-only when the **Group Type** field is **integrated**. When the **Group Type** field is **combined**, enter the communication speed in bits per second of the conversion resources in the group. Enter one to three speeds separated by slashes (for example, 300/1200/2400) to indicate a maximum of three running speeds.

Valid entries	Usage
LOW	0 to 300 blind sampled
300	
1200	
2400	

1 of 2

Valid entries	Usage
4800	
9600	
19200	
	2 of 2

Synchronization

Display-only when the **Group Type** field is **integrated**. When the **Group Type** field is **combined**, enter the synchronization mode of the conversion resources in the group.

Valid entries	Usage
sync	Synchronous
async	Asynchronous

CIRCUIT PACK ASSIGNMENTS are optional on "integrated" conversion resource screens only.

Time Delay

This field appears only when the **Group Type** field is **combined**.

Valid entries	Usage
0 to **255**	Enter the time delay in seconds to insert between sending the ringing to the modem and the off-hook alert to the data module.

CIRCUIT PACK ASSIGNMENTS

Circuit Pack Location

Displays when the **Group Type** field is **integrated**. Enter the port associated with the conversion resource on the integrated modem pool circuit pack. Enter the necessary characters.

Valid entries	Usage
01 to **44** (DEFINITY R) or **01** to **03** (DEFINITY CSI, SI) or **1** to **64** (S8700 or S8710 IP-Connect / S8700 or S8710 Multi-Connect / S8300 Media Server)	First and second characters are the cabinet number
A to **E**	Third character is the carrier
0 to **20**	Fourth and fifth character are the slot number
01 to **04** (Analog TIE trunks) **01** to **31** **1** to **80** (DEFINITY R, CSI) or **1** to **10** (DEFINITY SI) or **1** to **250** (S8700 or S8710 IP-Connect / S8700 or S8710 Multi-Connect / S8300 Media Server)	Six and seventh characters are the circuit number Gateway
V1 to **V9**	Module
01 to **31**	Circuit

For example, 01A0612 is in cabinet 01, carrier A, slot 06, and circuit number (port) 12.

PORT PAIR ASSIGNMENTS are optional on "combined" pooled modem screens only.

PORT PAIR ASSIGNMENTS

Analog Digital

Displays when the **Group Type** field is **combined**. Enter the port numbers of the modem/TDM pair in a conversion resource.

Two port entries are required. Enter the necessary characters.

Valid entries	Usage
01 to **44** (DEFINITY R) or **01** to **03** (DEFINITY CSI, SI) or **1** to **64** (S8700 or S8710 IP-Connect / S8700 or S8710 Multi-Connect / S8300 Media Server)	First and second characters are the cabinet number
A to **E**	Third character is the carrier
0 to **20**	Fourth and fifth character are the slot number
01 to **04** (Analog TIE trunks) **01** to **31**	Six and seventh characters are the circuit number
1 to **80** (DEFINITY R, CSI) or **1** to **10** (DEFINITY SI) or **1** to **250** (S8700 or S8710 IP-Connect / S8700 or S8710 Multi-Connect / S8300 Media Server)	Gateway
V1 to **V9**	Module
01 to **31**	Circuit

Note:

For example, 01A0612 is in cabinet 01, carrier A, slot 06, and circuit number (port) 12.

Multifrequency-Signaling-Related Parameters

This screen sets the system or location parameters associated with multifrequency signaling. With the Multinational Locations feature enabled, multifrequency signaling can be administered per location, rather than system-wide. This screen appears when **Incoming Call Type** is **group-ii-mfc** and **Outgoing Call Type** is **none**. Page 2 of this screen appears when both **Incoming Call Type** and **Outgoing Call Type** are **group-ii-mfc**.

If the field **Use COR for All Group II Responses** is set to **y**, the **Group II Called Party Category** and **Use COR for Calling Party Category** fields do not appear.

Note:

With the Multinational Locations feature enabled, you can assign MFC signal sets per trunk group, rather than system-wide.

Figure 470: Multifrequency-Signaling-Related Parameters screen

```
change multifrequency-signaling                                Page  1 of  X

            MULTIFREQUENCY-SIGNALING-RELATED SYSTEM PARAMETERS

         Incoming Call Type:                        ANI Prefix:
         Outgoing Call Type:                      Default ANI:
       Maintenance Call Type:             NEXT ANI DIGIT
                                                     Incoming:
     Maximum Resend Requests:                        Outgoing:
   Received Signal Gain (dB):
Transmitted Signal Gain (dB):

                          Request Incoming ANI (non-AAR/ARS)?
                    Outgoing Forward Signal Present Timer (sec):
                     Outgoing Forward Signal Absent Timer (sec):
MF Signaling Intercept Treatment - Incoming? _  Outgoing: _____________ _____
                             Collect All Digits Before Seizure?
                   Overlap Sending on Link-to-Link Tandem Calls?
           Private Group II Permissions and Public Interworking?
                             Convert First Digit End-of-ANI To: _
                            Use COR for All Group II Responses? _
                                Group II Called Party Category:
                            Use COR for Calling Party Category?
```

The **ANI Prefix**, **Default ANI**, and **Collect All Digits Before Seizure** fields appear only when the value of the **Outgoing Call Type** field is **group-ii-mfc** or **mfe**.

If **Collect All Digits Before Seizure** is **y**, **Overlap Sending on Link-to-Link Tandem Calls and Convert First Digit End-of-ANI** are not displayed.

Default ANI

This field appears only when **Outgoing Call Type** is **group-ii-mfc** or **mfe**.

Valid entries	Usage
2 to **15**	Enter the PBX identification number that is sent to the CO when ANI is requested (by the CO) on a particular call but is not available, such as on tandem tie trunk calls.
blank	Use for tandeming. If this field is blank, you must enter a value in the **ANI-Not-Available** field.

ANI Prefix

This field appears only when **Outgoing Call Type** is **group-ii-mfc** or **mfe**.

Valid entries	Usage
1 to 6 digits or blank	Enter the prefix to apply to an extension when ANI is sent to the CO.

Backward Cycle Timer (sec)

Appears when the **Incoming Call Type** field is **mfe**.

Valid entries	Usage
1 to **255**	Enter the number of seconds to wait to send the check frequency after receiving an MFE signal.

Collect All Digits Before Seizure

Appears when the **Outgoing Call Type** is **group-ii-mfc** or **mfe**.

Valid entries	Usage
y	The system collects all the digits before seizing the trunk and the **ANI Req** field on the **AAR and ARS Digit Conversion Table** does not apply.
n	Enter **n** to control ANI collection via the **ARS** screens.

Convert First Digit End-of-Dial To

Appears when the **Private Group II Permissions and Public Interworking** field is **y**.

Valid entries	Usage
0 to **9**, **#**, or blank	Enter the digit used when the incoming initial end-of-ani or end-of-dial MF signal is converted on a per-switch basis.

Forward Cycle Timer (sec)

Appears when the **Incoming Call Type** field is **mfe**.

Valid entries	Usage
1 to **255**	Enter the number of seconds to wait to receive the check frequency after sending an MFE signal. Communication Manager drops the call if the time runs out before it receives check frequency.

Group II Called Party Category

Appears when the **Outgoing Call Type** field is **group-ii-mfc** and the **Use COR for All Group II Responses** field is **n**. Enter the type of group II signals that should be used on the outgoing R2-MFC call.

Note:

For India, the ANI can be requested without the **Incoming Call Type** information.

Valid entries	Usage
user-type	The type of telephone making the call determines the type of group II signal that the PBX sends (normal = ordinary telephone set, attendant = attendant console, data-call = data modules and similar data endpoints).
call-type	The dialed digits determine the type of group II signal that the PBX sends.

This field defines the signal type that a CO uses to place an incoming call to the PBX.

Valid entries	Usage
group-ii-mfc	If the value of this field is **group-ii-mfc**, the second page of the screen will display entries for all group-I, group-II, group-A, and group-B signal types with a set of default values (see page 2 of screen).
non-group-ii-mfc	If the value is **non-group-ii-mfc**, the second page displays only group-I and group-A signal types.
mfe	Use only in Spain (multi-frequency Espanol)

Incomplete Dial Timer (sec)

Appears when the **Incoming Call Type** field is **mfe**.

Valid entries	Usage
45 to **255**	Enter the number of seconds to wait from the start of a call until the end of the check frequency of the last signal. Communication Manager drops the call if the time runs out before it receives the check frequency.

Maintenance Call Type

Appears when the **Incoming Call Type** field is **group-ii-mfc** or **non-group-ii-mfc**.

Valid entries	Usage
1	The Belgium maintenance sequence is indicated when the CO sends an MFC maintenance tone.
2	The Saudi Arabian sequence is indicated when the CO sends an MFC maintenance tone.
none	

Maximum Resent Requests

Valid entries	Usage
1 to **99**	Enter the threshold number of resend type MFC signals your server running Communication Manager accepts during an outgoing call.
1	The call is dropped if one resend signal is received.
blank	An unlimited number of resend requests is allowed.

MF Signaling Intercept Treatment - Incoming

Valid entries	Usage
y	Send the group B signal for the intercept to the CO and play intercept tone on the trunk.
n	Use normal DID/TIE/ISDN intercept treatment.

MF Signaling Intercept Treatment - Outgoing

Displays when the **Outgoing Call Type** field is **group-ii-mfc**.

Valid entries	Usage
announcement	Plays a recorded announcement for outgoing calls that cannot be completed as dialed. You select and record the message. Enter the extension number for the announcement in the associated field.
tone	Plays intercept tone for outgoing calls that cannot be completed as dialed.

MFE Type

This field only appears when **Incoming Call Type** is **mfe**.

Valid entries	Usage
2/5	Determines which public signaling Communication Manager will use.
2/6	

Outgoing Call Type

This field defines the signal type that the PBX uses to place an outgoing call into a CO.

Valid entries	Usage
group-ii-mfc	If the content of this field is **group-ii-mfc**, the system displays the third page of the screen. The third page displays entries for all group-I, group-II group-A, and group-B signal types with a set of default values.
mfe	Use only in Spain (multi-frequency Espanol)
none	If the content of this field is **none**, the system does not display the third page. In addition, **Outgoing Forward Signal Present Timer**, **Outgoing Forward Signal Absent Timer**, **ANI Prefix**, **ANI for PBX**, **Next ANI Digits**, and **Collect All Digits Before Seizure** will not display on field descriptions for page 1.

Outgoing Forward Signal Absent Timer (sec)

This field appears only when the content of **Outgoing Call Type** is **group-ii-mfc**.

Valid entries	Usage
11 to **255**	Enter the maximum number of seconds to elapse between forward signals on outgoing calls. The timer starts (and restarts) when a forward tone is taken off the link and it stops when the next forward tone is applied to the link.

Outgoing Forward Signal Present Timer (sec)

This field appears only when the value of **Outgoing Call Type** is **group-ii-mfc**.

Valid entries	Usage
1 to **255**	Enter the maximum number of seconds to elapse between signals on a call. This timer runs when MFC tones are being sent or received on an outgoing call. The timer starts (and restarts) when Communication Manager begins sending a forward signal and stops when Communication Manager receives the backward signal.

Outgoing Start Timer (sec)

Appears when the **Incoming Call Type** field is **mfe**.

Valid entries	Usage
1 to **255**	Enter the number of seconds to time from seizure until the beginning of the first Group A signal from the receiving end, and from the end of the check frequency until the beginning receipt of the first digit following the Group II signal.

Overlap Sending on Link-to-Link Tandem Calls

Does not appear if the **Collect All Digits Before Seizure** field is **y**. When Avaya Communication Manager has this field set to **y**, and calls are tandeming between media servers or switches, then ANI for PBX will be sent to the terminating switch if that switch requests ANI before Avaya Communication Manager receives it from the originating server/switch. The terminating server/switch may request ANI before the receipt of the last address digit if it is not running Avaya Communication Manager, or if it is Avaya Communication Manager with the **Request Call Category at Start of Call** field set to **y**.

Valid entries	Usage
y/n	If **y**, Avaya Communication Manager sends and receives digits one digit at a time instead of enbloc. (With enbloc, digits are not sent until the entire group of digits is received).

Private Group II Permissions and Public Interworking

Displays when the **Incoming Call Type** field is **group-ii-mfc** or **non-group-ii-mfc** and the **Outgoing Call Type** field is **group-ii-mfc** or **none**.

Valid entries	Usage
y/n	If **y**, then Avaya Communication Manager: • Sends the category for MFC ANI for the COR of the originating party for non-private-MFC-trunk to MFC-private-trunk calls. • Sends the Group II category received over the incoming private trunk as the outgoing Group II category on tandem private MFC calls. • Applies MFC group II-CPC termination restrictions on incoming MFC private trunk calls. • Checks station permissions if you call forward off-net.

Received Signal Gain (dB)

Valid entries	Usage
-15 to **3**	Enter the number for the loss/gain when the MFC port listens to the trunk port. Communication Manager listens with a range of -5 to -35 and this value moves the range (for example, a value of -5 provides a range of -10 to -40). This value also applies to Russian MF Shuttle trunks.

Request Incoming ANI (non-AAR/ARS)

Appears when the **Incoming Call Type** field is **group-ii-mfc** or **mfe** and the **Outgoing Call Type** field is **group-ii-mfc** or **mfe**. This field only applies if the incoming call via the R2-MFC trunk is terminating to a local station on this PBX.

Valid entries	Usage
y/n	If **y**, ANI should be requested on incoming R2-MFC calls.

Transmitted Signal Gain (dB)

Valid entries	Usage
-15 to **3**	Enter the number for the loss/gain when the trunk port listens to the MFC port. The MFC port generates at -5 for MFC and -8 for MFE, and this field adds gain or loss to the starting value of -5. This value also applies to Russian Shuttle trunks and Russian multi-frequency ANI.

Use COR for All Group II Responses

Appears if the **Outgoing Call Type** field is **group-ii-mfc**.

Valid entries	Usage
y/n	Enter **y** to allow the COR administered category to be used for both the calling party and called party categories.

Use COR for Calling Party Category

Appears when the **Outgoing Call Type** field is **group-ii-mfc** and the **Use COR for All Group II Responses** field is **n**. Indicates the category to send with ANI if requested on an outgoing R2-MFC call.

Valid entries	Usage
y	Use the calling facility's COR to determine category.
n	Use the calling party's user-type COR to determine category.

NEXT ANI DIGIT

Incoming

Appears when the **Incoming Call Type** field is **group-ii-mfc** and the **Outgoing Call Type** field is **group-ii-mfc** or **mfe**.

Valid entries	Usage
next-digit **next_ani_digit** **send-ani**	Enter a value to determine whether the Next ANI Digit signal will be the same as the "send-ani" signal or the "next-digit" signal or another signal defined as "next_ani_digit."

Outgoing

Appears when the **Outgoing Call Type** field is **group-ii-mfc**.

Valid entries	Usage
next-digit **next_ani_digit** **send-ani**	Enter a value to determine whether the Next ANI Digit signal will be the same as the "send-ani" signal or the "next-digit" signal or another signal defined as "next_ani_digit."

Field descriptions for page 2

The fields on Page 2 define call category and ANI information. For India, the ANI can be requested without the call category information.

Figure 471: Multifrequency-Signaling-Related Parameters screen

```
change multifrequency-signaling 1                              Page   2 of   3
            MULTIFREQUENCY-SIGNALING-RELATED PARAMETERS  1

     Request Call Category at Start of Call? n            Outgoing II by COR
           Restart ANI from Caller Category? y             1:  1
              Number of Incoming ANI Digits: 0             2:  2
              Number of Outgoing ANI Digits: 0             3:  3
             Truncate Station Number in ANI: no            4:  4
Address Digits Include End-of-digits Signal? n             5:  5
         Call Category for Vector ii-digits? n             6:  6
               Request CPN at Start of Call? n             7:  7
          Do Not Send Group B Signals to CO? n             8:  8
   ANI Source for Forwarded & Covered Calls: caller        9:  9
                                                           10: 10
                        INCOMING    OUTGOING
          ANI Available:
      ANI Not Available:
```

Address Digits Include End-of-Digits Signal

Indicates that an outgoing forward Group I end-of-digit signal is always sent after completion of address digits upon request from the Central Office for outgoing calls.

Valid entries	Usage
y/n	Enter **y** to send an outgoing forward Group I end-of-digit signal after completion of address digits upon request from the Central Office for outgoing calls.

ANI Source for Forwarded & Covered Calls

Valid entries	Usage
caller	Send the calling party's ANI when calls are redirected.
forwarder	Send the forwarding party's ANI when calls are redirected.

Call Category for Vector ii-digits

Allows you to use the call category digit as the ii-digits on call vector steps.

Valid entries	Usage
y/n	If **y**, the call category digit, which is a part of ANI, is used as the ii-digits on call vector steps.

Do Not Send Group B Signals to CO

This field appears only if the **Incoming Call Type** field is **group-ii-mfc**. This field allows completion of a call without Group-B signals.

Valid entries	Usage
y	If **y**, does not send Group-B signals to complete an incoming call.
n	If **n**, sends Group-B signals to complete an incoming call.

Number of Incoming ANI Digits

Valid entries	Usage
0 to **15**	Enter the number of ANI digits for incoming MFC calls.

Number of Outgoing ANI Digits

This field applies to Russian shuttle trunks, and MFC and MFE trunks.

Valid entries	Usage
0 to **15**	Enter the number of ANI digits for outgoing MFC calls. In India or any country where end-of-ani and end-of-digits are not defined for Tones to CO on Outgoing Forward Calls - Group I, Avaya Communication Manager appends ANI-Not-Available digits to ANI digits if the actual ANI length is less than the number entered in this field. If end-of-ani or end-of-digits are defined, this field is used in conjunction with Truncate Station Number in ANI as a maximum ANI length. For India, even if the length of ANI is defined, if the timeout occurs during the ANI collection, the call is routed with the ANI digits already collected.

Outgoing II by COR

Appears only if either **Use COR for Calling Party Category** or **Use COR for All Group II Responses** on page 1 are set to **y**. The Group II signal sent to the CO on outgoing calls can be administered per COR (Class of Restriction) and per trunk group. The Group II signal is administered per COR. That per-COR value in turn can be mapped into a possibly different outgoing signaling parameter set. The values in the **Outgoing II by COR** fields administer that outgoing mapping.

Valid entries	Usage
1 to **10**	Enter a number between 1 and 10 that maps to the Group II signal Communication Manager sends to the CO on outgoing calls

Request Call Category at Start of Call

Indicates that the Send-ANI backward signal requesting for the caller-category information will be sequenced differently in the MFC signaling flow. The Caller-category Request backward signal is disjointed from the ANI request.

Valid entries	Usage
y/n	If **y**, the Send-ANI backward signal corresponds exclusively to the caller-category request. In response to this signal, Avaya Communication Manager sends a forward signal containing the caller-category information on outgoing calls. On incoming calls, Communication Manager sends the Send-ANI backward signal upon receipt of the first address signal.

Request CPN at Start of Call

This field appears only if the **Incoming Call Type** field is **group-ii-mfc**. Provides for Communication Manager to collect ANI and call category immediately after receipt of the first address digit.

Valid entries	Usage
y/n	If **y**, provides ANI (Calling Party Number (CPN) and call category) immediately after receiving the first address digit.

Restart ANI from Caller Category

Display-only field.

Valid entries	Usage
y/n	If **y**, Avaya Communication Manager sends the caller-category signal later again when the signals for Caller-Category and ANI requests are the same and this signal is received after the Next-Digit forward signals have been received.

Truncate Station Number in ANI

This field applies to Russian shuttle trunks, and MFC and MFE trunks.

Valid entries	Usage
beginning **ending** **no**	This field defines the side of the extension number from which to truncate when station ANI is sent to the CO and the combined length of the ANI prefix and extension number is greater than Number of Outgoing ANI Digits. The ANI prefix (either MFC or COR) is not truncated. There is no effect if ANI for PBX is sent.

INCOMING / OUTGOING

ANI Available

Valid entries	Usage
1 to **15** or blank	Enter the number of incoming and outgoing ANI-Available signals.

ANI Not Available

You must enter a value if the **ANI** for **PBX** field is blank.

Valid entries	Usage
1 to **15** or blank	Enter the number of the incoming and outgoing ANI-Not-Available signals. Communication Manager outpulses the End-of-Dial backward signal when the ANI-Not-Available forward signal is received on incoming calls. Communication Manager outpulses the ANI-Not-Available forward signal to the CO on outgoing calls where ANI is not possible.

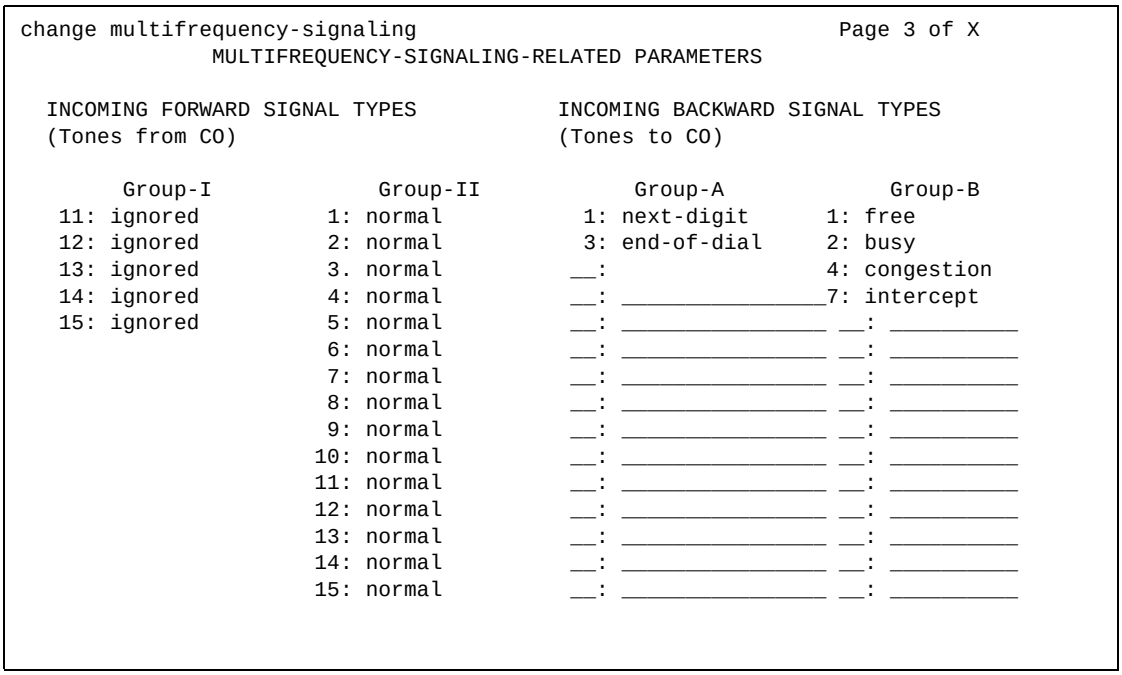
Field descriptions for page 3

The fields shown on Page 3 of the **Multifrequency-Signaling-Related System Parameters** screen define the meaning of MFC tones for calls originated at the CO. See Definitions of Group I, II, A, and B signals on page 1316 for more information. This screen appears only if the **Incoming Call Type** field is **group-ii-mfc** or **non-group-ii-mfc**.

When the screen initially appears, either of two sets of default values is possible. One set is for the group II call type; the other set is for non-group II call type. In each set, the default value for each field is set to the most common value.

The **Multifrequency-Signaling-Related Parameters** screen shows the defaults when the **Incoming Call Type** field is **group-ii-mfc**. A variation appears if **Incoming Call Type** is **non-group-ii-mfc**. When **Incoming Call Type** is **non-group-ii-mfc**, group II and group B columns do not appear.

Figure 472: Multifrequency-Signaling-Related Parameters screen



```
change multifrequency-signaling                              Page 3 of X
              MULTIFREQUENCY-SIGNALING-RELATED PARAMETERS

  INCOMING FORWARD SIGNAL TYPES          INCOMING BACKWARD SIGNAL TYPES
  (Tones from CO)                        (Tones to CO)

       Group-I            Group-II            Group-A            Group-B
   11: ignored          1: normal           1: next-digit      1: free
   12: ignored          2: normal           3: end-of-dial     2: busy
   13: ignored          3. normal           __:                4: congestion
   14: ignored          4: normal           __: _______________7: intercept
   15: ignored          5: normal           __: _____________ __: ___________
                        6: normal           __: _____________ __: ___________
                        7: normal           __: _____________ __: ___________
                        8: normal           __: _____________ __: ___________
                        9: normal           __: _____________ __: ___________
                       10: normal           __: _____________ __: ___________
                       11: normal           __: _____________ __: ___________
                       12: normal           __: _____________ __: ___________
                       13: normal           __: _____________ __: ___________
                       14: normal           __: _____________ __: ___________
                       15: normal           __: _____________ __: ___________
```

INCOMING FORWARD SIGNAL TYPES (Tones from CO)

Group I

Message codes 11 to 15 display. (Numbers 1 through 10 are assigned to the digits of the destination telephone number.) Assign a meaning to each code. See Definitions of Group I, II, A, and B signals on page 1316 for signal type.

Valid entries	Usage
drop	If **Incoming Call Type** is **group-ii-mfc**
ani-avail	
end-of-ani	
end-of-dial	
ignored	
maint-call	
ani-not-avail	
send-congest	
drop	If the **Incoming Call Type** is **non-group-ii-mfc**
ignored	

Group II

Message codes 1 to 15 display. Assign a meaning to each code.

Valid entries	Usage
attendant	See Definitions of Group I, II, A, and B signals on page 1316 for signal type.
busy-rt-attd	
data-call	
data-verify	
drop	

1 of 2

Valid entries	Usage
maint-call	
send-intercept	
toll-auto	
toll-operator	
normal	
	2 of 2

INCOMING BACKWARD SIGNAL TYPES (Tones to CO)

Group A

Message codes 11 to 15 display. (Numbers 1 through 10 are assigned to the digits of the destination telephone number.) Assign a meaning to each code.

Valid entries	Usage
congestion	See Definitions of Group I, II, A, and B signals on page 1316 for signal type.
end-of-dial	
intercept	
next-ani-digit	
next-digit	
send-ani	
setup-sppath	

Group B

This field does not appear if the **Do Not Send Group B Signals to CO** field is **y**. Message codes between 1 and 15 display. Assign a meaning to each code.

Valid entries	Usage
busy	See Definitions of Group I, II, A, and B signals on page 1316 for signal type.
congestion	
free	
mct	
tariff-free	
tie-free	
toll-busy	
intercept	

Field descriptions for page 4

The fields shown on this page define the meaning of MFC tones for calls originated at the PBX. See Definitions of Group I, II, A, and B signals on page 1316 for more information.

Page 4 of the **Multifrequency-Signaling-Related System Parameters** screen only appears if **Outgoing Call Type** is **group-ii-mfc** or **mfe**.

Figure 473: Multifrequency-Signaling-Related System Parameters screen

```
change system-parameters multifrequency-signaling                Page 4 of x
               MULTIFREQUENCY-SIGNALING-RELATED SYSTEM PARAMETERS

  OUTGOING FORWARD SIGNAL TYPES          OUTGOING BACKWARD SIGNAL TYPES
  (Tones to CO)                          (Tones from CO)

       Group-I            Group-II            Group-A            Group-B
  11: _____________   2: normal           1: next-digit      1: free
  12: _____________   5: attendant        2: congestion      2: busy
  13: _____________   6: data-call        3: end-of-dial     3: congestion
  14: _____________   __: _____________   4: congestion       4: congestion
  15: _____________   __: _____________   5: call-info-ani    5: congestion
                      __: _____________   6: congestion       6: free
                      __: _____________   7: last-2-digits    7: intercept
                      __: _____________   8: last-3-digits    8: congestion
                      __: _____________   9: congestion       9: congestion
                      __: _____________  10: congestion      10: congestion
                      __: _____________  11: congestion      11: congestion
                      __: _____________  12: congestion      12: congestion
                      __: _____________  13: congestion      13: congestion
                      __: _____________  14: congestion      14: congestion
                      __: _____________  15: congestion      15: congestion
```

OUTGOING FORWARD SIGNAL TYPES (Tones to CO)

Group I

Message codes 11 to 15 appear. (Numbers 1 through 10 are assigned to the digits of the destination telephone number.)

Valid entries	Usage
end-of-digits	Assign a meaning to each code. See Definitions of Group I, II, A, and B signals on page 1316 for signal type.
ani-avail	
end-of-ani	
ani-not-avail	

Group II

Message codes between 1 and 15 display. Assign a meaning to each code. Each entry can only appear once in the group II column.

Valid entries	Usage
attendant	See Definitions of Group I, II, A, and B signals on page 1316 for signal type.
data-call	
toll-auto	
normal	

OUTGOING BACKWARD SIGNAL TYPES (Tones from CO)

Group A

Message codes between 1 and 15 display. Assign a meaning to each code.

Valid entries	Usage
send-ani	See Definitions of Group I, II, A, and B signals on page 1316 for signal type.
congestion	
drop	
end-of-dial	
last-2-digits	
last-3-digits	
last-digit	
next-ani-digit	
next-digit	
restart	
intercept	

1 of 2

Valid entries	Usage
resend-digit	
setup-sppath	
	2 of 2

Group B

Message codes between 1 and 15 display. Assign a meaning to each code. See Definitions of Group I, II, A, and B signals on page 1316 for signal type.

Valid entries	Usage
busy	
congestion	
free	
mct	
tariff-free	
toll-busy	
intercept	

Definitions of Group I, II, A, and B signals

Group I signals

Group I signals are a set of forward signals generated by the originating media server or switch.

ani-avail

Used in Hungary. If this signal is defined and ANI is requested on outgoing R2-MFC calls, ANI is sent to the CO before ANI caller digits are sent. This signal is sent after the ANI caller category signal.

ani-not-avail

Used on DOD calls in Brazil and Columbia. Communication Manager sends this signal to the CO when it receives an ANI request and the caller’s number is not available.

digits 1 to 10

The signals from group I.1 to I.10 are reserved for address digits 0 to 9.

drop

When this signal is received from the CO, Avaya Communication Manager starts the disconnect sequence and drops the call.

end-of-ani

This signal is used on DOD and DID calls. Communication Manager sends this signal to indicate the end-of-ANI digits when ANI digits are sent to the CO.

end-of-dial

This signal is used when open numbering is used on DID calls. The CO sends this signal to indicate the end-of-dial digits and Communication Manager responds with a request for a group II signal.

end-of-digits

This signal is sent by the origination media server or switch that makes outgoing calls, sends digits, and receives a next-digit group A signal from the destination server or switch when there are no more digits to be sent.

This signal is also sent when Communication Manager does not have end-of-ani assigned, makes an outgoing call, sends ANI, and receives a call-info-ani group A signal from the destination end when there are no more ANI digits to be sent.

If both end-of-digits and end-of-ani are assigned, Communication Manager uses end-of-ani after it sends the last ANI digit and end-of-digits after sending the last called-number digit.

ignored

If this signal is received from the CO, Communication Manager sends a corresponding signal (A.1, and so on) but no action is taken in the response and it is not counted as a digit. In Belgium, this signal is not acknowledged.

maint-call

The CO sends a signal to indicate that a call is a maintenance call and Communication Manager prepares the special maintenance call sequences for the CO. This signal may be used on DID calls in Saudi Arabia.

send-congestion

When Communication Manager receives this signal from the CO on a DID call, it returns a congestion signal (group A), in compel (not pulse) mode, to the CO.

Group II signals

Group II signals are a more elaborate set of forward signals generated by the originating media server or switch.

attendant

If Communication Manager receives this signal on DID calls, the call terminates at an attendant regardless of the extension dialed. On DOD calls, this signal is sent to the CO if the CO requests calling-category information and the originating extension is an attendant. This signal is used on both DID and DOD calls.

busy-rt-attd

If Communication Manager receives this signal on DID calls, the call terminates at an attendant if the called extension is busy or at the called extension if it is not busy. This signal is used on DID calls.

data-call

This signal is treated the same as the data-verify signal except that it does not require a terminating extension to be a data extension.

data-verify

If Communication Manager receives this signal on DID calls and the terminating extension is not a data extension, it sends intercept treatment. On DOD calls, this signal is sent to the CO if the CO requests calling-category information and the originating extension is a data extension. This signal is used on both DID and DOD calls.

drop

When this signal is received from the CO, Communication Manager starts the disconnect sequence and drops the call.

maint-call

The CO sends a signal to indicate that a call is a maintenance call and Avaya Communication Manager prepares the special maintenance call sequences for the CO.

normal

This signal indicates that the caller is a normal subscriber. If it is received on a DID call, the call is terminated at the called extension. For an outgoing MF signaling call that uses group II signaling, this signal is sent to the CO when the CO requests calling-category information and the originating extension is a station. This signal is used in both DID and DOD calls.

send-intercept

If Communication Manager receives this signal from the CO on a DID call, it returns group B intercept signal to the CO.

toll-auto

This signal is used in China. This signal indicates that a call is an automatic toll call. When the call terminates at a busy station and a special busy signal is defined, the busy signal is sent to the CO. You can define the special busy signal by choosing the option toll-busy on the incoming group B signals.

toll-operator

This signal, used in China, is treated as a normal subscriber signal. See the normal definition.

Group A signals

Group A signals are backward signals generated by the destination server/switch.

send-ani

The CO sends this signal to request calling-party category and sends additional signals to request ANI digits. This signal is sent to the CO when Avaya Communication Manager requests ANI digits on DID calls. This signal is used on both DOD and DID calls.

congestion

The CO sends this signal to indicate that it is experiencing network congestion. When Communication Manager receives this signal on DOD calls, it drops the trunk and plays reorder tone to the calling party. This signal is used on DOD calls.

drop

When this signal is sent, the receiving end starts the disconnect sequence.

end-of-dial

This signal is sent to indicate the end of the address digit string. For MF group II calls, this signal requests a group II signal and switches the sender over to the group B signaling mode. This signal is used on both DID and DOD calls.

intercept

The CO sends this signal to indicate the call has been terminated to an invalid destination. When Communication Manager receives this signal on DOD calls, it drops the trunk and plays intercept tone to the calling party. This signal is used on DOD calls.

resend-digit

Communication Manager sends this signal to adjust the outpulsing pointer so that the last digit can be resent again. This signal is used on DOD calls.

last-digit

Communication Manager sends this signal to adjust the outpulsing pointer so that the last 2 digits can be resent. This signal is used on DOD calls.

last-2-digits

Communication Manager sends this signal to adjust the outpulsing pointer so that the last 3 digits can be resent. This signal is used on DOD calls.

last-3-digits

Communication Manager sends this signal to adjust the outpulsing pointer so that the last 4 digits can be resent. This signal is used on DOD calls.

next-digit

Communication Manager sends this signal to request the next digit. This signal is used on both DID and DOD calls.

next-ani-digit

Communication Manager sends this signal to request the next ANI digit. This signal is used on DID and DOD calls.

restart

Communication Manager sends this signal to request the whole digit string again. This signal is used on DOD calls.

setup-sppath

The CO sends this signal to Communication Manager to set up a speech path. This signal is used on DOD calls and on DID calls in Belgium.

Group B signals

Group B signals enhance group A signals for backward signaling from the destination end by providing the status of the called party. In addition, if the originating media server or switch uses group II signals, the destination end answers with group B signals. Group B signals are as follows:

busy

This signal is sent to indicate that the called party is busy. On DID calls, the signal is sent to the CO if there is no coverage point to terminate the call. If Communication Manager receives this signal on DOD calls, it plays busy tone to the calling party and drops the trunk.

congestion

This signal is sent to indicate that the system is congested and the call cannot be terminated successfully. On DID calls, the signal is sent to the CO to indicate that a resource is not available. On DOD calls, if Communication Manager receives this signal, reorder tone is played to the calling party and the trunk is dropped.

free

This signal indicates that the called party is idle. On DID calls, the signal is sent to the CO to indicate that the called party is idle and the call is terminated successfully. If Communication Manager receives this signal on DOD calls, it connects the trunk to the calling party.

intercept

This signal indicates that the called party number is not in service or is not correct. On DID calls, if intercept treatment is set to provide a tone, tone is sent to the CO to indicate that the called number is not valid. If Communication Manager receives the signal on DOD calls, it plays intercept tone to the calling party and drops the trunk.

mct

This signal identifies the call as one that needs to be traced by the CO. Avaya Communication Manager then generates an MFC Call Trace Backward Signal (administered on the **Multifrequency-Signaling-Related System-Parameters** screen) during call setup instead of the "free" signal. If the terminating station's COR has this feature set to **y**, the CO collects trace information before releasing the calling party.

Note:

If the station's COR has **MF Incoming Call Trace** set to **y** and the "mct" signal is not defined, then the "free" signal is sent.

tariff-free

This signal is sent when the trunk group provides an "800" service. Avaya Communication Manager generates an MFC tariff-free backward signal (administered on the **System-Parameters Multifrequency-Signaling** screen) during call setup instead of the "free" signal, facilitating CO billing.

Note:

If the trunk is administered as a tariff-free trunk and the "tariff-free" signal is not defined, then the "free" signal is sent.

tie-free

This signal is used only when an incoming call is received and defined and the incoming facility is a tie trunk. Otherwise, the free signal is used.

toll-busy

This signal, used in China, is sent to indicate that the called party is busy if the call is an automatic toll call.

Multiple Level Precedence & Preemption (MLPP) Parameters

Use this screen to set up system parameters for the Multiple Level Precedence & Preemption feature.

Field descriptions for page 1

Figure 474: Multiple Level Precedence and Preemption Parameters screen

```
change system-parameters mlpp                                  Page  1 of  1

                 MULTIPLE LEVEL PRECEDENCE & PREEMPTION PARAMETERS
  ANNOUNCEMENTS
          Blocked Precedence Level: 6801          Service Interruption: 6803
     Unauthorized Precedence Level: 6802            Busy, Not Equipped: 6804
                       Vacant Code: 6805

  PRECEDENCE CALLING-DIALED DIGIT ASSIGNMENT
     Flash Override: 0  Flash: 1  Immediate: 2  Priority: 3  Routine: 4

         Attendant Diversion Timing (sec): 60
            Remote Attendant Route String:
     Worldwide Numbering Dial Plan Active? y      Default Route Digit: 0
            Precedence Call Timeout (sec): 30
      Line Load Control Restriction Level: 0
          WNDP Emergency 911 Route String:
                    Default Service Domain: 1
       ISDN Precedence Call Timeout (sec): 30
```

ANNOUNCEMENTS

Blocked Precedence Level

Valid entries	Usage
Valid extension or blank	Enter the extension of the Blocked Precedence Level announcement you want to use.

Busy, Not Equipped

Valid entries	Usage
Valid extension or blank	Enter the extension of the Busy, Not Equipped for Preemption announcement you want to use.

Service Interruption

Valid entries	Usage
Valid extension or blank	Enter the extension of the Service Interruption announcement you want to use.

Unauthorized Precedence Level

Valid entries	Usage
Valid extension or blank	Enter the extension of the Unauthorized Precedence Level announcement you want to use.

Vacant Code

Valid entries	Usage
Valid extension or blank	Enter the extension of the Vacant Code announcement you want to use.

PRECEDENCE CALLING-DIALED DIGIT ASSIGNMENT

CAUTION:

Avaya recommends that you do not change the default Precedence Calling dialed digits unless you are coordinating this change with other companion networks in your system. If the Precedence Calling digits do not match across networks, the system does not properly process the calls. Each of the Precedence Calling digits must be different. You cannot use the same digit for two different precedence levels.

Attendant Diversion Timing (sec)

Valid entries	Usage
10-99 or blank	

Default Route Digit

Appears only when Worldwide Numbering Dial Plan Active is **y**. You must enter a valid digit in this field.

Valid entries	Usage
0	Voice call (the default value)
1	Circuit switched data call
2	Satellite avoidance call
3	(reserved)
4	(reserved)
5	Hotline voice grade call
6	Hotline data grade call
7	(reserved)
8	(reserved)
9	(reserved)

Default Service Domain

Valid entries	Usage
0 to **16777215**	This number defines the system service domain, and must be unique within a switching network. The system uses the system service domain to determine eligibility for precedence calling when interswitch precedence calls over non-ISDN trunks occur.

Flash

Valid entries	Usage
0 to **9** or blank	Enter the digit assignment for Flash precedence level calls.Default is **1**.

Flash Override

Valid entries	Usage
0 to **9** or blank	Enter the digit assignment for Flash Override precedence level calls.Default is **0**.

Immediate

Valid entries	Usage
0 to **9** or blank	Enter the digit assignment for Immediate precedence level calls.Default is **2**.

ISDN Precedence Call Timeout (sec)

Valid entries	Usage
4 to **30**	This timeout is used instead of the Precedence Call Timeout when the call is from an MLPP ISDN-PRI trunk. Default is 30 seconds.

Line Load Control Restriction Level

These system levels determine what stations, based on their COR, will be restricted from originating calls.

Valid entries	Usage
0	Feature not active (no restrictions) (default).
2	Restrict stations with a COR assigned to LLC levels 2, 3, and 4.
3	Restrict stations with a COR assigned to LLC levels 3 and 4.
4	Restrict stations with a COR assigned to LLC level 4.

Precedence Call Timeout (sec)

A busy user receives a precedence call waiting tone only if the incoming call cannot be connected and cannot preempt the user. The called party hears the tone every 10 seconds until answered or the administered time-out occurs. If ignored, the caller is diverted to an attendant or a call-forwarded station.

Valid entries	Usage
4 to **30**	Default is **30**. Enter the number of seconds before a precedence call remains in call waiting status before it is diverted.

Priority

Valid entries	Usage
0 to **9** or blank	Enter the digit assignment for Priority precedence level calls.Default is **3**.

Remote Attendant Route String

Valid entries	Usage
1 to 24 digits or blank	Enter a user-defined telephone to which a precedence call can be routed when no console or night telephone is administered.

Routine

Valid entries	Usage
0 to **9** or blank	Enter the digit assignment for Routine precedence level calls. Default is **4**.

WNDP Emergency 911 Route String

Valid entries	Usage
1 to 24 digits or blank	Valid entries for this field can be a trunk access code (TAC), the AAR or the ARS access code, a WNDP access code, or an extension. If you use a WNDP access code, use the access code for the lowest precedence calling level in the system. **Note:** An Emergency/911 call is a call that routes using the ARS table with the call type defined as either "alrt" or "emer."

Worldwide Numbering Dial Plan Active

Valid entries	Usage
y/n	Enter **y** to enable the Worldwied Numbering Dial Plan. Default is **n**.

Music Sources

This screen defines music sources for Tenant Partitions. Each music source defined on the screen can be used by one or more Tenant Partitions. However, a partition may have only one music source.

Note:

If you use equipment that rebroadcasts music or other copyrighted materials, you may be required to obtain a copyright license from, or pay fees to, a third party. You can purchase a Magic Hold system, which does not require such a license, from Avaya Inc. or Avaya's business partners.

Field descriptions for page 1

Figure 475: Music Sources screen

```
change music-source            Music Sources                      Page 1 of X

               Source    Type      Port       Description

                 1      music      01A1003    Easy listening
                 2      tone                  Tone-on-Hold
                 3      music      01A1004    Rock
                 4      none
                 5      none
                 6      none
                 7      music      12B1301    Oldies
                 8      none
                 9      none
                10      none
                11      music      04C2003    Classical
                12      none
                13      none
                14      none
                15      none
```

Description

This field appears only if you entered **music** or **tone** in **Type**.

Note:

When Tenant Partitioning is enabled, **Music/Tone on Hold** on the **Feature-Related System Parameters** screen disappears. However, the value in that field (**tone**, **music**, or **none**) will appear as the first entry on the **Music Sources** screen. If the value was **music**, the port number also appears on the **Music Sources** screen. When Tenant partitioning is disabled, **Music/Tone on Hold** reappears on the **Feature-Related System Parameters** screen, along with the values from the **Music Sources** screen.

Valid entries	Usage
20 alpha-numeric character (max)	Enter a description of the administered music source.

Port

Enter the auxiliary trunk or analog port address of the music source. Duplicates are not allowed. This field appears only if you entered **music** in **Type**. Enter the necessary characters.

Valid entries	Usage
01 to **44** (DEFINITY R) or **01** to **03** (DEFINITY CSI, SI) or **1** to **64** (S8700 or S8710 IP-Connect / S8700 or S8710 Multi-Connect / S8300 Media Server)	First and second characters are the cabinet number
A to **E**	Third character is the carrier
0 to **20**	Fourth and fifth character are the slot number
01 to **04** (Analog TIE trunks) **01** to **31**	Six and seventh characters are the circuit number

1 of 2

Valid entries	Usage
1 to **80** (DEFINITY R, CSI) or **1** to **10** (DEFINITY SI) or **1** to **250** (S8700 or S8710 IP-Connect / S8700 or S8710 Multi-Connect / S8300 Media Server)	Gateway
V1 to **V9**	Module
01 to **31**	Circuit
	2 of 2

Source

Display only field - the number assigned to this source. The maximum number of music sources is 20 for DEFINITY CSI and SI and 100 for DEFINITY R. This screen appears with the appropriate pages to accommodate the number of music sources your system can support.

Type

If you entered a value in **Music/Tone on Hold** on the **Feature-Related System Parameters** screen, that value will appear in this field.

Valid entries	Usage
music	Enter the type of treatment to be provided by the music source.
tone	Only one music source may use this value.
none	

Node Number Routing

This screen specifies the routing pattern associated with each node in a public or private network. Node Number Routing is a required capability for Extension Number Portability (ENP) and is associated with the Uniform Dial Plan (UDP).

Field descriptions for page 1

Figure 476: Node Number Routing screen

```
change node-routing 1                                          Page 1 of x
                          NOE NUMBER ROUTING
                       Partitioned Group Number: 1

     Route      Route      Route      Route      Route      Route      Route
      Pat        Pat        Pat        Pat        Pat        Pat        Pat

                15 ___     30 ___     45 ___     60 ___     75 ___     90 ___
    1 ___       16 ___     31 ___     46 ___     61 ___     76 ___     91 ___
    2 ___       17 ___     32 ___     47 ___     62 ___     77 ___     92 ___
    3 ___       18 ___     33 ___     48 ___     63 ___     78 ___     93 ___
    4 ___       19 ___     34 ___     49 ___     64 ___     79 ___     94 ___
    5 ___       20 ___     35 ___     50 ___     65 ___     80 ___     95 ___
    6 ___       21 ___     36 ___     51 ___     66 ___     81 ___     96 ___
    7 ___       22 ___     37 ___     52 ___     67 ___     82 ___     97 ___
    8 ___       23 ___     38 ___     53 ___     68 ___     83 ___     98 ___
    9 ___       24 ___     39 ___     54 ___     69 ___     84 ___     99 ___
   10 ___       25 ___     40 ___     55 ___     70 ___     85 ___
   11 ___       26 ___     41 ___     56 ___     71 ___     86 ___
   12 ___       27 ___     42 ___     57 ___     72 ___     87 ___
   13 ___       28 ___     43 ___     58 ___     73 ___     88 ___
   14 ___       29 ___     44 ___     69 ___     74 ___     89 ___
```

Partitioned Group Number

This read-only field displays the partitioned group number associated with the node numbers being administered.

Valid entries	Usage
Display only	The partitioned group number is either specified on the command line or defaults to partitioned group number 1.

Node Number

This display-only field lists the node number to be changed.

Valid entries	Usage
Display only	Two pages display simultaneously for a total of 200 nodes (100 per page). For example, entering change node-routing 87 displays nodes 1 through 199, and entering change node-routing 151 displays nodes 100 through 299. However, entering change node-routing 999 displays nodes 900 through 999 on one page.

Route Pat

Enter the routing pattern associated with the corresponding node number. This field repeats the same number of times as there are node numbers on the page.

Valid entries	Usage
1 to **254**	Enter a number between 1 and 254, or blank.

Packet Gateway Board

Use this screen to administer the Packet Gateway (PGATE) circuit pack.

Note:

The **PGATE** screen is used with DEFINITY R configurations.

Field descriptions for page 1

Figure 477: Packet Gateway Board screen

```
change pgate xxxxx
                          PACKET GATEWAY BOARD                Page 1 of 1
        Board Location: _____                          Name: _______________
           Application:  X.25
   External cable type: rs232
    Port configuration: 1) rs232  2)rs232  3)rs232 4)rs232
```

Application

Display-only field that shows the communications protocol used to transmit messages over the PGATE.

Board Location

Enter the slot location of the PGATE circuit pack.

Valid entries	Usage
1 to **x**	cabinet (Maximum value varies according to type of switch)
A to **E**	carrier
0 to **20**	slot

External Cable Type

Display-only field that shows the type of physical interface between the PGATE port and the adjunct.

Name

Valid entries	Usage
Up to 15 alphanumeric characters	Enter the name of the adjunct with which the PGATE circuit pack communicates.

Port Configuration

Display-only field that shows that the port is configured for "rs232" communication.

Partition Routing Table

Use this table to identify routing for partition groups associated with an ARS analysis entry.

Figure 478: Partition Routing Table screen

```
change partition route-table                                    Page 1 of X
                            Partition Routing Table

                                 Routing Patterns
 Route
 Index        PGN 1   PGN 2   PGN 3   PGN 4   PGN 5   PGN 6   PGN 7   PGN 8
 -----        -----   -----   -----   -----   -----   -----   -----   -----
 196          _____   _____   _____   _____   _____   _____   _____   _____
 197          _____   _____   _____   _____   _____   _____   _____   _____
 198          _____   _____   _____   _____   _____   _____   _____   _____
 199          _____   _____   _____   _____   _____   _____   _____   _____
 200          _____   _____   _____   _____   _____   _____   _____   _____
 201          _____   _____   _____   _____   _____   _____   _____   _____
 202          _____   _____   _____   _____   _____   _____   _____   _____
 203          _____   _____   _____   _____   _____   _____   _____   _____
 204          _____   _____   _____   _____   _____   _____   _____   _____
 205          _____   _____   _____   _____   _____   _____   _____   _____
 206          _____   _____   _____   _____   _____   _____   _____   _____
 207          _____   _____   _____   _____   _____   _____   _____   _____
 208          _____   _____   _____   _____   _____   _____   _____   _____
 209          _____   _____   _____   _____   _____   _____   _____   _____
 210          _____   _____   _____   _____   _____   _____   _____   _____
```

PGN 1 (through PGN 8)

Enter the routing for each partition group associated with each route index number.

Valid entries	Usage
1 to **640**	Specifies the route pattern used to route the call
r1 to **r32**	Specifies the remote home numbering plan area table used to route the call
node	Designates node number routing
deny	Blocks the call

Personal CO Line Group

Use this screen to set up a personal central office line trunk group.

Figure 479: Personal CO Line Group screen

```
add personal-CO-line                                          Page 1 of x
                          PERSONAL CO LINE GROUP

 Group Number: __                    Group Type:_____________  CDR Reports:
_
   Group Name: ____________________                                  TAC:
____
Security Code: ____              Coverage Path: ____     Data Restriction? _
                               Outgoing Display? _

TRUNK PARAMETERS
                 Trunk Type: _____________          Trunk Direction:
________
                 Trunk Port: ________         Disconnect Timing(msec): ____
                 Trunk Name: ___________            Trunk Termination: _______
         Outgoing Dial Type: __________             Analog Loss Group: ___
                   Prefix-1? _                     Digital Loss Group: ___
Disconnect Supervision - In? _                       Call Still Held? _
 Answer Supervision Timeout: ___           Receive Answer Supervision? _
                 Trunk Gain: ____                              Country: __
           Charge Conversion: _____              DS1 Echo Cancellation: _
               Decimal Point: ______
             Currency Symbol: ___
                 Charge Type: _______
```

The following fields are unique to this screen.

Analog Loss Group

This field determines which administered 2-party row in the loss plan applies to this trunk group if the call is carried over an analog signaling port in the trunk group.

Valid entries	Usage
1 to **17**	Shows the values from the loss plan and tone plan.

Coverage Path

Valid entries	Usage
1 to **9999**	Enter the number of the call coverage path you want to use for incoming calls.
t1 to **t999**	Enter the number of a time-of-day table.
blank	Assigning a coverage path is optional: leave this field blank if you do not want to assign one.

Digital Loss Group

This field determines which administered 2-party row in the loss plan applies to this trunk group if the call is carried over a digital signaling port in the trunk group.

Valid entries	Usage
1 to **17**	Shows the values from the loss plan and tone plan.

DS1 Echo Cancellation

Allows you to administer echo cancellation per channel.

Valid entries	Usage
y/n	Enter **y** to activate echo cancellation capability.

Security Code

Valid entries	Usage
3 to 8 digits	Enter a code that users must dial to retrieve voice messages and to use the Demand Print Message feature.
blank	Leave this field blank if you do not want to use a security code to control access.

Trunk Direction

Valid entries	Usage
incoming	Enter the direction of the traffic on this trunk group. The entry in this field affects which timers appear on the Administrable Timers page. For **WATS Group Types**, only **incoming** or **outgoing** may be entered.
outgoing	
two-way	

Trunk Name

Valid entries	Usage
1 to 10 characters	Enter a descriptive name for this trunk. Don't use the group type (CO, FX, WATS) here. For example, you might use names that identify the vendor and function of the trunk group: USWest Local; Sprint Toll, etc.
blank	

Trunk Port

Enter the necessary characters.

Valid entries	Usage
01 to **44** (DEFINITY R) or **01** to **03** (DEFINITY CSI, SI) or **1** to **64** (S8700 or S8710 IP-Connect / S8700 or S8710 Multi-Connect / S8300 Media Server)	First and second characters are the cabinet number
A to **E**	Third character is the carrier
0 to **20**	Fourth and fifth character are the slot number
01 to **04** (Analog TIE trunks) **01** to **31**	Six and seventh characters are the circuit number

1 of 2

Valid entries	Usage
1 to **80** (DEFINITY R, CSI) or **1** to **10** (DEFINITY SI) or **1** to **250** (S8700 or S8710 IP-Connect / S8700 or S8710 Multi-Connect / S8300 Media Server)	Gateway
V1 to **V9**	Module
01 to **31**	Circuit
	2 of 2

Field descriptions for page 2

Figure 480: Personal CO Line Group screen

```
add personal-CO-line                                          Page 1 of x
                            PERSONAL CO LINE GROUP

ASSIGNED MEMBERS (Stations with a button for this PCOL Group)

        Ext    Name                                Ext    Name

     1:                                        9:
     2:                                        10:
     3:                                        11:
     4:                                        12:
     5:                                        13:
     6:                                        14:
     7:                                        15:
     8:                                        16:
```

Ext

This display-only field shows the extension of phones that have a **CO Line** button.

Name

This display-only field shows the name assigned to phones that have a **CO Line** button.

Field descriptions for page 3

Administrable timers for Personal CO Line groups appear on field descriptions for page 3. See Tone Generation on page 1529 for standard field definitions of the available timers.

Related topics

See Adding a PCOL Trunk Group on page 431 for instructions.

See Trunk Group on page 1534 for definitions of all trunk group fields that are *not* unique to the **PCOL** screen.

Pickup Group

This screen implements call pickup groups with up to 50 extensions per group. A pickup group is a group of users authorized to answer calls to a telephone extension within that group of users. A telephone extension number can only belong to one pickup group.

Field descriptions for pages 1 and 2

Figure 481: Pickup Group screen

```
change pickup-group 1                                    Page   1 of   x
                                  PICKUP GROUP

                   Group Number: 1        Extended Group Number: ____

GROUP MEMBER ASSIGNMENTS

   Ext      Name (first 26 characters)      Ext      Name (first 26 char.)
 1:51001    mips 2a1501                     14:
 2:                                         15:
 3:                                         16:
 4:                                         17:
 5:                                         18:
 6:                                         19:
 7:                                         20:
 8:                                         21:
 9:                                         22:
10:                                         23:
11:                                         24:
12:                                         25:
13:
```

Ext

Enter the extension assigned to a station.

Valid entries	Usage
An extension number.	A VDN cannot be assigned to a Call Pickup group.

Extended Group Number

This field appears only when the **Group Call Pickup** field is set to **flexible** on the **Feature-Related System Parameters** screen. The extended group is a collection of pickup groups that can answer calls from other pickup groups in the same extended group.

Valid entries	Usage
1 to **100** (DEFINITY SI, CSI) **1** to **400** (DEFINITY R)	Enter the extended group number or blank.

Group Number

Valid entries	Usage
Pickup Group number	This display-only field appears when the screen is accessed using an administration command such as **add** or **change**.

Name

This display-only field shows the name assigned to the above extension number when the users and their associated extensions were administered.

Precedence Routing Digit Analysis Table

Avaya Communication Manager compares dialed numbers with the dialed strings in this table and determines the route pattern of an outgoing Multiple Level Precedence and Preemption (MLPP) call.

Figure 482: Precedence Routing Digit Analysis Table screen

```
change precedence-routing analysis 10                          Page 1 of 2

                    PRECEDENCE ROUTING DIGIT ANALYSIS TABLE
                                                      Percent Full: 22

        Dialed           Total     Route    Preempt
        String          Min  Max  Pattern   Method
   002383               9    9     36       group
   002385               9    9     35       group
   002388               9    9     86       group
   003032383            12   12    36       group
   003032388            12   12    86       group
   003033383            12   12    34       group
   003033388            12   12    84       group
   003034383            12   12    32       group
   003034388            12   12    82       group
   003035383            12   12    30       group
   003035388            12   12    80       group
   003383               9    9     34       group
   003385               9    9     33       group
   003388               9    9     84       group
   004383               9    9     32       group
```

Dialed String

User-dialed numbers are matched to the dialed string entry that most closely matches the dialed number.

Valid entries	Usage
0 to **9**	Enter up to 18 digits that the call-processing server analyzes.
***, x, X**	wildcard characters

Max

Valid entries	Usage
Between Min and **28**	Enter the maximum number of user-dialed digits the system collects to match to the dialed string.

Min

Valid entries	Usage
Between **1** and Max	Enter the minimum number of user-dialed digits the system collects to match to the dialed string.

Percent Full

Valid entries	Usage
0 to **100**	Display only. Shows the percent of the Precedence Routing Digit Analysis Table that is currently in use.

Preempt Method

Enter the preemption method you want the server running Avaya Communication Manager to use for this dialed string.

Valid entries	Usage
group	Default.The system checks the first trunk group in the route pattern to determine if any trunks are idle. If the system finds an idle trunk, the sustem connects the call.
route	The system checks each trunk group in the route pattern to determine if any trunks are idle. If the system finds an idle trunk, the call is connected.

Route Pattern

Enter the route number you want the server running Avaya Communication Manager to use for this dialed string.

Valid entries	Usage
1 to **999**	Specifies the route pattern used to route the call.
deny	Blocks the call

Precedence Routing Digit Conversion Table

Use the **Precedence Routing Digit Conversion** screen to assign the Precedence Routing digit conversion. Digit conversion takes digits dialed on incoming calls and converts the digits to local telephone numbers, usually extension numbers.

Figure 483: Precedence Routing Digit Conversion Table screen

```
change precedence-routing digit-conversion 0             Page   1 of   2

                  PRECEDENCE ROUTING DIGIT CONVERSION TABLE
                                                        Percent Full: 11

  Matching Pattern     Min   Max   Del   Replacement String   Net  Conv

  x2386                8     8     4                          ext   n
  x3032386             11    11    7                          ext   n
  x3033386             11    11    7                          ext   n
  x3034386             11    11    7                          ext   n
  x3035386             11    11    7                          ext   n
  x3386                8     8     4                          ext   n
  x4386                8     8     4                          ext   n
  x5386                8     8     4                          ext   n
  x6                   5     5     1                          ext   n
  xx2386               9     9     5                          ext   n
  xx3032386            12    12    8                          ext   n
  xx3033386            12    12    8                          ext   n
  xx3034386            12    12    8                          ext   n
  xx3035386            12    12    8                          ext   n
```

Conv

Valid entries	Usage
y	Enter **y** to allow more conversions.
n	Enter **n** to prevent further conversions.

Del

Valid entries	Usage
0 to **Min**	Enter the number of leading digits to delete.

Matching Pattern

Valid entries	Usage
0 to **9**, *****, **x**, or **X**	Enter the precedence digit and the address digits. For WNDP dialing, you must also enter the route code.

CAUTION:

The **Matching Pattern** field requires the following format for routing DSN numbers: For precedence dialing (non-WNDP dialing), enter the precedence digit (typically 0-4) and the address digits. For WNDP dialing, enter the precedence digit (typically 0-4), the route code, and the address digits. An **x** in the digit string is a wildcard that matches on any single digit.

Max

Valid entries	Usage
Between **Min** and **28**	Enter the maximum number of user-dialed digits the system collects to match to the dialed string.

Min

Valid entries	Usage
Between **1** and **Max**	Enter the minimum number of user-dialed digits the system collects to match to the dialed string.

Net

Valid entries	Usage
ext	Extension. Uses ARS tables or AAR tables to route the call.
pre	Precedence routing. Uses the Precedence Analysis Tables to route the call.

Replacement String

Valid entries	Usage
0 to **9, *, #,** or blank, up to 18 characters	Enter the digits that replace the deleted portion of the dialed number. Leave this field blank to simply delete the digits. The # sign, if present in the string, should be the last character in the string. This signifies the end of the modified digit string.

Route Pattern

Valid entries	Usage
1 to **999**	Specifies the route pattern used to route the call.
deny	Blocks the call.

PRI Endpoint

This screen administers PRI Endpoints for the Wideband Switching feature.

Note:

A PRI Endpoint with a width greater than 1 may be administered only if the **Wideband Switching** feature has been enabled on the **System-Parameters Customer-Options** screen.

A PRI Endpoint is an endpoint application connected to line-side ISDN-PRI facilities and has standard ISDN-PRI signaling interfaces to the system. For information on endpoint applications connected to line-side non-ISDN T1 or E1 facilities, see Access Endpoint on page 730 in this module.

A PRI Endpoint is defined as 1 to 31 adjacent DS0s/B-channels, addressable via a single extension, and signaled via a D-channel (Signaling Group) over a standard T1 or E1 ISDN-PRI interface.

Figure 484: PRI Endpoint screen

```
add pri-endpoint next                                        Page   1 of   1
                                PRI ENDPOINT

                    Extension: 300
                         Name: 27 character PRI Endpoint 1
               (Starting) Port:                          Width: 1
 Originating Auto Restoration? n              Signaling Group:
                          COR: 1                        COS: 1
                           TN: 1           Simultaneous Calls? n
            Maintenance Tests? y

                            WIDEBAND SUPPORT OPTIONS

                               H0? n
                              H11? n
                              H12? n
                            NXDS0? y   Contiguous? n
```

COR

Valid entries	Usage
0 to **95**	Enter class of restriction (COR) to determine calling and called party privileges

COS

Valid entries	Usage
0 to **15**	Enter the Class of Service (COS) to determine the features that can be activated by, or on behalf of, the endpoint.

Extension

A display-only field when the screen is accessed using an administration command such as **change** or **display**.

Valid entries	Usage
Extension	This is the extension number used to access the PRI endpoint. Enter a valid unassigned extension number when completing a paper screen.

Maintenance Tests

Valid entries	Usage
y/n	Enter **y** to run hourly maintenance tests on this PRI Endpoint.

Name

Identifies the endpoint.

Valid entries	Usage
Up to 27 alphanumeric characters.	Enter a name for the endpoint.

Originating Auto Restoration

Valid entries	Usage
y/n	Enter **y** to automatically restore calls originating from this PRI Endpoint (while maintaining endpoint call status) in the case of network failure if the call is over SDDN network facilities.

Signaling Group

Valid entries	Usage
1 to **416** or blank (DEFINITY R, S8300 Media Server, S8700 or S8710 IP-Connect, S8700 or S8710 Multi-Connect) **1** to **110** or blank (DEFINITY CSI, SI)	Enter the D-channel or D-channel pair that will provide the signaling information for the set of B-channels that make up the PRI Endpoint.

Simultaneous Calls

Valid entries	Usage
y/n	Enter **y** to specify that multiple simultaneous calls can be placed to/from the PRI Endpoint.

(Starting) Port

Enter the seven-character starting port of the PRI Endpoint. Enter the necessary characters.

Valid entries	Usage
01 to **44** (DEFINITY R) or **01** to **03** (DEFINITY CSI, SI) or **1** to **64** (S8700 or S8710 IP-Connect / S8700 or S8710 Multi-Connect / S8300 Media Server)	First and second characters are the cabinet number
A to **E**	Third character is the carrier
0 to **20**	Fourth and fifth character are the slot number
01 to **04** (Analog TIE trunks) **01** to **31**	Six and seventh characters are the circuit number

TN

Valid entries	Usage
1 to **20** (DEFINITY CSI, SI) **1** to **100** (DEFINITY R, S8300 Media Server, S8700 or S8710 IP-Connect, S8700 or S8710 Multi-Connect)	Enter the Tenant Partition number.

Width

Enter the number of adjacent DS0 ports beginning with the specified Starting Port, that make up the PRI Endpoint.This field cannot be blank.

Valid entries	Usage
1 to **31**	A width of **6** defines a PRI Endpoint that can support data rates up to 384 Kbps.

WIDEBAND SUPPORT OPTIONS

Contiguous

Specifies whether to hunt contiguous NXDS0 channels. This field only appears if **y** is entered in **NXDS0**. The hunt algorithm to satisfy an NXDS0 call is as follows:

Valid entries	Usage
y/n	Enter **y** to specify the "floating" scheme. NXDS0 calls are placed on a contiguous group of B-channels large enough to satisfy the requested bandwidth without constraint on the starting channel (no fixed starting point trunk).
n	Enter **n** to specify the "flexible" scheme. NXDS0 calls are placed on any set of B-channels on the same facility as long as the requested bandwidth is satisfied. There are no constraints, such as contiguity of B-channels or fixed starting points.

H0

Valid entries	Usage
y/n	Enter **y** to specify the ISDN information transfer rate for 384 Kbps of data, which is comprised of six B-channels. When a PRI Endpoint is administered to support H0, the hunt algorithm to satisfy a call requiring 384 Kbps of bandwidth uses a fixed allocation scheme.

H11

Valid entries	Usage
y/n	Enter **y** to specify the ISDN information transfer rate for 1536 Kbps of data, which is comprised of 24 B-channels. When a PRI Endpoint is administered to support H11, the hunt algorithm to satisfy a call requiring 1536 Kbps of bandwidth uses a fixed allocation scheme.

H12

Valid entries	Usage
y/n	Enter **y** to specify the ISDN information transfer rate for 1920 Kbps data, which includes 30 B-channels. When a PE is administered to support H12, the hunt algorithm to satisfy a call requiring 1920 Kbps of bandwidth uses a fixed allocation scheme.

NXDS0

Valid entries	Usage
y/n	Enter **y** to specify the NXDS0 multi-rate service.

QSIG to DCS TSC Gateway

The **QSIG to DCS TSC Gateway** screen determines when and how to convert messages from a QSIG NCA-TSC to an administered AUDIX NCA-TSC. This screen maps the QSIG subscriber number to the appropriate AUDIX signaling group and TSC index.

This screen only appears if the **Interworking with DCS** field is **y** on the **System-Parameters Customer-Options** screen.

Figure 485: QSIG to DCS TSC Gateway screen

```
change isdn qsig-dcs-tsc-gateway                                Page 1 of x

                          QSIG TO DCS TSC GATEWAY

      Subscriber          Sig   TSC          Subscriber          Sig     TSC
       Number             GRP  Index          Number            GRP    Index
  ____________________    ___   ___     ____________________    ___     ___
  ____________________    ___   ___     ____________________    ___     ___
  ____________________    ___   ___     ____________________    ___     ___
  ____________________    ___   ___     ____________________    ___     ___
  ____________________    ___   ___     ____________________    ___     ___
  ____________________    ___   ___     ____________________    ___     ___
  ____________________    ___   ___     ____________________    ___     ___
  ____________________    ___   ___     ____________________    ___     ___
```

Sig Grp

Valid entries	Usage
1 to **110**	Enter the assigned signaling group number for DEFINITY CSI, SI.
1 to **416**	Enter the assigned signaling group number for DEFINITY R.
1 to **650**	Enter the assigned signaling group number for S8300 Media Server, S8700 or S8710 IP-Connect, S8700 or S8710 Multi-Connect.

Subscriber Number

You can enter up to 28 subscriber numbers.

Valid entries	Usage
0 to **9**, *****, '**x**', '**X**'	Enter a subscriber number up to 20 characters in length. You can use wildcards '**x**' and '**X**' to enter subscriber numbers.

TSC Index

You must complete the **TSC Index** field for each machine ID.

Valid entries	Usage
1 to **64**	Enter the assigned signaling group number for **qsig-mwi** application type on the **Signaling Group** screen.

Valid entries	Usage
0 to **9**, *****, #	Enter up to 4-digit access code.

Remote Access

The **Remote Access** screen is used to implement the Remote Access feature. Remote Access permits a caller located outside the system to access the system through the public or private network and then use the features and services of the system.

Remote Access users can dial into the system using central office (CO), Foreign Exchange (FX), Wide Area Telecommunications trunks (WATS), and Integrated Services Digital Network Primary Rate Interface (ISDN-PRI) trunks. In addition, a dedicated Remote Access Direct Inward Dialing number can be provided.

SECURITY ALERT:

Avaya designed the Remote Access feature incorporated in this product that, when properly administered by the customer, enables the customer to minimize the ability of unauthorized persons to gain access to the network. It is the customer's responsibility to take the appropriate steps to properly implement the features, evaluate and administer the various restriction levels, protect access codes and distribute them only to individuals who have been advised of the sensitive nature of the access information. Each authorized user should be instructed concerning the proper use and handling of access codes.

In rare instances, unauthorized individuals make connections to the telecommunications network through use of remote access features. In such an event, applicable tariffs require the customer pay all network charges for traffic. Avaya cannot be responsible for such charges, and will not make any allowance or give any credit for charges that result from unauthorized access.

To ensure the security of your system, consider the following:

- Make all remote access facilities unlisted directory telephone numbers.
- Require users to enter a Barrier Code of at least seven random digits AND an Authorization Code of at least 13 random digits to make network calls.
- Make Authorization Codes nonconsecutive (random) and change them, at least, quarterly.
- Deactivate Authorization Codes immediately if the user leaves the company or changes assignments.
- Assign the minimum level of calling permissions required to each Authorization Code.
- Block off-hours and weekend remote access calling, when possible. Use Alternative Facility Restriction Levels, if available.
- Use a voice recording, warble tone, or no tone and avoid use of a dial tone as a prompt when the remote access unit answers.
- Assign the lowest possible FRL to only allow calls internal to Avaya Communication Manager.

As an additional step to ensure System security, you can permanently disable the Remote Access feature if you do not intend to use it now or in the future. If you do decide to permanently disable the feature, it will require Avaya Services intervention to activate the feature again.

CAUTION:

Your attempt to disable the Remote Access feature will be lost if the server running Communication Manager is rebooted without saving translations. Therefore, execute a "save translation" command after permanently disabling the Remote Access feature.

To assist you in maintaining the security of your system, Avaya Communication Manager provides the status remote access command, which provides status and information on each remote access barrier code and on the remote access feature. A sample **Status Remote Access** screen follows the **Remote Access** screen.

Field descriptions for page 1

Figure 486: Remote Access screen

```
change remote-access

                                REMOTE ACCESS
   Remote Access Extension: _________              Barrier Code Length:____
Authorization Code Required? y              Remote Access Dial Tone: n

     Barrier      COR  TN  COS   Expiration    No. of     Calls
      Code        COR  TN  COS      Date       Calls      Used
   1:_________    1__  1_  1__   __/__/__      _______    _______
   2:_________    1__  1_  1__   __/__/__      _______    _______
   3:_________    1__  1_  1__   __/__/__      _______    _______
   4:_________    1__  1_  1__   __/__/__      _______    _______
   5:_________    1__  1_  1__   __/__/__      _______    _______
   6:_________    1__  1_  1__   __/__/__      _______    _______
   7:_________    1__  1_  1__   __/__/__      _______    _______
   8:_________    1__  1_  1__   __/__/__      _______    _______
   9:_________    1__  1_  1__   __/__/__      _______    _______
  10:________     1__  1_  1__   __/__/__      _______    _______

Permanently Disable? __ Disable Following A Security Violation? y
(NOTE:  You must logoff to effect permanent disabling of Remote Access)
```

Figure 487: Remote Access Status screen

```
status remote-access

                              REMOTE ACCESS STATUS

                     Remote Access Status: enabled
                       Date/Time Modified: 01/30/95 17:00

   Barrier  Date     Expiration No. of Calls Status Date/Time Expired Cause
   Code     Modified Date       Calls  Used
 1:2374745  01/30/95 03/31/95   50     50    expired 02/15/95 20:43   calls
 2:3374837  01/30/95   /  /     20     4     active    /  /
 3:3285038  01/30/95 01/31/96          13    expired 02/10/95 09:32   date
 4:5738557  01/30/95 07/31/95   20     20    expired 02/03/95 10:14   calls
 5:7764884  01/30/95 05/20/95          0     active    /  /     :
 6:           /  /     /  /                          /  /     :
 7:           /  /     /  /                          /  /     :
 8:           /  /     /  /                          /  /     :
 9:           /  /     /  /                          /  /     :
10:           /  /     /  /                          /  /     :
```

Authorization Code Required

When you use an authorization code in conjunction with a barrier codes it increases the security of the Remote Access feature.

Valid entries	Usage
y/n	Enter **y** to require an authorization code be dialed by Remote Access users to access the system's Remote Access facilities.

Barrier Code

You must assign a barrier code that conforms to the number entered in the **Barrier Code Length** field. You may enter up to 10 barrier codes per system. Duplicate entries are not allowed. You must keep your own records regarding the distribution of these barrier codes to your personnel.

Valid entries	Usage
0 to **9** or blank	Enter a 4- to 7-digit number in any combination of digits.
none	Must be specified in the first **Barrier Code** field, if the **Barrier Code Length** field is blank.

Barrier Code Length

Assign a barrier code length of 7 to provide maximum security.

Valid entries	Usage
4 to **7** or blank	Enter a number to indicate the length of the barrier code.

Calls Used

This display-only field shows the number of calls placed using the corresponding barrier code. This field is incremented each time a barrier code is successfully used to access the Remote Access feature. A usage that exceeds the expected rate indicates improper use.

COR

Assign the most restrictive class of restriction (COR), that provides only the level of service required, to provide the maximum security.

Valid entries	Usage
0 to **95**	Enter the COR number associated with the barrier code that defines the call restriction features.

COS

Assign the most restrictive class of service (COS), that provides only the level of service required to provide the maximum security.

Valid entries	Usage
0 to **15**	Enter the COS number, associated with the barrier code, that defines access permissions for Call Processing features.

Disable Following a Security Violation

This field appears on the screen when the **SVN Authorization Code Violation Notification Enabled** field on the **Security-Related System Parameters** screen is set to **y**.

Valid entries	Usage
y/n	Enter **y** to disable the remote access feature following detection of a remote access security violation. The system administrator can re-enable Remote Access using the enable remote access command.

Expiration Date

Assign an expiration date based on the expected length of time the barrier code will be needed. If it is expected the barrier code is to be used for a 2-week period, assign a date two weeks from the current date. If the Expiration Date is assigned, a warning message is displayed on the **System Copyright** screen seven days prior to the expiration date. The system administer can modify the expiration date to extend the time interval if needed.

Valid entries	Usage
A date greater than the current date or blank	Enter the date you want the barrier code to expire.

No. of Calls

The **Expiration Date** and **No. of Calls** fields can be used independently or in conjunction to provide the maximum security. If both the **Expiration Date** and **No. of Calls** fields are assigned, the corresponding barrier code expires when the first of these criteria is satisfied.

Valid entries	Usage
1 to **9999** or blank	Enter the number of Remote Access calls that can be placed using the associated barrier code.

Permanently Disable

Reactivation of remote access to the interface requires Avaya Services intervention.

Valid entries	Usage
y/n	Enter **y** to permanently block remote access to the administration interface.

Remote Access Dial Tone

Set this field to **n** for maximum security. This field appears when the **Authorization Code Required** field is set to y.

Valid entries	Usage
y/n	Enter **n** so that there is no Remote Access Dial Tone prompt.

Remote Access Extension

The remote access extension is used as if it was a DID extension. Only one DID extension can be assigned as the remote access extension. Calls to that number are treated the same as calls on the remote access trunk.

When a trunk group is dedicated to Remote Access, the remote access extension number is administered on the trunk group's incoming destination field.

Valid entries	Usage
Extension number	Enter the extension number for Remote Access associated with each trunk that supports the Remote Access feature. You cannot assign a Vector Directory Number (VDN) extension as the remote access extension. May be blank if no barrier codes.

TN

Valid entries	Usage
1 to **20** (DEFINITY CSI, SI) **1** to **100** (DEFINITY R, S8300 Media Server, S8700 or S8710 IP-Connect, S8700 or S8710 Multi-Connect))	Enter the appropriate Tenant Partition number.

Related Topics

See Setting up Remote Access on page 380 for step-by-step instructions for configuring remote access.

See "Remote Access" in *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205, for more information.

Remote Call Coverage Table

The Remote Call Coverage Table allows you to provide automatic redirection of certain calls to alternate non-local answering positions in a coverage path.

Non-local numbers may be any ARS or AAR number, any number on the public network, any international number, or a UDP/DCS extension up to 16 digits or blank, which includes any ARS/AAR facility access code, any trunk dial access code (TAC), long distance dialing code, or international dial code. Up to 999 remote call coverage points can be entered on the multiple pages of this screen.

Field descriptions for page 1

Figure 488: Remote Call Coverage Table screen

```
change coverage remote

                      REMOTE CALL COVERAGE TABLE
                      ENTRIES FROM 1  TO 1000

01: _________________     16: _________________     31: _________________
02: _________________     17: _________________     32: _________________
03: _________________     18: _________________     33: _________________
04: _________________     19: _________________     34: _________________
05: _________________     20: _________________     35: _________________
06: _________________     21: _________________     36: _________________
07: _________________     22: _________________     37: _________________
08: _________________     23: _________________     38: _________________
09: _________________     24: _________________     39: _________________
10: _________________     25: _________________     40: _________________
11: _________________     26: _________________     41: _________________
12: _________________     27: _________________     42: _________________
13: _________________     28: _________________     43: _________________
14: _________________     29: _________________     44: _________________
15: _________________      30: _________________      45: _________________
```

01 to 1000

Valid entries	Usage
0 to **9** or blank	Enter the destination up to 16 digits.

Remote Office

This screen supports an arrangement whereby a user can set up a remote office without having an on-premises physical desk-set. An R300 is sued to connect remote DCP and analog phones, IP telephones, and H.323 trunks to the DEFINITY gatekeeper via IP.

Field descriptions for page 1

Figure 489: Remote Office screen

```
change remote office x

                              REMOTE OFFICE

     Node Name: _________________
Network Region: ___
      Location: __
     Site Data: _____________________________________
                _____________________________________
                _____________________________________
```

Location

This field appears on the screen when the **SVN Authorization Code Violation Notification Enabled** field on the **Security-Related System Parameters** screen is set to **y**.

Valid entries	Usage
1 to **64**	Specify the location (comprised of the associated time zone and the appropriate numbering plan).

Network Region

This field appears on the screen when the **SVN Authorization Code Violation Notification Enabled** field on the **Security-Related System Parameters** screen is set to **y**.

Valid entries	Usage
1 to **250** or blank	Specify the network region to be assigned to all stations supported on this remote office.

Node Name

This field appears on the screen when the **SVN Authorization Code Violation Notification Enabled** field on the **Security-Related System Parameters** screen is set to **y**.

Valid entries	Usage
character string	Specify the node name of the remote office.

Site Data

This field appears on the screen when the **SVN Authorization Code Violation Notification Enabled** field on the **Security-Related System Parameters** screen is set to **y**.

Valid entries	Usage
30 characters or blank	Any desired information.

RHNPA Table

The **RHNPA Table** defines route patterns for specific 3-digit codes, usually direct distance dialing (DDD) prefix numbers. The appearance of the screen is different slightly depending on the type of media server or switch.

Figure 490: RHNPA Table screen

```
change rhnpa                                                  Page 1 of X
                                RHNPA TABLE: __
                                CODES: 000-999

                             Pattern Choices
          1: ___   3: ___   5: ___   7: ___   9: ___   11: ___
          2: ___   4: ___   6: ___   8: ___  10: ___   12: ___

       Code - Pattern Choice Assignments (from 1-12 above)
00: 1__ 10: 1__ 20: 1__ 30: 1__ 40: 1__ 50: 1__ 60: 1__ 70: 1__ 80: 1__ 90: 1__
01: 1__ 11: 1__ 21: 1__ 31: 1__ 41: 1__ 51: 1__ 61: 1__ 71: 1__ 81: 1__ 91: 1__
02: 1__ 12: 1__ 22: 1__ 32: 1__ 42: 1__ 52: 1__ 62: 1__ 72: 1__ 82: 1__ 92: 1__
03: 1__ 13: 1__ 23: 1__ 33: 1__ 43: 1__ 53: 1__ 63: 1__ 73: 1__ 83: 1__ 93: 1__
04: 1__ 14: 1__ 24: 1__ 34: 1__ 44: 1__ 54: 1__ 64: 1__ 74: 1__ 84: 1__ 94: 1__
05: 1__ 15: 1__ 25: 1__ 35: 1__ 45: 1__ 55: 1__ 65: 1__ 75: 1__ 85: 1__ 95: 1__
06: 1__ 16: 1__ 26: 1__ 36: 1__ 46: 1__ 56: 1__ 66: 1__ 76: 1__ 86: 1__ 96: 1__
07: 1__ 17: 1__ 27: 1__ 37: 1__ 47: 1__ 57: 1__ 67: 1__ 77: 1__ 87: 1__ 97: 1__
08: 1__ 18: 1__ 28: 1__ 38: 1__ 48: 1__ 58: 1__ 68: 1__ 78: 1__ 88: 1__ 98: 1__
09: 1__ 19: 1__ 29: 1__ 39: 1__ 49: 1__ 59: 1__ 69: 1__ 79: 1__ 89: 1__ 99: 1__
```

CODES

Display-only field showing the desired 100-block, for example 000 through 099 or 900 through 999 based upon the **change rhnpa** command. A separate screen displays for each 100-block.

Code-Pattern Choice Assignments

Valid entries	Usage
1 to **12**	Enter a Pattern Choice number from the list above to choose the route pattern for each prefix code. For Server R.
1 to **24**	For S8700 or S8710 IP-Connect.

Pattern Choices

There are 12 pattern choices for DEFINITY R, CSI, and SI; there are 24 pattern choices for the S8300 Media Server, S8700 or S8710 IP-Connect, and S8700 or S8710 Multi-Connect. Enter the route pattern number you want associated with each code. The pattern choice you list on one screen automatically defaults to the other screens of the same table. If you use one pattern for most of the codes, assign that pattern to choice 1.

Valid entries	Usage
1 to **640** or blank	For DEFINITY R.
1 to **999** or blank	For S8300 Media Server, S8700 or S8710 IP-Connect, and S8700 or S8710 Multi-Connect
1 to **254**	For DEFINITY SI and CSI.

RHNPA TABLE

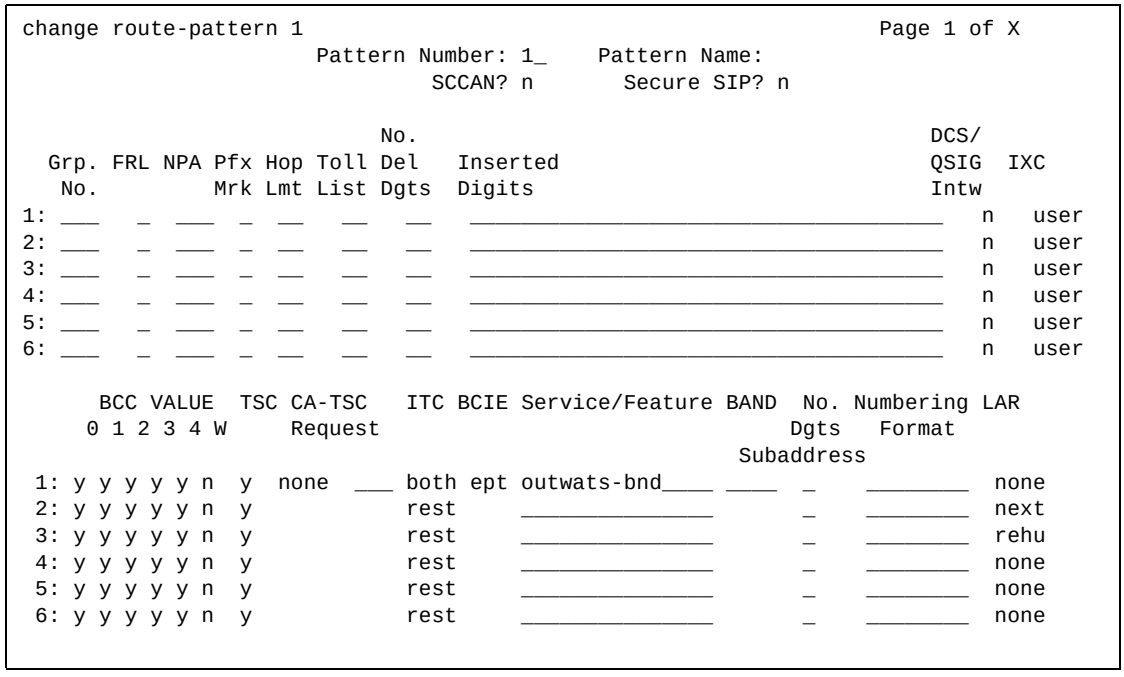
Display-only field indicating the table number.

Route Pattern

The **Route Pattern** screen defines the route patterns used by your server running Communication Manager. Each route pattern contains a list of trunk groups that can be used to route the call. The maximum number of route patterns and trunk groups allowed depends on the configuration and memory available in your system.

Use this screen to insert or delete digits so AAR or ARS calls route over different trunk groups. You can convert an AAR number into an international number, and insert an area code in an AAR number to convert an on-network number to a public network number. Also, when a call directly accesses a local central office (CO), if the long-distance carrier provided by your CO is not available, Avaya Communication Manager can insert the dial access code for an alternative carrier into the digit string.

Figure 491: Route Pattern screen



```
change route-pattern 1                                          Page 1 of X
                     Pattern Number: 1_    Pattern Name:
                               SCCAN? n       Secure SIP? n

                           No.                                      DCS/
  Grp. FRL NPA Pfx Hop Toll Del   Inserted                          QSIG  IXC
   No.         Mrk Lmt List Dgts  Digits                           Intw
1: ___   _  ___  _  __   __   __   ____________________________________   n   user
2: ___   _  ___  _  __   __   __   ____________________________________   n   user
3: ___   _  ___  _  __   __   __   ____________________________________   n   user
4: ___   _  ___  _  __   __   __   ____________________________________   n   user
5: ___   _  ___  _  __   __   __   ____________________________________   n   user
6: ___   _  ___  _  __   __   __   ____________________________________   n   user

     BCC VALUE  TSC CA-TSC   ITC BCIE Service/Feature BAND  No. Numbering LAR
    0 1 2 3 4 W     Request                                  Dgts   Format
                                                          Subaddress
 1: y y y y y n  y  none  ___ both ept outwats-bnd____ ____  _    _________  none
 2: y y y y y n  y           rest     ________________       _    _________  next
 3: y y y y y n  y           rest     ________________       _    _________  rehu
 4: y y y y y n  y           rest     ________________       _    _________  none
 5: y y y y y n  y           rest     ________________       _    _________  none
 6: y y y y y n  y           rest     ________________       _    _________  none
```

Band

Enter a number that represents the OUTWATS band number (US only).

WATS is a voice-grade service that provides both voice and low-speed data transmission calls to defined areas (bands) for a flat rate charge.

This field appears when the **Services/Features** field is **outwats-bnd** and when **ISDN-PRI** or **ISDN-BRI Trunks** field is **y** on the **System-Parameters Customer-Options** screen. **Band** is required by Call-by-Call Service Selection.

BCC Value

Bearer Capability Class (BCC) identifies the type of call appropriate for this trunk group, such as voice calls and different types of data calls. This field appears when the **ISDN-PRI** or **ISDN-BRI Trunks** field is **y** on the **System-Parameters Customer-Options** screen.

Valid entries	Usage
y/n	Enter **y** in appropriate BCC column (0, 1, 2, 3, 4, or W) if the BCC is valid for the associated route pattern. A trunk group preference may have more than one BCC.

The following table explains BCC values

:

BCC Value	Description
0	Voice-Grade Data and Voice
1	56-kbps Data (Mode 1)
2	64-kbps Data (Mode 2)
3	64-kbps Data (Mode 3)
4	64-kbps Data (Mode 0)
W	128 to 1984-kbps Data (Wideband)

BCIE (Bearer Capability Information Element)

This field applies to ISDN trunks and appears if ITC is **both**.

Valid entries	Usage
ept (endpoint)	Use BCIE to determine how to create the ITC codepoint in the setup message.
unr (unrestricted)	

CA-TSC Request

Use CA-TSC on ISDN B-channel connections.

Valid entries	Usage
as-needed	The CA-TSC is set up only when needed. This causes a slight delay. Avaya recommends this entry for most situations.
at-setup	The CA-TSC is automatically set up for every B-channel call whether or not it is needed.
none	No CA-TSC is set up. Permits tandeming of NCA-TSC setup requests0.

DCS/QSIG Intw

This field only appears if the **Interworking with DCS** field on the **System-Parameters Customer-Options** screen is set to **y**.

Valid entries	Usage
y/n	Enter **y** to enable DCS/QSIG Voice Mail Interworking.

FRL

Valid entries	Usage
0 to **7**	Enter the Facility Restriction Level (FRL) associated with the entries on this row (preference). **0** is the least restrictive and **7** is the most restrictive. The calling party's FRL must be greater than or equal to this FRL to access the associated trunk-group.

SECURITY ALERT:
For system security reasons, Avaya recommends using the most restrictive FRL possible.

Grp No

Valid entries	Usage
1 to **666**	Enter the trunk group number associated with this row (preference). For DEFINITY R, CSI, SI.
1 to **2000**	For S8300 Media Server, S8700 or S8710 IP-Connect, S8700 or S8710 Multi-Connect.

Hop Lmt

Enter the number of hops for each preference. A hop is when a call tandems through a media server or switch to another destination. Limiting the number of hops prevents circular hunting, which ties up trunk facilities without ever completing the call. Avaya Communication Manager blocks a hop equal to or greater than the number you enter

.

Valid entries	Usage
blank	Indicates that there is no limit to the number of hops for this preference.
1 to **9**	To limit the number of hops if using the tandem hop feature.
1 to **32**	If using the transit feature.

Inserted Digits

Enter the digits you want inserted for routing. Communication Manager can send up to 52 digits. This includes up to 36 digits you may enter here plus up to 18-digits originally dialed. Special symbols count as two digits each.

Valid entries	Usage
0 to 36 digits (**0** to **9**)	Enter the digits you want inserted for routing.
*****	When * is in the route pattern and the outgoing trunk is signaling type "mf", the MFC tone for the "end-of-digits" is sent out to the CO in place of the *.
#	When # is in the route pattern and the outgoing trunk is signaling type "mf", the MFC tone for the "end-of-digits" is sent out to the CO in place of the #.

1 of 2

Valid entries	Usage
‘,’	Use 2 places. Creates a 1.5 second pause between digits being sent. Do not use as the first character in the string unless absolutely necessary. Misuse can result in some calls, such as Abbreviated Dialing or Last Number Dialed, not completing.
+	Wait for dial tone up to the Off Premises Tone Detection Timer and then send digits or intercept tone based on **Out Pulse Without Tone** y/n on the **Feature-Related System Parameters** screen.
%	Start End-to-End Signaling.
!	Wait for dial tone without timeout and then send DTMF digits.
&	Wait for ANI (used for Russian pulse trunks)
p	The associated trunk group must be of type **sip**. Enter the single digit **p** for fully qualified E.164 numbers. The p is translated to a + and is prepended to the digit string.

2 of 2

ITC (Information Transfer Capability)

Use Information Transfer Capability (**ITC**) to identify the type of data transmission or traffic that this routing preference can carry. The ITC applies only to data calls (BCC 1 through 4).

This field must be **unre** or **both** if the **BCC** is **y** and the **BCC** value is **W**

Valid entries	Usage
both	Calls from restricted and unrestricted endpoints can access the route pattern.
rest(ricted)	Calls from restricted endpoints can access the route pattern.
unre(stricted)	Calls from unrestricted endpoints can access the route pattern.

IXC

Inter-Exchange Carrier (**IXC**) identifies the carrier, such as AT&T, used for calls that route via an IXC, and for Call Detail Recording (CDR).

This field appears when the **ISDN-PRI** or **ISDN-BRI Trunks** field is **y** on the **System-Parameters Customer-Options** screen.

Valid entries	Usage
Valid carrier code	Identifies the carrier for IXC calls
user	For presubscribed carrier. Used when an IXC is not specified.
none	This field must be **none** for non-ISDN trunk groups and for Bellcore NI-2 Operator Service Access. If you need to send an IXC code for a non-ISDN trunk group, enter the IXC code in the Inserted Digits field.

LAR

Enter the routing-preference for Look Ahead Routing.

Valid entries	Usage
next	Go to the next routing preference and attempt the call again.
rehu	Rehunt within the current routing-preference for another trunk to attempt the call again.
none	Look Ahead Routing is not enabled for the preference.

No. Del. Digits

Use this field to modify the dialed number so an AAR or ARS call routes over different trunk groups that terminate in servers/switches with different dial plans.

Valid entries	Usage
0 to **28** or blank	Enter the total number of digits you want the system to delete before it sends the number out on the trunk. Use for calls that route: • to or through a remote media server or switch • over tie trunks to a private network server or switch • over Central Office (CO) trunks to the serving CO

No. Dgts Subaddress

Allows a caller to reach a number where the media server's digit processing deletes the dialed number and inserts the listed directory number (LDN). The LDN then is sent to the destination address and the dialed extension is sent in the calling party subaddress information element (IE). At the receiving end, the call terminates to the user indicated by the subaddress number instead of the attendant. Administrable when, on the **System-Parameters Customer-Options** screen, the **ISDN Feature Plus** field is **y**.

Valid entries	Usage
1 to **5** blank	Enter the number of dialed digits to send in the calling party subaddress IE.

NPA

This entry is not required for AAR.

Valid entries	Usage
3-digit number	Enter the 3-digit Numbering Plan Area (NPA) (or area code) for the terminating endpoint of the trunk group. Call your local telephone company to verify this number if you need help. For WATS trunks, the terminating NPA is the same as the home NPA unless the Local Exchange Carrier requires 10 digits for local NPA calls.
blank	For AAR calls and for tie trunks

Numbering Format

This field applies only to ISDN trunk groups. Enter a value from table below. This field specifies the format of the routing number used for the trunk group for this preference.

This field appears when the **ISDN-PRI** or **ISDN-BRI Trunks** field is **y** on the **System-Parameters Customer-Options** screen.

Valid entries	Numbering Plan Identifier	Type of Numbering
blank	E.164(1)	1-MAX
natl-pub	E.164(1)	national(2)
intl-pub	E.164(1)	international(1)
locl-pub	E.164(1)	local/subscriber(4)
pub-unk	E.164(1)	unknown(0)
lev0-pvt	Private Numbering Plan - PNP(9)	local(4)
levl0-pvt (enter to allow Network Call Redirection/ Transfer		
lev1-pvt	Private Numbering Plan - PNP(9)	Regional Level 1(2)
lev2-pvt	Private Numbering Plan - PNP(9)	Regional Level 2(1)
unk-unk	unknown(0)	unknown(0)

Note:

To access Bellcore NI-2 Operator Service Access, the **Inserted Digits** field must be **unk-unk**.

Pattern Name

Enter an alphanumeric name for identification purposes.

Pattern Number

This display-only field shows the route pattern number (**1** to **640**).

Prefix Mark

This entry is not required for AAR. For ARS, enter a number from **0** to **4** or blank.

Prefix Marks set the requirements for sending a prefix digit 1, indicating a long-distance call. Prefix Marks apply to 7- or 10-digit Direct Distance Dialing (DDD) public network calls. A prefix digit 1 is sent only when call type is foreign number plan area (FNPA) or home numbering plan area (HNPA) in the ARS Digit Analysis table.

For a WATS trunk, the Prefix Mark is the same as the local CO trunk.

Valid entries	Usage
0	• Suppress a user-dialed prefix digit 1 for 10-digit FNPA calls. • Leave a user-dialed prefix digit 1 for 7-digit HNPA calls. • Leave a prefix digit 1 on 10-digit calls that are not FNPA or HNPA calls. Do not use **Prefix Mark 0** in those areas where all long-distance calls must be dialed as 1+10 digits. Check with your local network provider.
1	• Send a 1 on 10-digit calls, but not on 7-digit calls. Use Prefix Mark 1 for HNPA calls that require a 1 to indicate long-distance calls.
2	• Send a 1 on all 10-digit and 7-digit long-distance calls. Prefix Mark 2 refers to a Toll Table to define long distance codes.
3	• Send a 1 on all long-distance calls and keep or insert the NPA (area code) so that all long distance calls are 10-digit calls. The NPA is inserted when a user dials a Prefix digit 1 plus 7-digits. Prefix Mark 3 refers to a Toll Table to define long distance codes.
4	• Always suppress a user-dialed Prefix digit 1. Use Prefix Mark 4, for example, when ISDN calls route to a media server or a switch that rejects calls with a prefix digit 1.
blank	For tie trunks, leave this field blank.

SCCAN

This field appears when **Enhanced EC500** on the **System Parameters - Customer Options** screen is set to **y**.

Note:

When the **SCCAN** field is set to **y**, non-SCCAN-associated fields are hidden. Only the **Pattern Number**, **Pattern Name**, **SCCAN**, **Secure SIP**, and **Grp No** fields appear.

Valid entries	Usage
y/n	Enter **y** to indicate that this route pattern supports incoming SCCAN calls.

Secure SIP

Valid entries	Usage
y/n	Specify whether the SIP or SIPS prefix will be used, if the call is routed to a SIP trunk preference. If SIP trunks are not specified on the **Route Pattern** screen, the call will be routed over whatever trunk is specified. Therefore, to ensure a SIP TLS connection when such a route-pattern is invoked, only SIP trunks should be specified. Default is **n**.

Service/Feature

Enter up to 15 characters to identify the Service/Feature carried by the information element (IE) in a call in this route pattern. This field is required by Call-by-Call Service Selection, and Network Call Redirection and Transfer.

Service/Feature appears when **ISDN-PRI** or **ISDN-BRI Trunks** is **y** on the **System-Parameters Customer-Options** screen.

Valid entries		
accunet	**multiquest**	**sdn** (Enter to allow Network Call Redirection/ Transfer)
i800	**operator**	**sub-operator**

1 of 2

Valid entries		
inwats	**oper-lds** (operator and lds)	**sub-op-lds** (sub-operator and lds)
lds	**oper-meg** (operator and megacon)	**sub-op-meg** (sub-operator and megacom)
mega800	**oper-sdn** (operator and sdn)	**sub-op-sdn** (sub-operator and sdn)
megacom	**outwats-bnd**	**wats-max-bnd**
		2 of 2

Toll List

This entry is not required for AAR.

Valid entries	Usage
1 to **32** or blank	For ARS, enter the number of the ARS Toll Table associated with the terminating NPA of the trunk group. You must complete this field if **Prefix Mark** is **2** or **3**.

TSC

Set **TSC** to **y** for feature transparency on DCS+ calls and to use QSIG Call Completion.

Valid entries	Usage
y/n	Enter **y** to allow Call-Associated TSCs, and to allow incoming Non-Call-Associated TSC requests to be tandemed out for each preference.

Security-Related System Parameters

Use this screen to determine when Avaya Communication Manager reports a security violation. Many of the fields on this screen repeat for each type of security violation. We have explained them once here, but the usage is the same for all. See "Security Violations Notifications" in *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205, for more information.

Field descriptions for page 1

Figure 492: Security-Related System Parameters screen

```
change system-parameters security                                Page 1 of 2
                     SECURITY-RELATED SYSTEM PARAMETERS

SECURITY VIOLATION NOTIFICATION PARAMETERS

  SVN Login Violation Notification Enabled? y
        Originating Extension: _____ Referral Destination: _____
              Login Threshold: 5_          Time Interval: 0:03
       Announcement Extension: _____

  SVN Remote Access Violation Notification Enabled? y
        Originating Extension: _____ Referral Destination: _____
       Barrier Code Threshold: 10          Time Interval: 0:03
       Announcement Extension: _____

  SVN Authorization Code Violation Notification Enabled? y
        Originating Extension: _____ Referral Destination: _____
 Authorization Code Threshold: 10          Time Interval: 0:03
       Announcement Extension: _____
```

SECURITY VIOLATION NOTIFICATION PARAMETERS

SVN Login (Violation Notification, Remote Access, Authorization Code) Enabled

Valid entries	Usage
y/n	Set to **y** if you want Communication Manager to notify you when a login violation occurs. If this field is **y**, the next 5 fields appear so you can establish the parameters for what is considered a security violation.

Announcement Extension

If you enter a value in this field, the server running Communication Manager *calls the referral destination, then plays this announcement upon answer.*

Valid entries	Usage
Valid extension	The announcement extension where SVN violation announcement resides.

Login (Barrier Code, Authorization Code) Threshold

The value assigned to this field, in conjunction with **Time Interval**, determines whether a security violation has occurred. For example, if this field is 5, and time interval is 0:03, then five invalid access attempts within three minutes constitutes a security violation.

Valid entries	Usage
1 to **99**, **1** to **255** for Station Security code	Enter the number of access attempts that are permitted before a referral call is made. *In general, it is good to keep this number low. If you are doing testing and do not want to generate alarms, you might change this threshold temporarily.*

Originating Extension

The originating extension initiates the referral call in the event of a security violation. It also sends the appropriate alerting message or display to the referral destination.

Valid entries	Usage
An unassigned extension	If you establish notification for more than one type of security violations, you must assign a different extension to each one. When Communication Manager generates a referral call, *this extension and the type of violation appear on the display at the referral destination.*

Referral Destination

The referral destination receives the referral call when a security violation occurs. The referral destination telephone must have a display, unless the you assign an Announcement Extension.

Valid entries	Usage
An extension	Enter the extension of the telephone, attendant console, or vector directory number (VDN) that you want to receive the referral call for each type of violation. This can be the same extension for all type of violations. If you use a VDN, you must complete the Announcement Extension field. You can also use Call Vectoring Time-of-Day routing to route the referral call to different destinations based on the time of day or the day of the week.

Time Interval

The value of this field, in conjunction with **Login Threshold**, determines whether a security violation has occurred.

Valid entries	Usage
0:01 to **7:59**	The range for the time interval is one minute to eight hours. Entered in the screen **x:xx**. For example, if you want the time interval to be one minute, you enter 0:01. If you want the time interval to be seven and one-half hours, you enter 7:30.

SVN Remote Access Violation Notification Enabled

Use the **SVN Remote Access Violation Notificatioin Enabled** and the **SVN Authorization Code Violation Notification Enabled** fields to establish parameters for remote access security violations. A remote access violation occurs if a user enters incorrect barrier codes. You cannot set the system to disable remote access following a security violation unless you have turned these fields on.

SVN Authorization Code Violation Notification Enabled

Use the **SVN Remote Access Violation Notificatioin Enabled** and the **SVN Authorization Code Violation Notification Enabled** fields to establish parameters for remote access security violations. A remote access violation occurs if a user enters incorrect barrier codes. You cannot set the system to disable remote access following a security violation unless you have turned these fields on..

Field descriptions for page 2

Figure 493: Security-Related System Parameters screen (for DEFINITY R)

```
change system-parameters security                                Page 2 of 2
                    SECURITY-RELATED SYSTEM PARAMETERS

  SECURITY VIOLATION NOTIFICATION PARAMETERS

    SVN Station Security Code Violation Notification Enabled? y
           Originating Extension: _____      Referral Destination: _____
  Station Security Code Threshold: 10               Time Interval: 0:03
           Announcement Extension: _____

  STATION SECURITY CODE VERIFICATION PARAMETERS

                       Minimum Station Security Code Length: 4
    Security Code for Terminal Self Administration Required? y

  ACCESS SECURITY GATEWAY PARAMETERS

    SYSAM-LCL? n      SYSAM-RMT? n
        MAINT? n       SYS-PORT? n
  Translation-ID Number Mismatch Interval (days): _
```

Figure 494: Security-Related System Parameters screen (for DEFINITY SI and CS)

```
change system-parameters security                              Page 2 of x
                    SECURITY-RELATED SYSTEM PARAMETERS

  SECURITY VIOLATION NOTIFICATION PARAMETERS

    SVN Station Security Code Violation Notification Enabled? y
            Originating Extension: _____      Referral Destination: _____
  Station Security Code Threshold: 10               Time Interval: 0:03
           Announcement Extension: _____

  STATION SECURITY CODE VERIFICATION PARAMETERS

                       Minimum Station Security Code Length: 4
    Security Code for Terminal Self Administration Required? y

  ACCESS SECURITY GATEWAY PARAMETERS

        MGR1? n    INADS? n
         EPN? n      NET? n
```

SECURITY VIOLATION NOTIFICATION PARAMETERS

SVN Station Security Code Violation Notification Enabled

Station Security codes are used to validate logins to a particular extension (for example, a home agent using an extender, or two part-time workers using the same telephone, but different extensions, through personal station access.) Enter **y** *here to establish parameters for this.*

STATION SECURITY CODE VERIFICATION PARAMETERS

Minimum Station Security Code Length

This determines the minimum required length of the Station Security Codes that you enter on the **Station** screen.

Valid entries	Usage
3 to **8**	Longer codes are more secure. If station security codes are used for external access to telecommuting features, the minimum length should be 7 or 8.

Security Code for Terminal Self Administration Required

Specifies if a Personal Station access code is required to enter the Self-Administration mode.

Valid entries	Usage
y/n	Enter **y** to indicate that a security code is required.

ACCESS SECURITY GATEWAY PARAMETERS

These eight fields appear only if the **Access Security Gateway (ASG)** field on the **System-Parameters Customer-Options** screen is **y**.

The following 4 fields display only for the DEFINITY R version:

MAINT

A direct connection to the EPN maintenance circuit pack RS-232 interface.

Valid entries	Usage
y/n	Any entry attempt through this port receives a challenge response.

SYSAM-LCL

A direct cable connection to the SYSAM-LCL G3-MT (local system administrator's terminal) port on the System Access and Maintenance circuit pack on the active DEFINITY R processor carrier. For more information on the circuit pack, see the *Hardware Guide for Avaya Communication Manager*.

Valid entries	Usage
y/n	Any entry attempt through this port receives a challenge response.

SYS-PORT

A dialed-in (or out) connection to the System Access Port.

Valid entries	Usage
y/n	Any entry attempt through this port receives a challenge response.

SYSAM-RMT

A dialed-in (or out) connection to the SYSAM-RMT port on the System Access and Maintenance circuit pack in the active DEFINITY R processor carrier. For more information on the circuit pack, see the *Hardware Guide for Avaya Communication Manager*.

Valid entries	Usage
y/n	Any entry attempt through this port receives a challenge response.

The following fields display only for the DEFINITY SI and CSI versions:

EPN

A direct connection to the Expansion Port Network.

Valid entries	Usage
y/n	Any entry attempt through this port receives a challenge response.

INADS

A direct cable connection to the Initialization and Administration System used to remotely initialize and administer Avaya Communication Manager.

Valid entries	Usage
y/n	Any entry attempt through this port receives a challenge response.

MGR1

The direct connect system administration and maintenance access interface located on the processor circuit pack. For more information on the circuit pack, see the *Hardware Guide for Avaya Communication Manager*.

Valid entries	Usage
y/n	Any entry attempt through this port receives a challenge response.

NET

A dialed-in (or out) connection to the Network Controller circuit pack. For more information on the circuit pack, see the *Hardware Guide for Avaya Communication Manager*.

Valid entries	Usage
y/n	Any entry attempt through this port receives a challenge response.

Translation-ID Number Mismatch Interval (days)

A display-only field for all logins except init; only init logins can change this field. This field specifies the interval (in days) that the system allows a mismatch between the translation ID stored in the Processor circuit pack(s) and on the flash card. Following expiration of this interval, the ability to execute system administration commands that modify translation data is denied for all logins, except for init.

Valid entries	Usage
1 to **90**	Enter a number to indicate the number of days the system allows access to system administration commands.

Signaling Group

Use this screen to establish signaling group parameters for ISDN-PRI, H.323, ATM, and SIP trunks. Because these trunk types vary in the types of parameters needed, the fields that appear on this screen change depending on the value of the **Group Type** field. Field descriptions are alphabetized for easier reference.

Field descriptions for page 1

Figure 495: Signaling Group screen when the Group Type field is atm

```
                                                          Page 1 of 5
                              SIGNALING GROUP

Group Number  ___           Group Type:  atm___       Name:
                                                Max Number of NCA TSC: ___
                             D-Channel:          Max number of CA TSC:  ___
                                               Trunk Group for NCA TSC:  ___
     Trunk Group for Channel Selection:  ___
        Supplementary Service Protocol: _      Network Call Transfer? n

CIRCUIT PARAMETERS
     Virtual Path Identifier: 0
  Virtual Channel Identifier: 0

              Signaling Mode: isdn-pri           Circuit Type: T1
                   Idle Code: 11111111                Connect: network
        Interface Companding: mulaw
            Country Protocol: 1
            Protocol Version: d

                                           Interworking Message: PROGress
```

Figure 496: Signaling Group screen when the Group Type field is h.323

```
                                                          Page 1 of 5
                              SIGNALING GROUP

Group Number  ___           Group Type:  h.323
                        Remote Office? n      Max Number of NCA TSC: ___
                                  SBS? y       Max number of CA TSC:  ___
                                              Trunk Group for NCA TSC:  ___
     Trunk Group for Channel Selection:  ___
        Supplementary Service Protocol: _
                       T303 Timer(sec): 10

      Near-end Node Name:              Far-end Node Name:
    Near-end Listen Port:            Far-end Listen Port:
                                   Far-end Network Region:
            LRQ Required? n      Calls Share IP Signaling Connection? n
       RRQ Required? n

Media Encryption? y                  Bypass If IP Threshold Exceeded? y
   Passphrase: ____________________________

DTMF over IP:                                    Direct IP-IP Audio Connections? y
                                               IP Audio Hairpining? y
                                              Interworking Message: Progress
```

Figure 497: Signaling Group screen when the Group Type field is isdn-pri

```
                                                          Page 1 of x
                              SIGNALING GROUP
Group Number  ___           Group Type:  isdn-pri
                  Associated Signaling?         Max Number of NCA TSC: ___
                     Primary D-Channel:          Max number of CA TSC:  ___
                                                Trunk Group for NCA TSC:  ___
     Trunk Group for Channel Selection:  ___ X-Mobility/Wireless Type: NONE
        Supplementary Service Protocol: _
```

Figure 498: Signaling Group screen when the Group Type field is sip

```
                                                                  Page 1 of x
                                SIGNALING GROUP

Group Number  ___           Group Type: sip
                        Transport Method: tls




  Near-end Node Name:               Far-end Node Name:
Near-end Listen Port: 5061       Far-end Listen Port: ____
                                Far-end Network Region: __
      Far-end Domain: ________________________________


                              Bypass If IP Threshold Exceeded? y

       DTMF over IP: rtp-payload             Direct IP-IP Audio Connections?
y
                                                          IP Audio Hairpinning?
y
```

Associated Signaling

Appears when the **Group Type** field is **isdn-pri**.

Valid entries	Usage
y	Enter **y** to use associated signaling.
n	Enter **n** to use non-facility associated signaling.

Bypass If IP Threshold Exceeded

Appears when the **Group Type** field is **h.323** or **sip**.

Valid entries	Usage
y/n	Enter **y** to automatically remove from service the trunks assigned to this signaling group when IP transport performance falls below limits administered on the **Maintenance-Related System Parameters** screen.

Calls Share IP Signaling Connection

Appears when the **Group Type** field is **h.323** or **sip**.

Valid entries	Usage
y/n	Enter **y** for inter-connection between servers running Avaya Communication Manager. If **y**, then "LRQ Required" must be n. Enter **n** if the local and/or remote media server or switch is not running Avaya Communication Manager.

Circuit Type

Appears when the **Group Type** field is **atm**.

Valid display	Usage
T1	Results in page 2 displaying 24 channels.
E1	Results in page 2 displaying 31 channels.

Connect

Appears when the **Group Type** field is **atm**. In order to control communications at layers 2 and 3 of the ISDN-PRI protocol, use this field to specify what is on the far end of this link.

Valid entries	Usage
host	Enter **host** when the link connects Communication Manager to a computer.
network	Enter **network** when the link connects Communication Manager to a central office or any other public network switch.
pbx	Enter **pbx** if this link is connected to another switch in a private network. If **pbx** is entered, the **Interface** field appears.

Country Protocol

Appears when the **Group Type** field is **atm**. The entry in this field must match the country protocol used by the far-end server. For connections to a public network, your network service provider can tell you which country protocol they are using.

Valid entries	Usage
1 to **25**	Enter the country protocol used by the central office at which this link terminates.
etsi	Enter **etsi** if your network service provider uses the protocol of the European Telecommunications Standards Institute (ETSI). Enter **etsi** only if the **Signaling Mode** field is **isdn-pri**.

D Channel

Appears when the **Group Type** field is **atm**. Enter the necessary characters.

Valid entries	Usage
01 to **44** (DEFINITY R) or **01** to **03** (DEFINITY CSI, SI) or **1** to **64** (S8700 or S8710 IP-Connect / S8700 or S8710 Multi-Connect / S8300 Media Server)	First and second characters are the cabinet number
A to **E**	Third character is the carrier
0 to **20**	Fourth and fifth character are the slot number
01 to **04** (Analog TIE trunks) **09** to **32**	Six and seventh characters are the circuit number

Direct IP-IP Audio Connections

Appears when the **Group Type** field is **h.323** or **sip**. Allows direct audio connections between H.323 endpoints. Initially not available for SIP.

Valid entries	Usage
y/n	Enter to y to save on bandwidth resources and improve sound quality of voice over IP transmissions. Initially, support for SIP Trunks requires the default of **n**.

DTMF Over IP

Appears when the **Group Type** field is **sip**.

Valid entries	Usage
rtp-payload	Initially, support for SIP trunks requires the default entry of **rtp-payload**.

Far-end Domain

Appears when the **Group Type** field is **sip**. The number of the network region that is assigned to the far-end of the trunk group.

Valid entries	Usage
Max. 40 character string	Enter the name of the IP domain for which the far-end proxy is responsible (i.e., authoritative), if different than the near-end domain. If the domains are the same, leave this blank.
blank	Far-end domain is unspecified. Note that If you leave this field blank, the system might display the following message: "Warning: unspecified far-end IP address is vulnerable to denial of service attacks."

Far-end Listen Port

Appears when the **Group Type** field is **h.323** or **sip**.

Valid entries	Usage
1 to **65535**	Enter the same number as entered in the **Near-end Listen Port** field. Typically, this is the default of 5061 for SIP over TLS.
blank	Far-end listen port is unspecified. Note that If you leave this field blank, the system might display the following message: "Warning: unspecified far-end IP address is vulnerable to denial of service attacks."

Far-end Network Region

Appears when the **Group Type** field is **h.323** or **sip**. The number of the network region that is assigned to the far-end of the trunk group.

Valid entries	Usage
1 to **250**	Enter the network region number that is assigned to the far end of the trunk group. The region is used to obtain the codec set used for negotiation of trunk bearer capability. Leave blank to select the region of the near-end node.
blank	Far-end network region is unspecified. Note that If you leave this field blank, the system might display the following message: "Warning: unspecified far-end IP address is vulnerable to denial of service attacks."

Far-end Node Name

Appears when the value of the entry in the **Group Type** field is **atm** or **sip**. Enter the node name for the far-end C-LAN IP interface used for trunks assigned to this signaling group. The node name must be administered on the **IP Node Names** screen.

Valid entries	Usage
Name of an administered IP node.	Describe the far-end node. Note that If you leave this field blank, the system might display the following message: "Warning: unspecified far-end IP address is vulnerable to denial of service attacks."

Group Number

A display-only field identifying the signaling group.

Group Type

This field describes the type of protocol to be used with the signaling group.

Valid entries	Usage
atm	Use for Asynchronous Transfer Mode signaling trunks
h.323	Use for h.323 protocol or when using SBS signaling trunks.
isdn-pri	Integrated Service Digital Network Primary Rate Interface
sip	For SIP on the Avaya S8300 Media Server, S8500 Media Server, S8700 or S8710 IP-Connect, or S8700 or S8710 Multi-Connect only.

Idle Code

Appears when the **Group Type** field is **atm**. This entry sets the signal sent out over idle DS0 channels. The string must be compatible with the protocol used by the far-end switch/server.

Valid entries	Usage
0, **1**	Enter an 8-digit string.

Interface

This field only appears when the **Connect** field is **pbx**. The **Interface** field controls how your server negotiates glare with the far-end switch.

Valid entries	Usage
Use the following 2 values for private network applications in the U.S.	
network	Enter **network** if your server overrides the other end when glare occurs. If you are connecting your server to a host computer, set this field to **network**.
user	Enter **user** if your server releases the contested circuit and looks for another when glare occurs. If you are connecting your server to a public network, set this field to **user**.

1 of 2

Valid entries	Usage
Use the following values for private networks (including QSIG networks) outside the U.S. Entering either of these values causes the **Peer Protocol and Side** fields to appear.	
peer-master	Enter **peer-master** if your switch overrides the other end when glare occurs.
peer-slave	Enter **peer-slave** if your switch releases the contested circuit and looks for another when glare occurs.
	2 of 2

Interface Companding

Appears when the **Group Type** field is **atm**. Indicates the companding algorithm expected by the system.The entry in this field must match the companding method used by the far-end switch.

Valid entries	Usage
alaw	Enter **alaw** for E-1 service.
mulaw	Enter **mulaw** for T-1 service.

Interworking Message

Appears when the **Group Type** field is **atm**, **h.323**, or **sip**. This field determines what message Communication Manager sends when an incoming ISDN trunk call interworks (is routed over a non-ISDN trunk group).

Valid entries	Usage
PROGress	Normally select this value. PROGress asks the public network to cut through the B-channel and let the caller hear tones such as ringback or busy tone provided over the non-ISDN trunk.
ALERTing	ALERTing causes the public network in many countries to play ringback tone to the caller. Select this value only if the DS1 is connected to the public network, and it is determined that callers hear silence (rather than ringback or busy tone) when a call incoming over the DS1 interworks to a non-ISDN trunk.

IP Audio Hairpinning

Appears when the **Group Type** field is **h.323** or **sip**. Allows H.323 endpoints to be connected through the IP circuit pack in the media server or switch, without going through the Avaya DEFINITY TDM bus. Initially not available for SIP.

Valid entries	Usage
y/n	Enter **y** to allow IP endpoints to be connected through the IP circuit pack in the media server or switch in IP format, without going through the Avaya DEFINITY TDM bus. Initially, support for SIP Trunks requires the default of **n**.

LRQ Required

Appears when the **Group Type** field is **h.323**.

Valid entries	Usage
y/n	Enter **n** if the far-end server is running Avaya Communication Manager. Enter **y** if the far-end server is not an Avaya DEFINITY or media server and requires a location request to obtain a signaling address in its signaling protocol. If **y**, "Calls Share IP Signaling Connection" must be **n**.

Max number of CA TSC

Appears when the **Group Type** field is **atm**, **h.323**, or **isdn-pri**.

Valid entries	Usage
0 to **619**	Maximum number of simultaneous call-associated Temporary Signaling Connections that can exist in the signaling group. Typically this is the number of ISDN-PRI trunk group members controlled by this signaling group.
0 to **400**	For DEFINITY Server SI only.

Max number of NCA TSC

Appears when the **Group Type** field is **atm**, **h.323**, or **isdn-pri**.

Valid entries	Usage
0 to **256**	Maximum number of simultaneous non-call-associated Temporary Signaling Connections. The TSCs carry signaling for features not associated with a specific call, for example, signals to turn on Leave Word Calling.

Media Encryption

Appears when the Media Encryption feature is enabled in Communication Manager and the **Group Type** field is **h.323**.

Valid entries	Usage
y/n	Enter **y** to enable encryption for trunk calls assigned to this signaling group. If encryption for the signaling group is not enabled, then trunk calls using this signaling group will not be encrypted regardless of IP Codec Set administration.

Name

Appears when the **Group Type** field is **atm**.

Valid entries	Usage
up to 15 alphnumeric	Enter 15 alphanumeric characters for identification.

Near-end Listen Port

Appears when **Group Type** is **h.323** or **sip**. Defaults to **5061** for SIP over TLS.

Valid entries	Usage
1719, 1720, or 5000 to **9999**	Enter an unused port number. The default for SIP is 5061. Avaya recommends **1720** for h.323 and **1719** if LRQ is **y**.

Near-end Node Name

Appears when the value of the entry in the **Group Type** field is **atm** or **sip**. Enter the node name for the C-LAN IP interface in this media server. The node name must be administered on the **IP Node Names** screen and the **IP Interfaces** screen.

Valid entries	Usage
Name of an administered IP node	Describe the near-end node.

Network Call Transfer

Appears when the **Group Type** field is **atm**. For more information, see *Avaya MultiVantage Call Center Software Release 11 — Network Call Redirection*.

Valid entries	Usage
y/n	Enter **y** to indicate D-channels are supporting ENCT.

Passphrase

Appears when Media Encryption is enabled and the **Group Type** field is **h.323**.

Valid entries	Usage
8 to 30 alphanumeric characters	Enter a value for the passphrase used to generate a shared "secret" for symmetric encryption of the media session key. The same passphrase must be assigned to the corresponding signaling groups at both ends of an IP trunk. The passphrase: • Is case sensitive • Must contain at least 1 alphabetic and at least 1 numeric • Valid characters also include letters, numbers, and these symbols: **!&*?;'^(),.:-**

Primary D Channel

Appears when the **Group Type** field is **isdn-pri**.

Valid entries	Usage
Cabinet number (**1** to **44**), Carrier (**A** to **E**), Slot (**00** to **20**), Circuit (**09** to **32**)	Enter the letter or number of the Primary D Channel..

Protocol Version

Appears when the **Group Type** field is **atm**.

Valid entries	Usage
a **b** **c** **d**	

RRQ Required

Appears when the **Group Type** field is **h.323**. This field specifies the signaling group that serves as a gateway rather than gatekeeper.

Valid display	Usage
y/n	Displays **y** if the signaling group serves a remote office (gateway). Displays **n** if the signaling group serves a gatekeeper.

Remote Office

Appears when the **Group Type** field is **h.323**.

Valid entries	Usage
y/n	Enter **y** if the signaling group serves a remote office.

SBS

Appears when the **Group Type** field is set to **h.323**. If you set this to **y**, you must set both the **Trunk Group for NCA TSC** field and the **Trunk Group for Channel Selection** field equal to the signaling group number administered for the SBS trunk group. The **Supplementary Service Protocol** field should always be set to **b** to obtain full QSI.

Valid entries	Usage
y/n	Enter **y** to use SBS signaling trunk groups. The default is **n**.

Signaling Mode

A display-only field that appears when the **Group Type** field is **atm**. This field always sets to **isdn-pri**.

Supplementary Service Protocol

Appears when the **Group Type** field is **atm**, **h.323**, or **isdn-pri**.

Valid entries	Usage
a	AT&T, Bellcore, Nortel. When the **Country Code** field on the **DS1** screen is **1A**, SSA selects AT&T custom supplementary services. When the **Country Code** field on the **DS1** screen is **1B**, SSA selects Bellcore Supplementary Services. When the **Country Code** field on the **DS1** screen is **1C**, SSA selects Nortel Proprietary Supplementary Services.
b	ISO QSIG. Also, use this entry for SBS signaling groups.
c	ETSI
d	ECMA QSIG
e	Allows DCS with rerouting. **DCS with Rerouting** must be **y**, and the **Used for DCS** field on the **Trunk Group** screen must be **y**.
f	Feature Plus
g	ANSI. Available only if, on the **System-Parameters Customer-Options** screen, the **ISDN-PRI** or **ISDN-BRI** field is **y** or the **Used for DCS** field is **y**.

T303 Timer (sec)

Use this field to enter the number of seconds the system waits for a response from the far end before invoking Look Ahead Routing. Appears when the **Group Type** field is **h.323**.

Valid entries	Usage
2 to **10**	Enter a number of seconds between 2 and 10. Default is 10.

Transport Method

Appears when the **Group Type** field is **sip**.

Valid entries	Usage
tls	Initially, SIP support requires the default method of **TLS**. In Avaya Communication Manager 2.0, this field is read-only.

Trunk Group for Channel Selection

Appears when the **Group Type** field is **atm**, **h.323**, or **isdn-pri**.

Valid entries	Usage
1 to **666** or blank	Enter the trunk group number. If there is more than one trunk group assigned to this signaling group, the group entered in this field will be the one that can accept incoming calls. For DEFINITY R.
1 to **99** or blank	For DEFINITY SI only.
1 to **2000**	For S8300 Media Server, S8700 or S8710 IP-Connect, S8700 or S8710 Multi-Connect.

Trunk Group for NCA TSC

Appears when the **Group Type** field is **atm**, **h.323**, or **isdn-pri**.

Valid entries	Usage
1 to **666** or blank	Enter the ISDN-PRI trunk group number whose incoming call-handling table will be used to handle incoming NCA-TSCs through this signaling group. For Avaya DEFINITY Server R.
1 to **2000** or blank	For Avaya S8300 Media Server, S8700 or S8710 IP-Connect, and S8700 or S8710 Multi-Connect.
1 to **99** or blank	For DEFINITY Server SI only.

Virtual Channel Identifier

Appears when the **Group Type** field is **atm**.

Valid entries	Usage
32 to **1023** or blank	Enter a number between 32 and 1023 or blank.

Virtual Path Identifier

A display-only field that appears when the **Group Type** field is **atm**. This field always sets to **0**.

Valid entries	Usage
32 to **1023** or blank	Enter a number between 32 and 1023 or blank.

X-Mobility/Wireless Type

Appears when the **Group Type** field is **isdn-pri**. This field indicates the type of X-Mobile endpoints allowed.

Valid entries	Usage
DECT	Indicates to Communication Manager that the remote end of the trunk group controlled by the signaling group is a DECT mobility controller. This allows X-Mobility to work over ISDN-PRI trunks between the server/switch and adjunct.
none	

Signaling Group screen (page 2)

This screen appears only when the **Group Type** is **atm**.

Figure 499: Signaling Group screen (when the Group Type field is atm)

```
                                                          Page 2 of s

SIGNALING GROUP

Chan Port Chan Port
  1:       17:
  2:       18:
  3:       19:
  4:       20:
  5:       21:
  6:       22:
  7:       23:
  8:       24:
  9:
 10:
 11:
 12:
 13:
 14:
 15:
 16:
```

Chan Port

Displays when the **Group Type** field is **atm**. If the **Circuit Type** field on page 1 is **T1**, this field displays 24 channels; if you specified **E1**, it displays 31 channels.

You must fill this screen in for ATM signaling groups. This provides two things:

- It allows you to define fractional T1 and fractional E1 facilities, specifying how many and which channels to use.
- It allows you to choose the port numbers to use (port numbers must be unique for all signaling groups on the same ATM board).

The signaling channel (port 16 for an E1 and port 24 for a T1) must be a port between **9** and **32**. Any port number used on this screen may not be used on any other ATM signaling group on the same board.

The channels used must match exactly the channels used on the other end of the signaling group. For example, if your T1 is set up to use channels 1 through 5, 7, and 24 (the signaling channel), the far end must use channels 1 through 5, 7, and 24.

Valid entries	Usage
009 to **256** or blank	Enter the port number for non-signaling channels.

Signaling Group Administered NCA TSC Assignments page

Figure 500: Signaling Group screen (Administered NCA-TSC Assignment Page)

```
                                                                Page 2 of x
                        ADMINISTERED NCA TSC ASSIGNMENT
Service/Feature: ___________    As-needed Inactivity Time-out(min):_
  TSC    Local                                                Adj.      Mach.
 Index    Ext.   Enabled  Established   Dest. Digits    Appl.    Name       ID
  1:     _____      __     _________    _____________   _______  _______    __
  2:     _____      __     _________    _____________   _______  _______    __
  3:     _____      __     _________    _____________   _______  _______    __
  4:     _____      __     _________    _____________   _______  _______    __
  5:     _____      __     _________    _____________   _______  _______    __
  6:     _____      __     _________    _____________   _______  _______    __
  7:     _____      __     _________    _____________   _______  _______    __
  8:     _____      __     _________    _____________   _______  _______    __
  9:     _____      __     _________    _____________   _______  _______    __
 10:     _____      __     _________    _____________   _______  _______    __
 11:     _____      __     _________    _____________   _______  _______    __
 12:     _____      __     _________    _____________   _______  _______    __
 13:     _____      __     _________    _____________   _______  _______    __
 14:     _____      __     _________    _____________   _______  _______    __
 15:     _____      __     _________    _____________   _______  _______    __
```

Field descriptions for page 2

Appl.

Specifies the application for this administered NCA-TSC.

Valid entries	Usage
audix	Use this for ISDN-PRI D-channel DCS Audix feature.
dcs	Use this for the DCS Over ISDN-PRI D-channel feature.
gateway	Use this when the administered NCA-TSC is used as one end in the gateway channel connecting to a BX.25 link. If gateway is entered, then the **ISDN TSC Gateway Channel Assignments** screen must be completed.
masi	Use this when the NCA-TSC is one end of a multimedia application server interface.
qsig-mwi	Use this to convert messages from an administered AUDIX NCA-TSC to a QSIG CISC. If you use this application type, then you must enter a Machine ID between 1 and 20.

As-needed Inactivity Time-out (min)

Valid entries	Usage
10 to **90** or blank	This field determines that if you administer any of the TSCs as "as-needed (as opposed to "permanent"), then the TSC is dropped if no message is sent over the TSC in the designated number of minutes.

Dest. Digits

Valid entries	Usage
Up to 15 characters **0** to **9**, *****, #	Enter the extension of the ISDN interface.

Enabled

Valid entries	Usage
y/n	Enter **y** to enable the administered NCA-TSC.

Established

Used to indicate the strategy for establishing this administered NCA-TSC.

Valid entries	Usage
permanent	Use permanent so that the administered NCA-TSC can be established by either the near end or the far end.
as-needed	Use as-needed so that the administered NCA-TSC will be established the first time the administered NCA-TSC is needed; it can be set up either by the near end or far end switch.

Local Ext

Valid entries	Usage
Extension	Enter the extension of the ISDN interface.

Mach ID

You can enter up to 20 machine IDs.

Valid entries	Usage
1 to **20**	Enter a unique machine ID. The system does not allow you to specify an ID that you already entered on the **Processor Channel** screen.
1 to **63** for DCS **1** to **99** for Audix **1** to **15** for MASI **1** to **20** for QSIG-MWI blank	For S8700 or S8710 IP-Connect.

Service/Feature

These are the encodings that are used in the SETUP messages for the Temporary Signaling Connections administered on pages 2 to 5 of this screen.

Valid entries
accunet
inwats
lds
mega800
megacom
multiquest
operator
sdn
sub-operator
wats-maxbnd

Site Data

Use this screen to enter information about buildings, floors and telephone set colors. You must supply values on this screen before you can enter information in the **Site Data** section of the **Station** screen.

Figure 501: Site Data screen

```
change site-data
SITE DATA USER DEFINITION
VALID BUILDING FIELDS


   _________   _________   _________   _________   _________
   _________   _________   _________   _________   _________
   _________   _________   _________   _________   _________
   _________   _________   _________   _________   _________
   _________   _________   _________   _________   _________
```

Field descriptions for pages 1, 2 and 3

These pages are available for you to enter descriptive information about the buildings, floors and telephone set colors. You may enter any valid keyboard character. If you want to indicate that a particular floor is in a particular building, you must include this in the floor entry, for example, B301-Fl4.

Station

This section provides descriptions of all of the fields that may appear on **Station** screens. Some of the fields are used for specific telephone types; others are used for all telephone types. To make it easier to find a specific field, they are listed in alphabetical order by field name.

Figure 502: Station screen

```
change station 1014                                          Page 1 of X
                              STATION

Extension: 1014               Lock Messages? n                      BCC: 0
     Type:                    Security Code:                         TN: 1
     Port:                  Coverage Path 1:                        COR: 1
     Name:                  Coverage Path 2:                        COS: 1
                            Hunt-to Station:

STATION OPTIONS
              Loss Group: 2               Personalized Ringing Pattern: 3
             Data Module? n                          Message Lamp Ext: 1014
           Speakerphone: 2-way                    Mute button enabled? y
       Display Language? English                     Expansion Module?
                  Model:

 Survivable GK Node Name:                            Media Complex Ext:
                                                          IP Softphone? y
                                                   Remote Office Phone? y
```

Figure 503: Station screen when Type field is XMOBILE (page 1)

```
change station 1014                                        Page 1 of x
                                    STATION

Extension: 30016                Lock Messages? n                      BCC: 0
     Type: XMOBILE               Security Code:                        TN: 1
                               Coverage Path 1:                        COR: 1
     Name:                     Coverage Path 2:                        COS: 1
                                    Hunt-to Station:

STATION OPTIONS

       XMOBILE Type: EC500                          Message Lamp Ext: 30016
     Display Module? n                           Message Waiting Type: NONE

  Mobility Trunk Group:                                    Calls Allowed: all
     Configuration Set: 1

  CELL PHONE NUMBER MAPPING
          Dial Prefix:
    Cell Phone Number:
         Mapping Mode: termination
```

Figure 504: Station screen (page 2)

```
change station 75001                                              Page 2 of X
                                    STATION

FEATURE OPTIONS
           LWC Reception? spe                Auto Select Any Idle Appearance? n
          LWC Activation? y                            Coverage Msg Retrieval? y
  LWC Log External Calls? n                                   Auto Answer: none
             CDR Privacy? n                                 Data Restriction? n
   Redirect Notification? y                      Idle Appearance Preference? n
 Per Button Ring Control? n
   Bridged Call Alerting? n                        Restrict Last Appearance? y
  Active Station Ringing: single

        H.320 Conversion? n             Per Station CPN - Send Calling Number? _
       Service Link Mode: as-needed          Busy Auto Callback without Flash? y
         Multimedia Mode: basic
    MWI Served User Type: ______                    Display Client Redirection? n
         Automatic Moves:
              AUDIX Name:                         Select Last Used Appearance? n
                          _                         Coverage After Forwarding? _
     Recall Rotary Digit? n                           Multimedia Early Answer? n

Remote Softphone Emergency Calls: as-on-local   Direct IP-IP Audio Connections? n
Emergency Location Ext: 75001      Always use? n          IP Audio Hairpinning? n
Precedence Call Waiting? y
```

Figure 505: Station screen (page 3)

```
add station 1014                                          Page 3 of X
                                   STATION

    SITE DATA
         Room: _______                                  Headset? n
         Jack: ___                                      Speaker? n
        Cable: ___                                     Mounting: d
        Floor: _______                            Cord Length: 0_
     Building: _______                              Set Color: _____


ABBREVIATED DIALING
     List1: ________          List2: _________          List3: _________

BUTTON ASSIGNMENTS
 1: call-appr                5:
 2: call-appr                6:
 3: call-appr                7:
 4: audix-rec  Ext: 4000     8:
```

Figure 506: Station screen (page 4)

```
change station 1014                                       Page 4 of x
                                   STATION

FEATURE BUTTON ASSIGNMENTS

 1:                                    13:
 2:                                    14:
 3:                                    15:
 4:                                    16:
 5:                                    17:
 6:                                    18:
 7:                                    19:
 8:                                    20:
 9:                                    21:
10:                                    22:
11:                                    23:
12:                                    24:
```

If the **Expansion Module** field is **y**, a fifth page appears.

Figure 507: Station screen (page 5)

```
change station 1014                                          Page 5 of x
                                        STATION
EXPANSION MODULE BUTTON ASSIGNMENTS

1:                                      13:
2:                                      14:
3:                                      15:
4:                                      16:
5:                                      17:
6:                                      18:
7:                                      19:
8:                                      20:
9:                                      21:
10:                                     22:
11:                                     23:
12:                                     24:
```

The standard **Station** screen field descriptions are listed alphabetically for easy reference.

1-Step Clearing

Valid entries	Usage
y/n	If set to **y**, the call terminates again at the WCBRI terminal when the user drops from the call.

Abbreviated Dialing List1, List2, List3

You can assign up to 3 abbreviated dialing lists to each telephone.

Valid entries	Usage
enhanced	Allows the telephone user to access the enhanced system abbreviated dialing list.
group	Allows the telephone user to access the specified group abbreviated dialing list. If you enter **group**, you also must enter a group number.
personal	Allows the telephone user to access and program their personal abbreviated dialing list. If you enter **personal**, you also must enter a personal list number.
system	Allows the telephone user to access the system abbreviated dialing list.

Access Code

This field appears when a wireless terminal model number is selected in the **Type** field. The Access Code is a temporary, shorter version of the complete User Authentication Key (UAK) required by the system when the terminal is first put into service. It is then used to automatically generate a unique UAK for that wireless terminal over-the-air.

Valid entries	Usage
any 5-digit decimal number	Default is blank. Enter the 5-digit access code to place the wireless terminal into service.

Active Station Ringing

Defines how call rings to the telephone when it is off-hook. This field does not affect how calls ring at this telephone when the telephone is on-hook.

Valid entries	Usage
continuous	Enter **continuous** to cause all calls to this telephone to ring continuously.
single	Enter **single** to cause calls to this telephone to receive one ring cycle and then ring silently.
if-busy-single	Enter **if-busy-single** to cause calls to this telephone to ring continuously when the telephone is off-hook and idle and calls to this telephone to receive one ring cycle and then ring silently when the telephone is off-hook and active.
silent	Enter **silent** to cause all calls to this station to just ring silently.

Adjunct Supervision

Adjunct Supervision appears when the **Type** field is **500**, **2500**, **k2500**, **8110**, **ops**, **ds1fd**, **ds1sa**, **VRU**, **VRUFD**, or **VRUSA**.

Valid entries	Usage
y	Enter **y** if an analog disconnect signal is sent automatically to the port after a call terminates. Analog devices (such as answering machines and speakerphones) use this signal to turn the devices off after a call terminates.
n	Set this field to **n** so hunt group agents are alerted to incoming calls. In a hunt group environment, the disconnect signal blocks the reception of zip tone and incoming call notification by an auto-answer station when a call is queued for the station.

Always Use

This field does not apply to SCCAN wireless telephones, or to extensions administered as type **h.323**.

Valid entries	Usage
y	When this field is **y**: • The **Remote Softphone Emegency Calls** field is hidden. A softphone can register no matter what emergency call handling settings the user has entered into the softphone. If a softphone dials 911, the **Emergency Location Extension** administered on the **Station** screen is used. The softphone's user-entered settings are ignored. • If an IP telephone dials 911, the **Emergency Location Extension** administered on the **Station** screen is used.
n	Default value. For more information, see the description for the **Emergency Location Extension** field on the **Station** screen.

Assigned Member — Ext

The system automatically assigns this extension. This is the extension of the user who has an associated **Data Extension** button and shares the module.

Assigned Member — Name

Display-only field that shows the name associated with the extension shown in the **Assigned Member - Ext** field.

Att. Call Waiting Indication

Attendant call waiting allows attendant-originated or attendant-extended calls to a busy single-line telephone to wait and sends distinctive call-waiting tone to the single-line user. Must be set to **y** when the **Type** field is set to **H.323**.

Valid entries	Usage
y/n	Enter **y** to assign Attendant Call Waiting to the telephone. You should not set this field to **y** if the **Data Restriction** field is **y** or the **Switchhook Flash** field is **n**, or if Data Privacy is enabled for the telephone's class of service (COS). If any of these conditions are true, the telephone cannot accept or handle call waiting calls.

Audible Message Waiting

The Audible Message Waiting tone indicates that the user has a waiting message. This field appears only if **Audible Message Waiting** is set to **y** on the **System-Parameters Customer-Options** screen.

Note that this field does not control the Message Waiting lamp.

Valid entries	Usage
y/n	Enter **y** if you want the telephone user to receive stutter dial tone when they have a waiting message and they go off-hook.

Audix Name

Specifies which AUDIX is associated with the station.

Valid entries	Usage
Names assigned to an AUDIX adjunct	Must contain a user-defined adjunct name that was previously administered in the **Node-Names** screen.

Auto Answer

In EAS environments, the auto answer setting on the **Agent LoginID** screen can override a station's setting when an agent logs in there.

Note:

For analog stations, if Auto Answer is set to acd and the station is off-hook and idle, only the ACD split/skill calls and direct agent calls auto answer; non-ACD calls receive busy treatment. If the station is active on an ACD call and a non-ACD call arrives, the Agent receives call-waiting tone.

Valid entries	Usage
all	Enter all to allow all calls (ACD and non-ACD) terminated to an idle station to be cut through immediately. Does not allow automatic hands-free answer for intercom calls.
acd	Enter acd to allow only ACD split /skill calls and direct agent calls to auto answer. If this field is set to acd, Non-ACD calls terminated to a station ring audibly.
none	Enter **none** to cause all calls terminated to this station to receive an audible ringing treatment.
icom	Enter icom to allow a telephone user to answer an intercom call from the same intercom group without pressing the **intercom** button.

Automatic Moves

Automatic Moves allows a DCP telephone to be unplugged from one location and moved to a new location without additional Communication Manager administration. Communication Manager automatically associates the extension to the new port.

CAUTION:

When a DCP telephone is unplugged and moved to another physical location, the **Emergency Location Extension** field must be changed for that extension or the USA Automatic Location Identification data base must be manually updated. If the **Emergency Location Extension** field is not changed or if the USA Automatic Location Identification data base is not updated, the DID number sent to the Public Safety Network could send emergency response personnel to the wrong location.

Valid entries	Usage
always	Enter **always** and the DCP telephone can be moved anytime without additional administration by unplugging from one location and plugging into a new location.
once	Enter **once** and the DCP telephone can be unplugged and plugged into a new location once. After a move, the field is set to done the next time that routine maintenance runs on the DCP telephone. Use **once** when moving a large number of DCP phones so each extension is removed from the move list. Use **once** to prevent automatic maintenance replacement.
no	Enter **no** to require administration in order to move the DCP telephone.
done	Done is a display-only value. Communication Manager sets the field to done after the telephone is moved and routine maintenance runs on the DCP telephone.
error	Error is a display-only value. Communication Manager sets the field to error, after routine maintenance runs on the DCP telephone, when a non-serialized telephone is set as a movable telephone.

Auto Select Any Idle Appearance

Valid entries	Usage
y/n	Enter **y** to allow automatic selection of any idle appearance for transferred or conferenced calls. Communication Manager first attempts to find an idle appearance of the call being transferred or conferenced. If that attempt fails, Communication Manager selects the first idle appearance.

Automatic Selection of DID Numbers

Communication Manager chooses a 2- to 5-digit extension from a predetermined list of numbers and assigns the extension to a hotel room telephone.

Valid entries	Usage
y/n	Enter **y** to use the Automatic Selection of DID Numbers for Guest Rooms feature.

BCC

Appears when **ISDN-PRI** or **ISDN-BRI Trunks** is enabled on the **System-Parameters Customer-Options** screen. Display-only field set to **0** for stations (that is, indicates voice or voice-grade data).

See "Generalized Route Selection" in *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205, for more information on Bearer Capability Classes (BCC) and their ability to provide specialized routing for various types of voice and data calls. The BCC value is used to determine compatibility when non-ISDN facilities are connected to ISDN facilities (ISDN Interworking).

Bridged Call Alerting

If **Bridged Call Alerting** is **n** and **Per Button Ring Control** is **n**, audible ringing is suppressed for incoming calls on bridged appearances of another telephone's primary extension.

Valid entries	Usage
y/n	Enter **y** to enable audible ringing for TEG, PCOL, bridged appearances, or Data Extension calls.

Building

Enter a valid building location. See Site Data on page 1408 for valid entries.

Busy Auto Callback without Flash

Appears on the **Station** screen for analog telephones, only if the **Without Flash** field in the **ANALOG BUSY AUTO CALLBACK** section of the **Feature-Related System Parameters** screen is set to **y**. The **Busy Auto Callback without Flash** field then defaults to **y** for all analog phones that allow Analog Automatic Callback.

Valid entries	Usage
y/n	Enter **y** to provide automatic callback for a calling analog station without flashing the hook.

Button Assignments

Enter the abbreviated software name to assign a feature button. For a list of feature buttons, see Telephone feature Buttons on page 123.

Note:

If you want to use Terminal Translation Initialization (TTI), you must assign a call appearance (**call-appr**) to the first button position. TTI needs the button on the first call appearance to get dial tone.

Cable

You can use this field to identify the cable that connects the telephone jack to the system. You also can enter this information in the Blank column on the Port Assignment Record.

Caller ID Message Waiting Indication

Appears when the **Type** field is **CallrID**. For CallrID type phones or analog phones with Caller ID adjuncts only.

Valid entries	Usage
y/n	Enter **y** to allow aliasing of various non-Avaya phones and adjuncts.

Note:

The Caller ID Message Waiting Indication administration is independent of the administration of LED or NEON-lamp Avaya Communication Manager Message Waiting Indication (MWI). For example, it is possible to administer a Caller ID telephone with the **Caller ID Message Waiting Indication** field set to **n** and the **Message Waiting Indicator** field set to **neon**.

Calls Allowed

Appears if the **XMOBILE Type** field is **EC500** and the **Mapping Mode** field is **termination** or **both**. Used to identify the EC500 call filter type for an XMOBILE station. This field allows an XMOBILE station to function as a bridge and still be restricted.

Valid entries	Usage
internal	External calls are blocked. Internal calls terminate to the XMOBILE station. Attendant-originated and attendant-delivered calls are not delivered
external	Internal calls are blocked. External calls terminate to the XMOBILE station.
all	All calls terminate to the XMOBILE station.
none	Prevents calls from terminating to the XMOBILE station. Can be used to prevent business-related calls from accruing telephone charges on cell phones that are lost, being transferred to a new user, or being disabled for other business reasons.

Note:

Interswitch calls on DCS trunks are treated as internal calls. When the **Calls Allowed** field is set to **internal** or **all**, DCS calls are delivered to the cell telephone. When the **Calls Allowed** field is set to **external** or **none**, DCS calls are not delivered.

Note:

Incoming calls from other EC500 users are internal if office caller ID is enabled for the XMOBILE station associated with the cell telephone. When the **Calls Allowed** field is set to **internal** or **all**, calls from other EC500 users are delivered. When the **Calls Allowed** field is set to **external** or **none**, calls from other EC500 users are not delivered.

Note:

The calling party receives busy treatment when call filtering blocks calls to a standalone XMOBILE. Calls delivered to standalone XMOBILE stations that are not answered will follow the call coverage or call forwarding paths administered for the standalone XMOBILE.

Call Waiting Indication

This allows user, attendant-originated, and outside calls to busy single-line telephone to wait and sends a distinctive call-waiting tone to the single-line user. This feature is denied if **Data Restriction** is **y** or **Switchhook Flash** is **n**, or if Data Privacy is active by way of the telephone COS assignment. Must be set to **y** when the **Type** field is set to **H.323**.

Valid entries	Usage
y/n	Enter **y** to activate Call Waiting Termination for the telephone.

CDR Privacy

This option allows digits in the called number field of an outgoing call record to be blanked, on a per-station basis. You administer the number of blocked digits system-wide in the **Privacy - Digits to Hide** field on the **CDR System Parameters** screen.

Valid entries	Usage
y/n	Enter **y** to enable Call Privacy for each station.

Cell Phone Number

Contains the unformatted cell telephone's published external number. This field may contain a 3-digit area code plus the 7-digit main number. If the same Cell Phone Number is administered on multiple XMOBILE station screens, then the Dial Prefix associated with each instance of the Cell Phone Number must be the same.

Valid entries	Usage
0 to **9**	Enter 1 to15 digits. Avaya recommends that you enter a full 10-digit Cell Phone Number regardless of whether the cell telephone is local or not. Note that your **ARS** screen has to be administered to handle this.

Configuration Set

This field is used to administer the Configuration Set number that contains the call treatment options desired for the XMOBILE station. This field must be administered if:

- The **XMOBILE Type** field is **EC500**.
- The **Mobility Trunk Group** field is a trunk group number and the administered trunk group is non-DECT or non-PHS.
- The **Mobility Trunk Group** field is **aar** or **ars**.

If the **Mobility Trunk Group** field is a trunk group number and the administered trunk group is **DECT** or **PHS**, this field may be left blank.

Valid entries	Usage
1 to **10** or blank	Enter any value corresponding to the appropriate **Configuration Set** screen.

COR

Enter a Class of Restriction (COR) number to select the desired restriction.

Cord Length

Enter a number to specify the length of the cord attached to the receiver. This is a free-form entry, and can be in any measurement units.

COS

Enter the desired Class of Service (COS) number to select allowed features.

Country Protocol

Enter the protocol that corresponds to your supported initialization and codesets. The Country Protocol must match any previously-administered endpoint on the same port.

Valid entries	Usage
1	United States (Bellcore National ISDN)
2	Australia
etsi	ETSI (Europe)
3	Japan
6	Singapore

Coverage After Forwarding

This field governs whether an unanswered forwarded call is provided coverage treatment.

Valid entries	Usage
y	Coverage treatment is provided after forwarding regardless of the value of the **Coverage After Forwarding** field on the **System Parameters - Call Coverage/Call Forwarding** screen.
n	No coverage treatment is provided after forwarding regardless of the value of the **Coverage After Forwarding** field on the ***System Parameters - Call Coverage/Call Forwarding*** screen.
s(ystem)	Indicates that call processing uses the **Coverage After Forwarding** field on the **System Parameters Call Coverage/Call Forwarding** screen. To override the system-wide parameter for a given station, set this field to **y** or **n**.

Coverage Msg Retrieval

Applies if the telephone is enabled for LWC Reception.

Valid entries	Usage
y/n	Enter **y** to allow users in the telephone's Coverage Path to retrieve Leave Word Calling (LWC) messages for this telephone.

Coverage Module

Valid entries	Usage
y	Enter **y** to indicate that a coverage module is connected to the station. Once you enter **y**, the system displays an additional page that allows you to assign the buttons for the module.

Coverage Path 1 or Coverage Path 2

Enter a coverage-path number or time-of-day table number from a previously-administered **Call Coverage Path** screen or **Time of Day Coverage Table** screen.

Note:

If Modified Misoperation is active (**Misoperation Alerting** is **y** on the **Feature-Related System Parameters** screen), you must assign a Coverage Path to all stations on Communication Manager.

CRV Length

Only for ASAI stations. Enter **1** or **2** to indicate the length of CRV for each interface.

Custom Selection of VIP DID Numbers

Custom Selection of VIP DID numbers allows you to select the DID number assigned to a room when a guest checks in.

Valid entries	Usage
y/n	Enter **y** to allow you to select the DID number assigned to a room when a guest checks in.

Data Extension

Enter the extension assigned to the data module.

Data Option

Valid entries	Usage
analog **data module** **none**	If a second line on the telephone is administered on the I-2 channel, enter **analog**. Otherwise, enter **data module** if applicable or **none**. See the *Terminals and Adjuncts Reference* for more information.

Data Restriction

Data restriction provides permanent protection and cannot be changed by the telephone user. Do not assign a Data Restriction if **Auto Answer** is **all** or **acd**. If **y**, whisper page to this station is denied.

Valid entries	Usage
y/n	Enter **y** to prevent tones, such as call-waiting tones, from interrupting data calls.

Default Dialing Abbreviated Dialing Dial Code

Appears only when the **Special Dialing Option** is set to default. Enter a list number associated with the abbreviated dialing list.

When the user goes off-hook for a data call and presses the **Enter** button following the DIAL prompt, the system dials the AD number. The data call originator also can perform data-terminal dialing by specifying a dial string that may or may not contain alphanumeric names.

Dial Prefix

Contains the unformatted sequence of digits or characters that are prepended to the cell telephone's published cell telephone number before dialing. If the same Cell Phone Number is administered on multiple XMOBILE station screens, then the Dial Prefix associated with each instance of the Cell Phone Number must be the same.

Valid entries	Usage
up to 4 digits: **0** to **9**, *****, #	Enter 1 to 4 digits.

Direct IP-IP Audio Connections

Allows direct audio connections between IP endpoints.

Valid entries	Usage
y/n	Enter **y** to save on bandwidth resources and improve sound quality of voice over IP transmissions.

Display Caller ID

Appears when the **Type** field is **CallrID**. For CallrID type phones or analog phones with Caller ID adjuncts only.

Valid entries	Usage
y/n	Enter **y** to allow transmission of calling party information to the Caller ID telephone or adjunct.

Display Cartridge

For 7404 D phones only. Enter **y** to indicate there is a display cartridge associated with the station. This displays an additional page to allow you to assign display buttons for the display cartridge.

Display Client Redirection

Only administrable if **Hospitality** is enabled on the **System-Parameters Customer-Options** screen. This field affects the telephone display on calls that originated from a station with Client Room Class of Service.

Note:

For stations with an audix station type, AUDIX Voice Power ports, or ports for any other type of messaging that needs display information, **Display Client Redirection** must be set to **y**.

Valid entries	Usage
y	When set to **y**, the redirection information for a call originating from a Client Room and terminating to this station displays.
n	When set to **n**, this station's display does not show the redirection information for all calls originating from a Client Room (even redirected calls) that terminate to this station. Only the client name and extension (or room, depending on what is administered on the **Hospitality** screen) display.

Display Language

Specifies the display language.

Valid entries	Usage
english	Enter the language you want users to see on their displays.
french	
italian	
spanish	

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Valid entries	Usage
user-defined	
unicode	**Note:** Unicode display is only available for Unicode-supported telephones. Currently, only the 4620SW and 4610SW support Unicode display. For more information on Unicode language displays, see Administering Unicode display on page 181.
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Display Module

Valid entries	Usage
y/n	Enter **y** if this telephone has a display module.

Distinctive Audible Alert

Valid entries	Usage
y/n	Enter **y** so the telephone can receive the 3 different types of ringing patterns which identify the type of incoming calls. Distinctive ringing may not work properly for off-premises telephones.

Emergency Location Ext

The **Emergency Location Ext** field defaults to the telephone's extension. This extension identifies the street address or nearby location when an emergency call is made. For more information about the use of this field, see the **Usage** description for the **Remote Softphone Emergency Calls** field later in this section.

Valid entries	Usage
1 to 8 digits	Enter the Emergency Location Extension for this station.

Note:

On the **ARS Digit Analysis Table** screen, you must administer 911 to be call type **emer** or **alrt** in order for the E911 Emergency feature to work properly.

Endpt ID

Appears only if **Endpt Init** is **y**. Enter a unique 2-digit number (**00** to **62**) for this endpoint. Each **Endpt ID** field must have a unique value for each endpoint on the same port.

This field provides for multipoint configuration conformance to the Bellcore Terminal Initialization procedures. In these procedures, a multipoint configuration requires the last 2 digits of the Service Profile Identifier (SPID) be between 00 and 63 and be binary unique for each endpoint.

For WorldClass BRI (WCBRI) data modules only, this field, combined with the SPID, gives the effective SPID administered into the terminal. Bellcore ISDN-1 requires the SPID programmed into the endpoint contain at least 9 digits. For example, if the SPID is **1234**, and **Endpt ID** is **01**, then the SPID administered on the terminal is 000123401. The three leading zeros are necessary to create a 9-digit SPID.

Endpt Init

Endpoint initialization is a procedure, required for multipoint operation, by which User Service Order Profile (USOP) is associated with an endpoint on the ISDN-BRI. This association is made via the SPID, administered into the system, and entered into the ISDN-BRI terminal. For an ISDN-BRI terminal to be operational in a multipoint configuration, both the administered SPID and the SPID programmed into the ISDN-BRI terminal must be the same. Therefore, the SPID of new or reused terminals must be programmed to match the administered SPID value. Appears only if **MIM Support** is **y** and indicates the terminal's endpoint initialization capability.

Valid entries	Usage
y	Enter **y** if the terminal supports Bellcore ISDN-1 terminal initialization procedures.
n	Enter **n** for all other country protocols.

Expansion Module

Valid entries	Usage
y/n	Enter **y** if this telephone has an expansion module. This will enable you to administer the buttons for the expansion module.

Extension

Displays the extension for this station—this is either the extension you specified in the station command or the next available extension (if you used add station next).

For a virtual extension, enter a valid physical extension or a blank. Blank allows an incoming call to the virtual extension to be redirected to the virtual extension's "busy" or "all" coverage path.

Avaya recommends that you consider the display for emergency notification when you complete the **Name** field on the **Station** screen. Put the most important identifying information at the beginning of the field. When an emergency call is made and a crisis alert station with a 27-character display is notified, only 17 characters of the name field appear on the first display line, followed by the extension. The second line contains the last 3 characters of the **Name** field, followed by the word "EMERGENCY." Characters 18 through 24 of the **Name** field do not appear at all.

Feature Module

Enter **y** to indicate the station is connected to a feature module. Entering **y** displays an additional page to allow you to assign feature buttons to the module.

Fixed TEI

This field appears only for ISDN-BRI data modules, NI-BRI phones, WCBRI data modules, and ASAI links. It indicates that the endpoint has a fixed Terminal Endpoint Identifier (TEI).

The TEI identifies a unique access point within a service. You must administer TEIs for fixed TEI terminals. However, for terminals with the automatic TEI capability, the system dynamically assigns the TEI.

Valid entries	Usage
y/n	Entering **y** displays the **TEI** field. For **ASAI**, enter **y**.

Floor

Enter a valid floor location. See Translation-ID Number Mismatch Interval (days) on page 1385 for valid entries.

H.320 Conversion

Allows H.320 compliant calls made to this telephone to be converted to voice-only. Because the system can only handle a limited number of conversion calls, you may need to limit the number of telephones with H.320 conversion.

Headset

Enter **y** if the telephone has a headset.

Home

This field appears when a wireless terminal model number is selected in the **Type** field.

Valid entries	Usage
y/n	Indicate the roaming status of the wireless user. This field will specify whether the system is the user's home or roaming system. The default value for this field is **y**.

HOT LINE DESTINATION — Abbreviated Dialing Dial Code

Appears only when **Special Dialing Option** is **hot-line**.

Use Hot Line Service when very fast service is required and when you use a telephone only for accessing a certain facility. Hot Line Service allows single-line telephone users, by simply lifting the handset, to automatically place a call to a preassigned destination (extension, telephone number, or feature access code).

The Hot Line Service destination number is stored in an Abbreviated Dialing List. When the user goes off-hook on a Data Hot Line call, the system dials the AD number.

A Direct Department Calling (DDC), a Uniform Call Distribution (UCD), a Terminating Extension Group (TEG) extension, or any individual extension within a group can be a Hot Line Service destination. Also, any extension within a DDC group, UDC group, or TEG can have Hot Line Service assigned.

Loudspeaker Paging Access can be used with Hot Line Service to provide automatic access to paging equipment.

HOT LINE DESTINATION — Abbreviated Dialing List Number

Enter the abbreviated dialing list where you stored the hotline destination number.

HOT LINE DESTINATION — Dial Code

Enter the dial code in the specified abbreviated dialing list where you stored the hotline destination number.

Hunt-to Station

Enter the extension the system should hunt to for this telephone when the telephone is busy. This field allows you to create a station hunting chain (by assigning a hunt-to station to a series of phones).

Idle Appearance Preference

Indicate which call appearance is selected when the user lifts the handset and there is an incoming call.

Valid entries	Usage
y	If you enter **y**, the user connects to an idle call appearance instead of the ringing call.
n	If you enter **n**, the Alerting Appearance Preference is set and the user connects to the ringing call appearance.

Ignore Rotary Digits

If this field is **y**, the short switch-hook flash (50 to 150) from a 2500-type set is ignored.

Valid entries	Usage
y	Enter **y** to indicate that rotary digits from the set should be ignored.
n	Enter **n** to make sure they are not ignored.

IPEA

This field appears when a wireless terminal model number is selected in the **Type** field.

Valid entries	Usage
9-character hexadecimal number	International Portable Equipment Identifier. Enter the unique ID number of the wireless terminal.

IP Audio Hairpinning

Allows IP endpoints to be connected through the IP circuit pack in the server.

Valid entries	Usage
y/n	Enter **y** to allow IP endpoints to be connected through the IP circuit pack in the server/switch in IP format, without going through the Avaya DEFINITY TDM bus.

IP Softphone

Appears only for DCP station types and IP Telephones.

Valid entries	Usage
y/n	Enter **y** indicate that this extension is either a PC-based multifunction station or part of a telecommuter complex with a call-back audio connection.

ITC (Information Transfer Capability)

Indicates the type of transmission facilities to be used for ISDN calls originated from this endpoint. The field does not display for voice-only or BRI stations.

Valid entries	Usage
restricted	If you set to **restricted**, either restricted or unrestricted transmission facilities are used to complete the call. A restricted facility is a transmission facility that enforces 1’s density digital transmission (that is, a sequence of 8 digital zeros are converted to a sequence of 7 zeros and a digital 1).
unrestricted	If you set to **unrestricted**, only unrestricted transmission facilities are used to complete the call. An unrestricted facility is a transmission facility that does not enforce 1’s density digital transmission (that is, digital information is sent exactly as is).

Lock Messages

Valid entries	Usage
y/n	Enter **y** to restrict other users from reading or canceling the voice messages or retrieving messages via Voice Message Retrieval.

Loss Group

This field determines which administered 2-party row in the loss plan applies to each station. Does not appear for stations that do not use loss (for example, x-mobile stations and MASI terminals).

Valid entries	Usage
1 to **17**	Shows the index into the loss plan and tone plans.

LWC Activation

Valid entries	Usage
y/n	Enter **y** to allow internal telephone users to leave short LWC messages for this extension. If the system has hospitality, enter **y** for guest-room telephones if the extension designated to receive failed wakeup messages should receive LWC messages that indicate the wakeup calls failed. Enter **y** if **LWC Reception** is **audix**.

LWC Log External Calls

Appears only where the **LWC Reception** field is available. When external calls are not answered, Communication Manager keeps a record of up to 15 calls (provided information on the caller identification is available) and the telephone's message lamp lights. The telephone set displays the names and numbers of unsuccessful callers.

Valid entries	Usage
y/n	Enter **y** to make unanswered external call logs available to end-users. Each record consists of the latest call attempt date and time.

LWC Reception

Valid entries	Usage
audix	Enter **audix** if the messages are stored on the Audio Information Exchange System.
none	Enter **none** if LWC messages will not be stored.
spe	Enter **spe** if LWC messages are stored in the system or on the switch processor element (spe).

Mapping Mode

Controls the mode of operation in which the cell telephone will operate when mapped to this XMOBILE extension. An XMOBILE station can be bridged to a deskset. These restrictions/ modes exist because the COR of a bridge is ignored; instead the principal's COR is used. This field allows an XMOBILE station to function as a bridge and still be restricted.

When a cell telephone is mapped to more than one XMOBILE station, then only one of the mapped XMOBILE station may have **origination** or **both** as its Mapping Mode. Therefore, only one of the XMOBILE stations mapped to the cell telephone number is permitted to originate calls.

Valid entries	Usage
both	The cell telephone can be used to originate and terminate calls from its associated XMOBILE extension. This is the default when the **XMOBILE Type** field is **PHS** or **DECT**.
none	The XMOBILE station is disabled administratively and cannot originate and terminate calls from its associated internal extension.
origination	The cell telephone may be used only to originate calls from its associated internal XMOBILE extension by dialing into the office server running Communication Manager.
termination	The cell telephone may be used only to terminate calls from its associated internal XMOBILE extension. This is the default when the **XMOBILE Type** field is **EC500**.

Map-to Station

This is the physical telephone used for calls to a virtual extension. Do not use an xmobile, xdid or another virtual extension.

Valid entries	Usage
Valid extension number	Enter the extension of the physical telephone used for calls to a virtual extension.

Media Complex Ext

When used with Multi-media Call Handling, indicates which extension is assigned to the data module of the multimedia complex. Users can dial this extension to place either a voice or a data call, and voice conversion, coverage, and forwarding apply as if the call were made to the 1-number.

For an IP Telephone or an IP Softphone, this is the extension already administered as an H.323 station type. You can administer this field if the **IP Station** field on the **System-Parameters Customer-Options** screen is **y**.

Valid entries	Usage
A valid BRI data extension	For MMCH, enter the extension of the data module that is part of this multimedia complex.
H.323 station extension	For 4600 series IP Telephones, enter the corresponding H.323 station. For IP Softphone, enter the corresponding H.323 station. If you enter a value in this field, you can register this station for either a road-warrior or telecommuter/Avaya IP Agent application.
blank	Leave this field blank for single-connect IP applications.

Message Lamp Ext

Enter the extension of the station you want to track with the message waiting lamp.

Message Server Name

Specifies which Message Server is associated with the station.

Valid entries	Usage
Names administered in alphabetical order	Must contain a user-defined adjunct name that was previously administered in the **Node-Names** screen.

Message Waiting Indicator

This field appears only for ISDN-BRI data modules and for 500, 2500, K2500, 7104A, 6210, 6218, 6220, 8110, H.323 and VRU telephones. (This field is independent of the administration of the Caller ID Message Waiting Indication for CallrID phones.) Must be set to a value other than **none** when the **Type** field is set to **H.323**.

Valid entries	Usage
led	Enter **led** if the message waiting indicator is a light-emitting diode (LED).
neon	Enter **neon** if the message waiting indicator is a neon indicator.
none	Default. No message waiting indicator is selected.

MIM Mtce/Mgt

Indicates if the telephone supports MIM Maintenance and Management capabilities other then endpoint initialization. Appears only if **MIM Support** is **y**.

MIM Support (Management Information Message Support)

This field appears only for ISDN-BRI data modules and ASAI. This field supports MIM endpoint initialization (SPID support) and other Maintenance or Management capabilities.

Valid entries	Usage
y	Enter **y** to display Endpt Init and MIM Mtce/Mgt.
n	Enter **n** for **ASAI**.

Mobility Trunk Group

This field associates the XMOBILE station to a trunk.

Valid entries	Usage
1 to **x** (where ***x*** is 666) for DEFINITY R **2000** for S8700 or S8710 IP Connect and S8700 or S8710 Multi-Connect **99** for DEFINITY CSI, S8300 Media Server blank	Enter a valid trunk group number for mobility routing. This trunk group is used for routing. The **Configuration Set** field may be blank if the trunk group is **DECT** or **PHS**. If the trunk group is non-DECT or non-PHS, administer the **Configuration Set** field.
aar	The routing capabilities of Communication Manager will be used to direct the call to an ISDN trunk. If no ISDN trunk is available, the call will not be extended out of the media server. It will provide ringback to the calling company and may eventually go to coverage.
ars	The routing capabilities of Communication Manager will be used to direct the call to an ISDN trunk. If no ISDN trunk is available, the call will not be extended out of the media server. It will provide ringback to the calling company and may eventually go to coverage.

Model

This field appears only for NI-BRI telephones.

Valid entries	Usage
L-3 Communication STE	The NI-BRI telephone is a model L-3 Communication STE.
Tone Commander	The NI-BRI telephone is a model 6210 and 6220 Tone Commander.
Other	The NI-BRI telephone is another model (for example, a Nortel 5317T).

Multimedia Early Answer

Allows you to establish multimedia early answer on a station-by-station basis.

Valid entries	Usage
y/n	If this station will receive coverage calls for multimedia complexes, but is not multimedia-capable, enter **y** to ensure that calls are converted and talk path is established before ringing at this station.

Mute Button Enabled

Valid entries	Usage
y/n	Enter **y** to allow the user to use the **Mute** button on this telephone.

MWI Served User Type

Controls the auditing or interrogation of a served user's message waiting indicator (MWI).

Valid entries	Usage
fp-mwi	Use if the station is a served user of an fp-mwi message center.
qsig-mwi	Use if the station is a served user of a qsig-mwi message center.
blank	Leave this field blank if you do not want to audit the served user's MWI or if the user is not a served user of either an fp-mwi or qsig-mwi message center.

Name

Enter a name for the person associated with this telephone or data module. The system uses the **Name** field to create the system Directory.

Note:

For 4620SW or 4610SW telephones, the **Name** field is supported by Unicode language display. For more information on Unicode language display, see Administering Unicode display on page 181.

Off Premises Station

Analog phones only.

Valid entries	Usage
y	Enter **y** if this telephone is not located in the same building with the system. If you enter **y**, you must complete R Balance Network.
n	Enter **n** if the telephone is located in the same building with the system.

PCOL/TEG Call Alerting

Appears only for 510 telephones.

Valid entries	Usage
y/n	Enter **y** to alert the station for Personal CO Line/ Terminating Extension Group calls.

Per Button Ring Control

Valid entries	Usage
y	Enter **y** to allow users to select ring behavior individually for each call-appr, brdg-appr, or abrdg-appr on the station and to enable Automatic Abbreviated and Delayed ring transition for each call-appr on the station. Also, enter **y** if you do not want the system to automatically move the line selection to a silently alerting call unless that call was audibly ringing earlier.
n	Enter **n** if you want calls on **call-appr** buttons to always ring the station and calls on **brdg-appr** or **abrdg-appr** buttons to always ring or not ring based on the **Bridged Call Alerting** field value. Also, enter **n** if you want the system to move line selection to a silently alerting call if there is no call audibly ringing the station.

Personalized Ringing Pattern

Personalized Ringing allows users of some telephones to have one of 8 ringing patterns for incoming calls. Users working closely in the same area can each specify a different ringing pattern. This enables the users to distinguish their own ringing telephone from other telephones in the same area. For virtual stations, this field dictates the ringing pattern on its mapped-to physical telephone.

Enter a Personalized Ringing Pattern. (L = 530 Hz, M = 750 Hz, and H = 1060 Hz).

Valid entries	Usage
1	MMM (standard ringing)
2	HHH
3	LLL
4	LHH
5	HHL
6	HLL
7	HLH
8	LHL

Per Station CPN - Send Calling Number

Valid entries	Usage
y	All outgoing calls from the station will deliver the Calling Party Number (CPN) information as "Presentation Allowed."
n	No CPN information is sent for the call.
r	Outgoing non-DCS network calls from the station will deliver the Calling Party Number information as "Presentation Restricted."
blank	The sending of CPN information for calls is controlled by any administration on the outgoing trunk group the calls are carried on.

Port

Enter 7 characters to specify a port, or an x. If this extension is registered as an IP endpoint, this field will display ***sXXXXXX***, where ***XXXXXX*** is the number of previously registered IP stations. For example, if there are 312 IP sets already registered when you register, your extension would get port **s000313**.

Valid entries	Usage
01 to **44** (DEFINITY R)	First and second numbers are the cabinet number
01 to **03** (DEFINITY SI)	
A to **E**	Third character is the carrier
01 to **20**	Fourth and fifth characters are the slot number
01 to **32**	Sixth and seventh characters are the circuit number
x	Indicates that there is no hardware associated with the port assignment since the switch was set up, and the administrator expects that the extension would have a non-IP set. Or, the extension had a non-IP set, and it dissociated. Use **x** for AWOH and CTI stations, as well as for SBS Extensions.
IP	Indicates that there is no hardware associated with the port assignment since the switch was set up, and the administrator expects that the extension would have an IP set. This is automatically entered for certain IP station set types, but can be entered for a DCP set with softphone permissions. This changes to the s00000 type when the set registers.

For DCP sets, the port can only be assigned once. ISDN-BRI provides a multipoint configuration capability that allows a previously assigned port to be specified more than once as follows: 2 stand-alone voice endpoints, 2 stand-alone data endpoints, or 1 integrated voice and data endpoint.

However, for the following cases, the port is assumed to be fully assigned:

- Maximum number of users (currently 2) are assigned on the port.
- One of the users on the port is a fixed TEI station.
- One of the users on the port has B-channel voice and B-channel data capability.
- One of the users on the port has no SPID assigned, which includes telephones that have no SPID initialization capability.
- This field is display-only for H.323 set types and 4600 set types. The system assigns an "IP" when the station is first administered.

Note:

Port 1720 is turned off by default to minimize denial of service situations. This applies to all IP softphones release 5.2 or later. You can change this setting, if you have root privileges on the system, by typing the command: **`/opt/ecs/sbin ACL 1720 on`** or **`off`**.

Precedence Call Waiting

Valid entries	Usage
y/n	Enter **y** to activate Precedence Call Waiting for this station.

R Balance Network

Valid entries	Usage
y	Enter **y** to select the R Balance Capacitor network. In all other cases except for those listed under **n**, enter **y**.
n	Enter **n** to select the standard resistor capacitor network. You must complete this field if **Off-Premise Station** is **y**. Enter **n** when the station port circuit is connected to terminal equipment (such as SLC carriers or impedance compensators) optioned for 600-ohm input impedance and the distance to the equipment from the system is less than 3,000 feet.

Recall Rotary Digit

This field only appears if type is 500 or 2500.

Valid entries	Usage
y/n	Enter **y** to allow the user of a rotary telephone to dial the administered Recall Rotary Digit to receive recall dial tone. This will enable this user to perform conference and transfer operations. You establish the Recall Rotary Digit on the **Feature-Related System Parameters** screen.

Redirect Notification

Valid entries	Usage
y/n	Enter **y** to give a half ring at this telephone when calls to this extension are redirected (via Call Forwarding or Call Coverage). Enter **y** if **LWC Reception** is **audix**.

Remote Office Phone

Valid entries	Usage
y/n	Enter **y** to use this station as an endpoint in a remote office configuration.

Remote Softphone Emergency Calls

Use this field to tell Communication Manager how to handle emergency calls from the IP telephone. This field appears when either the **IP Softphone** field or the **Remote Office Station** field is set to **y** on the **Station** screen.

CAUTION:

An Avaya IP endpoint can dial emergency calls (for example, 911 calls in the U.S.). It only reaches the local emergency service in the Public Safety Answering Point area where the telephone system has local trunks. Please be advised that an Avaya IP endpoint cannot dial to and connect with local emergency service when dialing from remote locations that do not have local trunks. You should not use an Avaya IP endpoint to dial emergency numbers for emergency services when dialing from remote locations. Avaya Inc. is not responsible or liable for any damages resulting from misplaced emergency calls made from an Avaya endpoint. Your use of this product indicates that you have read this advisory and agree to use an alternative telephone to dial all emergency calls from remote locations. Please contact your Avaya respresentative if you have questions about emergency calls from IP phones.

Valid entries	Usage

as-on-local	Type **as-on-local** to achieve the following results:
	• If the administrator chooses to leave the **Emergency Location Extension** fields (that correspond to this station's IP address) on the **IP Address Mapping** screen blank, the value **as-on-local** sends the extension entered in the **Emergency Location Extension** field in the **Station** screen to the Public Safety Answering Point (PSAP).
	• If the administrator populates the **IP Address Mapping** screen with emergency numbers, the value **as-on-local** functions as follows:
	  • If the **Emergency Location Extension** field in the **Station** screen is the same as the **Emergency Location Extension** field in the **IP Address Mapping** screen, the value **as-on-local** sends the extension to the Public Safety Answering Point (PSAP).
	  • If the **Emergency Location Extension** field in the **Station** screen is different from the **Emergency Location Extension** field in the **IP Address Mapping** screen, the value **as-on-local** sends the extension in the **IP Address Mapping** screen to the Public Safety Answering Point (PSAP).

block	Enter **block** to prevent the completion of emergency calls. Use this entry for users who move around but always have a circuit-switched telephone nearby, and for users who are farther away from the media server or switch than an adjacent area code served by the same 911 Tandem office. When users attempt to dial an emergency call from an IP Telephone and the call is blocked, they can dial 911 from a nearby circuit-switched telephone instead.

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Valid entries	Usage
cesid	Enter **cesid** to allow Communication Manager to send the CESID information supplied by the IP Softphone to the PSAP. The end user enters the emergency information into the IP Softphone. Use this entry for IP Softphones with road warrior service that are near enough to the media server or switch that an emergency call routed over the it's trunk reaches the PSAP that covers the server or switch. If the media server or switch uses ISDN trunks for emergency calls, the digit string is the telephone number, provided that the number is a local direct-dial number with the local area code, at the physical location of the IP Softphone. If the media server or switch uses CAMA trunks for emergency calls, the end user enters a specific digit string for each IP Softphone location, based on advice from the local emergency response personnel.
option	Enter **option** to allow the user to select the option (extension, block, or cesid) that the user selected during registration and the IP Softphone reported. Use this entry for extensions that may be swapped back and forth between IP Softphones and a telephone with a fixed location. The user chooses between **block** and **cesid** on the softphone. A DCP or IP telephone in the office automatically selects **extension**.
	2 of 2

Restrict Last Appearance

Valid entries	Usage
y/n	Enter **y** to restrict the last idle call appearance for incoming priority calls and outgoing call originations only.

Room

Valid entries	Usage
Up to 10 characters	To identify the telephone location.
Up to 5 characters	To identify the guest room number, if this station is one of several to be assigned a guest room and the **Display Room Information in Call Display** is **y** on the **Hospitality-Related System Parameters** screen.

Security Code

Enter the security code required by users for specific system features and functions, including the following: Personal Station Access, Redirection of Calls Coverage Off-Net, Leave Word Calling, Extended Call Forwarding, Station Lock, Message Retrieval, Terminal Self-Administration, and Demand Printing. The required security code length is determined by **Minimum Security Code Length** on the **Feature-Related System Parameters** screen.

Select Last Used Appearance

Valid entries	Usage
y	Enter **y** to indicate a station's line selection is not to be moved from the currently selected line button to a different, non-alerting line button. If you enter **y**, the line selection on an on-hook station only moves from the last used line button to a line button with an audibly alerting call. If there are no alerting calls, the line selection remains on the button last used for a call.
n	Enter **n** so the line selection on an on-hook station with no alerting calls can be moved to a different line button, which may be serving a different extension.

Service Link Mode

The service link is the combined hardware and software multimedia connection between an Enhanced mode complex's H.320 DVC system and a server running Avaya Communication Manager which terminates the H.320 protocol. When the user receives or makes a call during a multimedia or IP Softphone or IP Telephone session, a "service link" is established.

Valid entries	Usage
as-needed	Use this setting for most multimedia, IP Softphone, or IP Telephone users. Setting the Service Link Mode to as-needed leaves the service link connected for 10 seconds after the user ends a call so that they can immediately place or take another call. After 10 seconds the link is dropped and a new link would have to be established to place or take another call.
permanent	Use **permanent** for busy call center agents and other users who are constantly placing or receiving multimedia, IP Softphone, or IP Telephone calls. In permanent mode, the service link stays up for the duration of the multimedia, IP Softphone, or IP Telephone application session.

Set Color

Enter a valid set color as defined in the **Signaling Group Administered NCA TSC Assignments page** screen. Valid entries include the following colors: beige, black, blue, brown, burg (burgundy), gray, green, ivory, orng (orange), red, teak, wal (walnut), white, and yel (yellow).

Speakerphone

This field controls the behavior of speakerphones.

Valid entries	Usage
1-way	Enter **1-way** to indicate that you want the speakerphone to be listen-only.
2-way	Enter **2-way** to indicate that you want the speakerphone to be both talk and listen.
grp-listen	Group Listen works with only 6400-series and 2420/2410 phones. Group Listen allows a telephone user to talk and listen to another party with the handset or headset while the telephone's 2-way speakerphone is in the listen-only mode. Others in the room can listen, but cannot speak to the other party via the speakerphone. The person talking on the handset acts as the spokesperson for the group. Group Listen provides reduced background noise and improves clarity during a conference call when a group needs to discuss what is being communicated to another party.
none	

Special Dialing Option

This field identifies the type of dialing for calls when this data module originates calls.

Valid entries	Usage
hot-line	
default	
blank	For regular (normal) keyboard dialing.

SPID — (Service Profile Identifier)

Enter a variable length parameter. This field appears only if the **Endpt Init** field is **y**.

The SPID is a numeric string, which means that the value of 00 is different from 000. The SPID must be different for all terminals on the BRI and from the Service SPID. The SPID should always be assigned. If the SPID is not assigned for the first BRI on a port, any other BRI assignment to that port are blocked.

Note:

If you have set the **Port** field to **X** for an ISDN-BRI extension and intend to use Terminal Translation Initialization (TTI) to assign the port, then the SPID number must equal the station number.

Survivable GK Node Name

Appears only when the **Type** field is **46xx**.This will cause ACM call processing to make an entry with the "survivable GK node name" at the bottom of the AlternateGK list. This will be sent down as a RAS message at registration time.

Valid entries	Usage
IP node name or blank	Enter any valid IP node name as administered on the **IP Node Names** screen to allow the station to be registered as an IP telelphone, associated to an H.323 gateway that is capable of supporting a gatekeeper in survivable mode. Default is blank.

Switchhook Flash

Must be set to **y** when the **Type** field is set to **H.323**.

Valid entries	Usage
y	Enter **y** to allow users to use the switchhook flash function to activate Conference/Transfer/Hold and Call Waiting.
n	Enter **n** to disable the flash function so that when the switchhook is pressed while active on a call, the call drops. If this field is **n**, you must set **Call Waiting Indication** to **n**.

TEI

Appears only when the **Fixed TEI** field is **y**.

Valid entries	Usage
0 to **63**	1 or 2-digit number

Tests

Valid entries	Usage
y	Enter **y** to enable port maintenance tests.
n	If the equipment (dictaphone) connected to the port does not support these tests, you must enter **n**.

TN

Enter the Tenant Partition number. Also, SBS Extensions may be partitioned.

Type

For each station that you want to add to your system, you must specify the type of telephone in the **Type** field. This is how you distinguish between the many different types of telephones.

The following table lists the telephones, virtual phones, and personal computers that you can administer on Avaya Communication Manager. To administer telephones that are not in the table, use the **Alias Station** screen.

Note:

You cannot change an analog telephone administered with hardware to a virtual extension if **TTI** is **y** on the **Feature-Related System Parameters** screen. Contact your Avaya representative for more information.

Table 15: Telephones *1 of 5*

Telephone type	Model	Administer as
Single-line analog	500	500
	2500, 2500 w/ Message Waiting Adjunct	2500
	6210	6210
	6211	6210
	6218	6218
	6219	6218
	6220	6220
	6221	6220

1 of 5

Table 15: Telephones *2 of 5*

Telephone type	Model	Administer as
	Analog telephone w/Caller ID	CallrID
	7101A, 7102A	7101A
	7103A Programmable and Original	7103A
	7104A	7104A
	8110	8110
	DS1FD	DS1FD
	7302H, 7303H	7303S
	VRU (voice response unit) with C&D tones	VRU
	VRU without C&D tones	2500
Single-line DS1/DSO (Lineside T1/DS1)	DS1 device without forward disconnect	ops
	VRU with forward disconnect without C&D tones	ds1fd or ds1sa
	VRU with forward disconnect without C&D tones	VRUFD or VRUSA
Terminals	510D	510
	515BCT	515
Multiappearance hybrid	7303S	7303S, 7313H
	7305H	7305S
	7305S	7305S, 7316H, 7317H
	7309H	7309H, 7313H
	7313H	7313H
	7314H	7314H
	7315H	7315H
	7316H	7316H
	7317H	7317H

2 of 5

Table 15: Telephones *3 of 5*

Telephone type	Model	Administer as
Multiappearance digital	2402	2402
	2410	2410
	2420	2420
	6402	6402
	6402D	6402D
	6408	6408
	6408+	6408+
	6408D	6408D
	6408D+	6408D+
	6416D+	6416D+
	6424D+	6424D+
	7401D	7401D
	7401+	7401+
	7403D	7403D
	7404D	7404D
	7405D	7405D
	7406D	7406D
	7406+	7406+
	7407D	7407D
	7407+	7407+
	7410D	7410D
	7410+	7410+
	7434D	7434D
	7444D	7444D
	8403B	8403B
	8405B	8405B
	8405B+	8405B+

3 of 5

Table 15: Telephones *4 of 5*

Telephone type	Model	Administer as
	8405D	8405D
	8405D+	8405D+
	8410B	8410B
	8410D	8410D
	8411B	8411B
	8411D	8411D
	8434D	8434D
	CALLMASTER I	602A1
	CALLMASTER II, III, IV	603A1, 603D1, 603E1, 603F1
	CALLMASTER VI	606A1
	IDT1	7403D
	IDT2	7406D
IP Telephone	4601	4601
	4602	4602
	4606	4606
	4610	4610
	4612	4612
	4620 IP (G3 hardware)	4620
	4620SW IP (G3.5 hardware)	4620
	4624	4624
IP SoftPhone	Road-warrior application	H.323 or DCP type
	Native H.323	H.323
	Single-connect	H.323 or DCP type
ISDN-BRI station	—	asai
	Any NI-BRI (N1 and N2) telephone	NI-BRI
		4 of 5

Table 15: Telephones *5 of 5*

Telephone type	Model	Administer as
	7505D	7505D
	7506D	7506D
	7507D	7507D
	8503D	8503D
	8510T	8510T
	8520T	8520T
Personal computer	6300/7300	PC
(voice/data)	6538/9	Constellation
Test Line	ATMS	105TL
No hardware assigned at the time of administration.		XDID (use when Avaya Communication Manager later assigns a DID number to this station) XDIDVIP (use when the administrator later assigns a DID number to this station) virtual (use to map this and other extensions to one physical telephone)
Key telephone system interface	—	K2500
ASAI	asai link computer telephony adjunct link	asai adjlk
AWOH	any digital set CTI station	same as "Multiappearance Digital" see table above CTI
CTI	CTI station	CTI
XMOBILE	EC500, DECT, PHS	XMOBILE
ISDN-BRI data module	7500	7500
SBS Extension	SBS test extension (no hardware)	sbs

5 of 5

XID

Appears only for an ISDN-BRI data module or an ASAI link. Used to identify Layer 2 XID testing capability.

XMOBILE Type

When the **Type** field is **XMOBILE**, the **Mobility Trunk Group** field must be administered.

Valid entries	Usage
DECT	For the DECT Access System or the AGCS (ROAMEO) IS-136 (TDMA cellular).
EC500	For any public cellular networks.
PHS	For the DENSO 300M.

System Parameters Call Coverage/Call Forwarding

This screen sets the system-wide parameters for call coverage and call forwarding.

Field descriptions for page 1

Figure 508: System-Parameters — Call Coverage/Call Forwarding screen

```
 change system-parameters coverage-forwarding                    page 1 of x

         SYSTEM PARAMETERS -- CALL COVERAGE / CALL FORWARDING

CALL COVERAGE/FORWARDING PARAMETERS
           Local Cvg Subsequent Redirection/CFWD No Ans Interval (rings): _
         Off-Net Cvg Subsequent Redirection/CFWD No Ans Interval (rings): _
                          Coverage - Caller Response Interval (seconds): _
     Threshold for Blocking Off-Net Redirection of Incoming Trunks Calls: 1

COVERAGE
                                        Keep Held SBA at Coverage Point? _
        External Coverage Treatment for Transferred Incoming Trunk Calls? _
         Immediate Redirection on Receipt of PROGRESS Inband Information? _
                                             Maintain SBA At Principal? _
              QSIG VALU Coverage Overrides QSIG Diversion with Rerouting? _
                                          Station Hunt Before Coverage? n

FORWARDING
                                                 Call Forward Override? _
                                               Coverage After Forwarding? _
```

CALL COVERAGE / FORWARDING PARAMETERS

Coverage - Caller Response Interval (seconds)

The time in seconds the caller (internal caller only) has before a call redirects to the called party's first coverage point. The calling party either can hang up, use Leave Word Calling, or press the **Go to Cover** button during this time interval.

Valid entries	Usage
0 to **10**	Enter the time in seconds the caller (internal caller only) has before a call redirects to the called party's first coverage point.

Local Cvg Subsequent Redirection/CFWD No Ans Interval (rings)

This field specifies:

- the number of rings applied at a local coverage point before a call redirects to the next coverage point
- the number of rings applied at the principal before a call forwards when Call Forwarding Busy/Don't Answer is activated.

Valid entries	Usage
1 to **99**	See note below.

Off-Net Cvg Subsequent Redirection/CFWD No Ans Interval (rings)

This field specifies:

- the number of rings applied at an off-net coverage point before a call is redirected to the next coverage point
- the number of rings applied at an off-net forwarded-to destination before the call is redirected to coverage.

Valid entries	Usage
1 to **99**	See note below.

Note:

When ringing local destinations (say in an office environment), a short interval often is appropriate because the intended party either is near the telephone or not present. However, when ringing off-net locations, you cannot assume how near the intended party is to the telephone. If the call is left at an off-net destination for only a short interval, the call may be redirected to the next destination before the intended party has any real chance of answering the call.

Threshold for Blocking Off-Net Redirection of Incoming Trunk Calls

This field applies for those occasions when an incoming call to a station redirects off-net. At that time, the Call Forward timer activates to block any further incoming calls to that station from being redirected off-net until the timer expires.

Valid entries	Usage
1 to **7**	The number of allowed calls to be routed off-net before blocking commences.
n (all)	Call processing never activates the Call Forward timer. Therefore, any number of calls to a principal may be redirected off-net.

COVERAGE

External Coverage Treatment for Transferred Incoming Trunk Calls

This field governs how an transferred incoming trunk call is handled if the call redirects to coverage.

Valid entries	Usage
y	Enter **y** to allow external coverage treatment for incoming trunk calls that redirect to coverage.
n	Enter **n** to allow internal coverage treatment for incoming trunk calls that redirect to coverage

Immediate Redirection on Receipt of PROGRESS Inband Information

This field appears only if one of the following is true:

- The **Coverage of Calls Redirected Off-Net Enabled** field on the **System Parameters Coverage/Forwarding** screen is **y**.
- The **Value-Added Avaya (VALU)** field on the **System-Parameters Customer-Options**, Page 6, screen is **y**.

This field pertains only to CCRON and QSIG VALU coverage calls redirected over end-to-end ISDN facilities. Some cellular telephone providers send an ISDN PROGRESS message with the **Progress Indicator** field set to 'inband information' when a cellular telephone is unavailable to receive a call. In these circumstances, the message indicates that an announcement is being played to the originating party and we should move the call immediately to the next available coverage point.

However, a PROGRESS message with a Progress Indicator of 'inband information' may be received for reasons other than an unavailable cellular telephone. In this case, you probably do not want to redirect the call to the next coverage point.

There is no way for Avaya Communication Manager to know the actual intent of such a PROGRESS message, yet you may choose how the system should handle this message. It is essentially an educated, but blind, choice and you should be aware that there will be instances when this choice leads to inappropriate call handling.

Avaya Communication Manager queries this field on receipt of a qualifying PROGRESS message and acts according to your instruction on how to treat it.

As a guide, users in European countries following the ETSI standard and redirecting to GSM cellular phones may want to consider setting this field to **y**.

In the United States, PROGRESS messages with the **Progress Indicator** field set to 'inband information' are sent for a variety of reasons not associated with unavailable cell phones and you should set this field to **n**.

Valid entries	Usage
y	Immediately redirect an off-net coverage/forwarded call to the next coverage point.
n	Do not immediately redirect an off-net coverage/ forwarded call to the next coverage point.

Keep Held SBA at Coverage Point

This field governs how a covering user who has placed an answered coverage call on hold is treated if the original principal bridges onto the call.

Valid entries	Usage
y	Keeps the coverage party on the call. The coverage party remains on hold, but may enter the call along with the principal and the calling party.
n	Drops the coverage party from the call.

Maintain SBA At Principal

Allows a user to maintain a simulated bridged appearance when a call redirects to coverage.

Valid entries	Usage
y	Enter **y** to maintain a simulated bridged appearance (SBA) on the principal's telephone when a call redirects to coverage. DCS with rerouting will not be attempted after coverage.
n	When set to **n**, no SBA is maintained on the principal's telephone. DCS with rerouting will be attempted, and if successful, the principal will lose the bridged appearance and the ability to bridge onto the coverage call.

QSIG VALU Coverage Overrides QSIG Diversion with Rerouting

This field appears if, on the **System-Parameters Customer-Options** screen, the **Basic Supplementary Services** and **Supplementary Services with Rerouting** fields are both set to **y**.

This field specifies whether, with both QSIG Diversion with Rerouting and QSIG VALU turned on, the Coverage After Forwarding option on the **Station** screen will work for a user for calls that go to remote coverage. Normally, with QSIG Diversion with Rerouting turned on, the local system passes control of a forwarded call to the remote QSIG media server or switch on which the forwarding destination resides. In this case, the forwarded call cannot return to coverage for the user who originally received the call, even when the remote destination is busy or does not answer.

However, you can enter **y** in this field to have QSIG VALU call coverage take precedence. In this case, if the **Coverage After Forwarding** option on the **Station** screen is enabled for a user, then QSIG Diversion with Rerouting is not be attempted.

Valid entries	Usage
y/n	Enter **y** to allow Coverage After Forwarding to work when it is activated on a user's **Station** screen and Diversion with Rerouting is also turned on.

Station Hunt Before Coverage

This field allows you to choose whether a call to a busy station performs station hunting before going to coverage.

Valid entries	Usage
y/n	Enter **y** to use Station Hunt Before Coverage.

FORWARDING

Call Forward Override

This field specifies how to treat a call from a forwarded-to party to the forwarded-from party.

Valid entries	Usage
y	Overrides the Call Forwarding feature by allowing a forwarded-to station to complete a call to the forwarded-from station.
n	Directs the system to forward calls to a station even when they are from the forwarded-to party.

Coverage After Forwarding

This field governs whether an unanswered forwarded call is provided coverage treatment.

Valid entries	Usage
y	Coverage treatment is provided to unanswered forwarded calls.
n	No coverage treatment is provided to unanswered forwarded calls. The call remains at the forwarded-to destination.

Field descriptions for page 2

Figure 509: System-Parameters Coverage-Forwarding screen

```
change system-parameters coverage-forwarding                    page 2 of x

           SYSTEM PARAMETERS -- CALL COVERAGE / CALL FORWARDING

COVERAGE OF CALLS REDIRECTED OFF-NET (CCRON)
                            Coverage Of Calls Redirected Off-Net Enabled? y
      Activate Answer Detection (Preserves SBA) On Final CCRON Cvg Point? y
                                      Ignore Network Answer Supervision? y
                       Disable call classifier for CCRON over ISDN trunks? n
                       Disable call classifier for CCRON over SIP trunks?  n
```

COVERAGE OF CALLS REDIRECTED OFF-NET (CCRON)

Activate Answer Detection (Preserves SBA) On Final CCRON Cvg Point

This field appears only if the **Coverage of Calls Redirected Off-Net Enabled** field on this screen is **y**.

When the system redirects a call off-net at the final coverage point in a coverage path, the system can apply no further coverage treatment even if the call is unanswered. The only reason for activating answer detection on such a call is to maintain the simulated bridged appearance (SBA) on the principal's telephone that allows the principal to answer or bridge onto the call. However, when the system monitors the call through use of a call classifier port, there is an inherent cut-through delay following the detection of answer at the far end. This field has no consequence when the off-net call is carried end-to-end by ISDN facilities; the SBA is maintained and there is no cut-through delay.

Valid entries	Usage
y	Directs the system to maintain a simulated bridged appearance on the principal when redirecting to a final off-net coverage point.
n	Allows the system to drop the SBA on the principal's telephone when the call redirects off-net at the last coverage point, eliminating the cut-through delay inherent in CCRON calls, but sacrificing the principal's ability to answer the call.

Coverage Of Calls Redirected Off-Net Enabled

This field allows you to enable/disable the Coverage of Calls Redirected Off-Net (CCRON) feature. This field provides a quick means of disabling this feature if the demand on the call classifier port resources degrades other services provided by Communication Manager.

To set this field to **y**, first set the **Coverage Of Calls Redirected Off-net** field on the **System-Parameters Customer-Options** screen to **y**. The **Coverage of Calls Redirected Off-Net** field on this screen must be **y** to administer this field.

Valid entries	Usage
y	Avaya Communication Manager monitors off-net coverage/forwarded calls and provides further coverage treatment for unanswered calls.
n	Avaya Communication Manager does not monitor off-net coverage/forwarded calls. No further coverage treatment is provided if such calls are unanswered.

Disable call classifier for CCRON over ISDN trunks

When a CCRON call routes offnet over ISDN end-to-end facilities, no call classifier is attached to the call. If, subsequently during the call, an ISDN PROGRESS or ALERT message is received that indicates that interworking has occurred, a call classifier is normally attached to the call and assumes precedences over ISDN trunk signalling. This field provides a customer the means of directing Communication Manager to dispense with the call classifier on interworked calls and rely on the ISDN trunk signalling messages.

Valid entries	Usage
y	Use **y** to disable the call classifier for CCRON calls over interworked trunk facilities.
n	Use **n** to enable the call classifier for CCRON calls over interworked trunk facilities.

Disable call classifier for CCRON over SIP trunks

This field provides a customer the means of directing Communication Manager to dispense with the call classifier on interworked calls and rely on the SIP trunk signalling messages.

Valid entries	Usage
y	Use **y** to disable the call classifier for CCRON calls over interworked trunk facilities.
n	Use **n** to enable the call classifier for CCRON calls over interworked trunk facilities.

Ignore Network Answer Supervision

This field appears only if the **Coverage of Calls Redirected Off-Net Enabled** field on this screen is **y**.

CCRON may use a call classifier port to determine whether an off-net coverage or forwarded call has been answered, discarding other information that may indicate an answered state. However, some customers pay the operating company to provide network answer supervision on their trunks and desire that CCRON not discard that information. They may preserve this service by setting this field to **n**.

On the other hand, beware when you tandem a call over a tie trunk through another media server or switch node from where it redirects to the public network over non-ISDN facilities. If the trunk on the far-end node sends a timed answer supervision, that may get tandemed back to the originating node as a network answer. In such a scenario, the originating media server or switch interprets the call as answered, leading to some undesirable behavior. To avoid these calls from mistakenly be treated as answered, set this field to **y**. An unfortunate consequence is that a short cut-through delay that is inherent to call classification is introduced when the call is answered.

Valid entries	Usage
y	Ignore network answer supervision and rely on the call classifier to determine when a call is answered.
n	Treat network answer supervision as a true answer.

System Parameters Country-Options

This screen implements parameters associated with certain international (including North American) call characteristics. You cannot change this screen. See your Avaya technical support representative if you want to modify any of the values here.

See the following table for country codes throughout this screen.

Country code table

Code	Country
1	United States, Canada
2	Australia, New Zealand
3	Japan
4	Italy
5	Netherlands
6	Singapore
7	Mexico
8	Belgium, Luxembourg
9	Saudi Arabia
10	United Kingdom
11	Spain
12	France
13	Germany
14	Czech Republic, Slovakia
15	Russia (CIS)
16	Argentina
17	Greece

1 of 2

Code	Country
18	China
19	Hong Kong
20	Thailand
21	Macedonia
22	Poland
23	Brazil
24	Nordic
25	South Africa
	2 of 2

Field descriptions for page 1

Figure 510: System Parameters Country-Options screen

```
change system-parameters country-options                        Page 1 of 1

                        SYSTEM PARAMETERS COUNTRY-OPTIONS

                      Set Layer 1 timer T1 to 30 seconds? n
        Display Character Set? Roman
       Howler Tone After Busy? y  Disconnect on No Answer by Call Type? y
              Enable Busy Tone Disconnect for Analog Loop-start Trunks?

TONE DETECTION PARAMETERS
        Tone Detection Mode: 5       dial tone validation timer (sec): 60
          Interdigit Pause: short
```

Dial Tone Validation Timer (sec)

This field appears only when **Tone Detection Mode** is equal to **4** or **5**. (Valid with TN420C or later Tone Detector circuit pack.).

Valid entries	Usage
0 to **6375** in increments of 25	Displays number of milliseconds that the dial tone validation routine will use to sample transmissions.

Disconnect on No Answer by Call Type

Drops outgoing trunk calls (except DCS and AAR) that users leave unanswered too long.

Valid entries	Usage
y/n	Enables the system to disconnect calls that are not answered.

Display Character Set

Appears only for 607, 46xx, 64xx, and 84xx telephone sets.

Valid entries	Usage
cyrillic **katakana** **roman** **ukranian**	Indicate the enhanced character set to display. See "Telephone Display" in *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205, for more information. **Note:** Cyrillic, Roman, and Ukranian map to the Eurofont character set.

Enable Busy Tone Disconnect for Analog Loop-start Trunks

This field allows Communication Manager to recognize a busy tone from the central office as a disconnect signal.

Valid entries	Usage
y/n	Enter **y** to allow Communication Manager to disconnect the trunk when a busy tone is received from the central office.

Howler After Busy

Plays howler tone when users leave their analog telephone off-hook too long.

Valid entries	Usage
y/n	Enables howler tone.

Set Layer 1 timer T1 to 30 seconds

Valid entries	Usage
y/n	Specifies whether the Layer 1 timer is set to 30 seconds.

TONE DETECTION PARAMETERS

Interdigit Pause

Specifies the maximum length of the inter-digit pause. Breaks lasting less than this range will be bridged or ignored. (Valid with TN420C or later Tone Detector circuit pack.)

Valid entries	Usage
short	5 to 30ms
long	20 to 40ms

Tone Detection Mode

The country code specifies the type of tone detection used on a TN420B (or later) tone-detection circuit pack.

Valid entries	Usage
1	Precise Italian tone detection algorithm
2	Precise Australian tone detection algorithm
3	Precise UK tone detection algorithm
4	Imprecise normal broadband filter algorithm (valid with TN420C or later Tone Detector circuit pack)
5	Imprecise wideband filter algorithm (valid with TN420C or later Tone Detector circuit pack)
6	

System-Parameters Customer-Options

This screen shows you which optional features are enabled for your system, as determined by the installed license file. All fields on this screen are display only. If you have any questions about disabling or enabling one of these features, contact your Avaya representative.

Field descriptions for page 1

Figure 511: System Parameters Customer-Options screen

```
display system-parameters customer-options                    Page   1 of  x
                              OPTIONAL FEATURES

G3 Version: V12                              RFA System ID (SID):
123456789012
  Location: 2                                RFA Module ID (MID): 123456
  Platform: 2
                                                          USED
                                     Maximum Ports: 300   174
                            Maximum XMOBILE Stations: 30    28
                    Maximum Off-PBX Telephones - EC500: 1200  0
                    Maximum Off-PBX Telephones -   OPS: 1200  0
                    Maximum Off-PBX Telephones - SCCAN: 0     0



        (NOTE: You must logoff & login to effect the permission changes.)
```

G3 Version

Identifies the version of Avaya Communication Manager being used.

Location

Indicates the location of this media server or switch. 1 indicates Canada or the United States. 2 indicates any other location, and allows the use of International Consolidation circuit packs and telephones.

Maximum Off-PBX Telephones - EC500

Default = **0**. A number greater than zero should appear in either this field or the **Maximum Off-PBX Telephones - OPS** field. Stations that are administered for any Extension to Cellular/OPS application count against this limit.

Maximum Off-PBX Telephones - OPS

Default = **0**. A number greater than zero should appear in either this field or the **Maximum Off-PBX Telephones - EC500** field. Stations that are administered for any Extension to Cellular/OPS application count against this limit.

Maximum Off-PBX Telephones - SCCAN

Stations that are administered for any Extension to Cellular/OPS application count against this limit.

Maximum Ports

Number of ports active, per contract.

Maximum XMOBILE Stations

Specifies the maximum number of allowable XMOBILE stations. In general, each XMOBILE station is assigned to a wireless handset. Each XMOBILE station counts as a station and a port in terms of system configuration.

Platform

A display-only field that identifies, via a number mapping, the platform being used for a specific customer.

Field descriptions for page 2

Figure 512: System Parameters Customer-Options screen

```
display system-parameters customer-options                      page 2 of x

                              OPTIONAL FEATURES

IP PORT CAPACITIES
                          Maximum Administered IP Trunks: 100    96
              Maximum Concurrently Registered IP Stations: 10     10
                Maximum Administered Remote Office Trunks: 0      0
  Maximum Concurrently Registered Remote Office Stations: 0      0
                 Maximum Concurrently Registered IP eCons: 0      0
                         Maximum Administered SIP Trunks: 100    50

     Maximum Number of DS1 Boards with Echo Cancellation: 0      0
                               Maximum TN2501 VAL Boards: 1      0
                          Maximum G700/G350 VAL Sources: 0      0

           (NOTE: You must logoff & login to effect the permission changes.)
```

IP PORT CAPACITIES

Maximum Administered IP Trunks

Defines limits of the number of IP trunks administered.

Maximum Administered SIP Trunks

Defines limits of the number of SIP trunks administered.

Maximum Administered Remote Office Trunks

Defines limits of the number of IP endpoints based on the endpoint. Use the smaller of this number or the number based on the **MAXIMUM IP REGISTRATIONS BY PRODUCT ID** page of this screen.

Maximum Concurrently IP eCons

Specifies the maximum number of IP SoftConsoles that may be registered at one time. The maximum number depends on the type of system.

Maximum Concurrently Registered IP Stations

Specifies the maximum number of IP stations that may be registered at one time. Must be less than or equal to the **Maximum Ports** field on this page. Avaya Communication Manager will use the smaller of this number or the number based on product ID on the **MAXIMUM IP REGISTRATIONS BY PRODUCT ID** page of this screen.

Maximum Concurrently Registered Remote Office Stations

Specifies the maximum number of remote office stations that may be registered at one time. Must be less than or equal to the **Maximum Ports** field on this page. Communication Manager will use the smaller of this number or the number based on product ID on the **MAXIMUM IP REGISTRATIONS BY PRODUCT ID** page of this screen.

Maximum G700/G350 VAL Sources

Specifies the maximum number of G700/G350 VAL announcement sources.

Maximum Number of DS1 Boards with Echo Cancellation

Shows the number of DS1 circuit packs that can have echo cancellation.

Maximum TN2501 VAL Boards

This display-only field indicates the maximum number of TN2501AP (Voice Announcement over LAN) boards allowed in this system.

Valid entries	Usage
0 to **10** (DEFINITY R and S8700 or S8710 Media Servers) **0** to **5** (DEFINITY SI and CSI, and S8300 Media Server)	• For values greater than 1, the **Val Full 1-Hour Capacity** field on page 4 of the **System-Parameters Customer-Options** screen must be set to **y**. • This field updates the **System Limit** field on the **System Capacity** report.

Field descriptions for page 3

Figure 513: System Parameters Customer-Options screen

```
display system-parameters customer-options                              page 3 of x

                                OPTIONAL FEATURES

           Abbreviated Dialing Enhanced List?             Audible Message Waiting?
               Access Security Gateway (ASG)?                 Authorization Codes?
               Analog Trunk Incoming Call ID?                           CAS Branch?
        A/D Grp/Sys List Dialing Start at 01?                             CAS Main?
       Answer Supervision by Call Classifier?                   Change COR by FAC?
                                       ARS?    Computer Telephony Adjunct Links?
                        ARS/AAR Partitioning?         Co-Res DEFINITY LAN Gateway?
                  ARS/AAR Dialing without FAC?     Cvg Of Calls Redirected Off-net?
                  ASAI Link Core Capabilities?                         DCS (Basic)?
                  ASAI Link Plus Capabilities?                   DCS Call Coverage?
              Async. Transfer Mode (ATM) PNC?                  DCS with Rerouting?
Async. Transfer Mode (ATM) Trunking?
               ATM WAN Spare Processor?  Digital Loss Plan Modification?
                                 ATMS?                                 DS1 MSP?
                   Attendant Vectoring?               DS1 Echo Cancellation?
```

Abbreviated Dialing Enhanced List

Provides the capability to store and retrieve dialing lists that simplify or eliminate dialing. You dial an abbreviated code or depress an assigned button. The stored entries are organized in number lists. There are three types of number lists: personal, group, and enhanced.

Access Security Gateway (ASG)

Provides an additional level of security for remote administration.

A/D Grp/Sys List Dialing Start at 01

Allows you to number Abbreviated Dialing group or system lists starting with 01, rather than simply 1. This allows Abbreviated Dialing under Avaya Communication Manager to operate like it did with the DEFINITY G2 system.

Analog Trunk Incoming Call ID

This field allows collection and display the name and number of an incoming call information on analog trunks.

Answer Supervision by Call Classifier

This circuit pack detects tones and voice-frequency signals on the line and determines whether a call has been answered. This field is set to **y** if the system contains a call-classifier circuit pack.

ARS

Provides access to public and private communications networks. Long-distance calls can be routed over the best available and most economical routes. Provides partitioning of ARS routing patterns.

ARS/AAR Partitioning

Provides the ability to partition AAR and ARS into 8 user groups within a single server running Avaya Communication Manager. Can establish individual routing treatment for each group.

ARS/AAR Dialing without FAC

Provides for Automatic Alternate Routing (AAR) and Automatic Route Selection (ARS) calls without dialing a Feature Access Code (FAC).

CAUTION:

Contact your Avaya technical support representative to discuss details of this feature before enabling it.

ASAI Link Core Capabilities

Provides linkage between Avaya Communication Manager and adjuncts. CallVisor ASAI improves the call handling efficiency of ACD agents and other system users by allowing an adjunct to monitor, initiate, control, and terminate calls on the server running Communication Manager.

Note:

ASAI Link Core Capabilities only applies to links administered as type asai. This field was previously named **ASAI Interface**.

If the **ASAI Link Core Capabilities** field is administered to **y** then it will be associated with the following ASAI capability groups:

- Adjunct Control
- Domain Control
- Event Notification
- Single Step Conference

- Request Feature
- II Digits
- Set Value
- Value Query

For more information, see the *Avaya MultiVantage CallVisor ASAI Technical Reference*.

ASAI Link Plus Capabilities

Provides linkage between Avaya Communication Manager and adjuncts. If the **ASAI Link Plus Capabilities** field is administered to **y**, then the following ASAI capability groups are enabled:

- Adjunct Routing
- Answering Machine Detection
- Selective Listening
- Switch Classified Outbound Calls
- ISDN Redirecting Number Information - the original dialed number information is provided within the ASAI messages if it arrives in ISDN SETUP messages from the public networks as either Original Dialed Number or Redirecting Party Number.

Note:

ASAI Link Plus Capabilities only applies to links administered as type asai.

For more information, see the *Avaya MultiVantage CallVisor ASAI Technical Reference*.

Asynch. Transfer Mode (ATM) PNC

ATM PNC can be enabled only if:

- all prior fiber-link administration has been removed
- all "switch-node" and "dup-switch-node" carrier types have been removed

Asynch. Transfer Mode (ATM) Trunking

If ATM trunking is enabled, multiple ISDN-PRI T1 or E1 trunks can be emulated on one ATM pipe. Can only be enabled if the **ISDN-PRI** field is set to **y**. Enables circuit emulation service (CES).

ATM WAN Spare Processor

Appears on DEFINITY R only. An ATM WAN spare processor acts as a PPN in the event of network failure, and can function as an SPE if the main PPN is not functional. Cannot be set to **y** if the **Async. Transfer Mode (ATM) Trunking** field is **n**.

ATMS

Provides for voice and data trunk facilities to be measured for satisfactory transmission performance.

Attendant Vectoring

Allows you to use attendant vectoring. Cannot be set to **y** if the **CAS Main** and **CAS Branch** fields are **y**.

Audible Message Waiting

Provides audible message waiting.

Authorization Codes

Permits you to selectively specify levels of calling privileges that override in-place restrictions. In addition to facilities access, authorization codes are used for unique identification for billing security purposes.

CAS Branch

Provides Centralized Attendant Service - Branch. See CAS Main for more information. Cannot be set to **y** if the **Attendant Vectoring** and **Centralized Attendant** fields are **y**.

CAS Main

Provides multi-location customers served by separate switching vehicles to concentrate attendant positions at a single, main Avaya Communication Manager location. The main Avaya Communication Manager is served by an attendant queue that collects calls from all locations (main and branch). Each branch location switches all of its incoming calls to the centralized attendant positions over release link trunks (RLTs). The calls are then extended back to the requested extension at the branch server/switch over the same RLT. When the call is answered, the trunks to the main server are dropped and can be used for another call. Cannot be set to **y** if the **Centralized Attendant** and **CAS Branch** fields are **y**.

Change COR by FAC

Provides certain users the ability to change the class of restriction of local extensions and local attendants via a telephone by using a feature access code (FAC). Cannot be set to **y** if the **Tenant Partitioning** field is **y**.

Co-Res DEFINITY LAN Gateway

(For S8300 Media Server only.) Provides communications between TCP/IP clients and call processing within Avaya Communication Manager. Cannot be set to **y** if the **ASAI Link Core Capabilities** and **Computer Telephony Adjunct Links** fields are **n**.

For more information see the *Avaya Communication Manager CallVisor ASAI Technical Reference*.

Computer Telephony Adjunct Links

Provides linkage between Avaya Communication Manager and adjuncts. Includes both the ASAI Link Core and ASAI Link Plus capabilities, plus the Phantom Calls and CTI Stations.

Note:

Computer Telephony Adjunct Links only applies to links administered as type **adjlk**. This field was previously named **ASAI Proprietary Adjunct Links**.

For more information see the *Avaya MultiVantage CallVisor ASAI Technical Reference*.

Cvg Of Calls Redirected Off-net

Provides continued monitoring for calls redirected to off-network (remote) coverage points. Uses call classification via call classifier circuit pack or ISDN trunk signaling.

DCS (Basic)

Provides transparent operation of selected features across a Distributed Communications System (DCS). Users on one server running Communication Manager can use features located on another server. Includes 4- and 5-digit uniform dialing and 1 to 4 digit steering. Does not support a 6/7-digit dial plan.

DCS Call Coverage

Provides DCS-based transparency of the call coverage feature across a DCS network of media servers or switches.

DCS with Rerouting

Provides for rerouting calls transferred among DCS nodes, enabling rerouting of the call for more effective use of facilities. Cannot be set to **y** if the **ISDN PRI** field is **n**.

Digital Loss Plan Modification

Allows you to customize the digital loss and digital tone plans.

DS1 MSP

Provides the ability to change fields on **DS1 Circuit Pack** screen without removing the related translations of all trunks from the trunk group.

DS1 Echo Cancellation

Removes perceivable echo from the system.

Field descriptions for page 4

Figure 514: System Parameters Customer-Options screen

```
display system-parameters customer-options                      Page   4 of  x
                              OPTIONAL FEATURES

   Emergency Access to Attendant? y                               IP Stations? y
            Enable ‘dadmin’ Login? y                Internet Protocol (IP) PNC? y
            Enhanced Conferencing? y                            ISDN Feature Plus? y
                   Enhanced EC500? y             ISDN Network Call Redirection? y
       Enterprise Wide Licensing? y                             ISDN-BRI Trunks? y
          Extended Cvg/Fwd Admin? y                                   ISDN-PRI? y
     External Device Alarm Admin? y                Local Survivable Processor? y
          Extended Cvg/Fwd Admin? y                      Malicious Call Trace? y
     External Device Alarm Admin? y     Mode Code for Centralized Voice Mail? y
  Five Port Networks Max per MCC? y
                 Flexible Billing? y                  Multifrequency Signaling? y
   Forced Entry of Account Codes? y Multimedia Appl. Server Interface (MASI)? y
      Global Call Classification? y          Multimedia Call Handling (Basic)? y
             Hospitality (Basic)? y       Multimedia Call Handling (Enhanced)? y
 Hospitality (G3V3 Enhancements)? y
                       IP Trunks? y

           IP Attendant Consoles? y

           (NOTE: You must logoff & login to effect the permission changes.)
```

Emergency Access to Attendant

Provides for emergency calls to be placed to an attendant. These calls can be placed automatically by Avaya Communication Manager or dialed by users.

Enable ‘dadmin’ Login

Provides business partners the ability to install, administer, and maintain Avaya media servers and DEFINITY switches. The dadmin login has access to all the same commands as other logins with the exception of **Go** and **WP**. **Go** is used for **go tcm** and **go debug** as well as **go server**. WP is for writing memory.

Enhanced Conferencing

Enhanced Conferencing allows the customer to use the Meet-me Conference, Selective Conference Party Display, Drop, and Mute, and the No Hold Conference features. Must be **y** to enable the Enhanced Conferencing features.

Enhanced EC500

Provides mobile call services including "Anytime Anywhere" accessibility with One Number availability and Origination mapping.

Extended Cvg/Fwd Admin

Provides basic telecommuting package capability for Extended User Administration of Redirected Calls.

External Device Alarm Admin

Provides for analog line ports to be used for external alarm interfaces. Allows identification of port location, adjunct associated with port location, and the alarm level to report.

Enterprise Wide Licensing

Enterprise Wide Licensing. See your Avaya representative for more information.

Five Port Networks Max Per MCC

Available only for DEFINITY R and S8700 or S8710 Multi-Connect. Allows system administrator to create five port networks in a multi-carrier cabinet. If there are any cabinets with more than 2 PNs assigned, this field cannot be set to **n**.

Flexible Billing

Provides an internationally accepted standard interface for end-to-end digital connectivity. Used with a T1 interface and supports twenty-three 64-KBPS voice or data B-Channels and one 64-Kbps signaling D Channel for total bandwidth of 1.544 Mbps.

Forced Entry of Account Codes

Allows system administration to force account users to enter account codes based on user or trunk class of restriction, or by an option on the Toll Analysis table. FEAC provides an easy method of allocating the costs of specific calls to the correct project, department, etc.

Global Call Classification

Provides call classification outside of North America. Listens for tones and classifies tones detected. Required for Call Coverage Off Net and Outgoing Call Management.

Hospitality (Basic)

Provides access to basic features including: Attendant Crisis Alert, Attendant Room Status, Automatic Wakeup, Custom Selection of VIP DID Numbers, Do Not Disturb, Names Registration, Single-Digit Dialing, and Mixed Station Numbering.

Hospitality (G3V3 Enhancements)

Software required for Property Management System and Automatic Wakeup. Property Management System Interface activates Forward PMS Messages to INTUITY Lodging and PMS Protocol Mode (transmit in ASCII mode). Cannot be set to **y** if the **Hospitality (Basic)** field is **n**.

Usage Note: standard hospitality features are included in basic system software.

Internet Protocol (IP) PNC

This field is only for the Avaya S8700 or S8710 IP-Connect and then should be set to **y**. This field cannot be administered if:

- the **Asynch. Transfer Mode (ATM) PNC** field is **y**
- the **PNC Duplication** field is **y**
- If fiber is administered

IP Attendant Consoles

Controls permission to administer the IP Attendant Console.

IP Stations

Controls permission to administer H.323 and/or SoftPhone stations. Must be **y** for IP phones.

IP Trunks

Controls permission to administer H.323 trunks. Must be **y** for IP trunks.

ISDN Feature Plus

Provides ISDN Feature Plus signaling. This option is enabled when either the **ISDN-BRI Trunks** field or the **ISDN-PRI** field is **y**.

ISDN Network Call Redirection

Administrable if the **ISDN-PRI** or **ISDN-BRI Trunk** field is **y**. Network Call Redirection (NCR) redirects an incoming ISDN call from a server running Avaya Communication Manager to another PSTN endpoint. It is used in Call Centers with Best Service Routing and Lookahead Interflow. For more information, see *Avaya MultiVantage Call Center Software Release 11 — Network Call Redirection*.

ISDN-BRI Trunks

Provides the capability to add ISDN-BRI trunks to Communication Manager. If enabled, can add isdn trunk groups and the following screens are accessible:

- **network-facilities**
- **private-numbering**
- **public-unknown- numbering**

ISDN-PRI

Provides Integrated Services Digital Network (ISDN-PRI) software for either a switching-hardware platform migration only or a switching-hardware platform migration in combination with a software release upgrade. Also provides signaling support for H.323 signaling. Must be **y** for IP trunks.

Local Survivable Processor

A display-only field that indicates that the server is a Local Survivable Processor (LSP) Enterprise Survivable Server (ESS). If **y**, the S8300 Media Server is configured to provide standby call processing in case the primary media server is unavailable.

Malicious Call Trace

Provides ability to retrieve certain information related to a malicious call.

Mode Code for Centralized Voice Mail

This feature provides the ability to share a Voice Mail System (VMS) among several servers/ switches using the Mode Code - Voice Mail System Interface.

Multifrequency Signaling

Provides for a screen of number signaling used between Communication Manager and the central office.

Multimedia Appl. Server Interface (MASI)

Allows users of the Multimedia Communications Exchange (MMCX) to take advantage of certain Avaya Communication Manager telephony features.

Multimedia Call Handling (Basic)

Allows administration of desktop video-conferencing systems as data modules associated with Avaya Communication Manager voice stations in a multimedia complex. Users can dial one number to reach either endpoint (voice or data) in the complex. Also provides support for IP SoftPhones.

Multimedia Call Handling (Enhanced)

Allows a multifunction telephone to control a multimedia call like a standard voice call.

Field descriptions for page 5

Figure 515: System Parameters Customer-Options screen

```
display system-parameters customer-options                          page 5 of x

                                OPTIONAL FEATURES

                 Multinational Locations?            Station and Trunk MSP? n
Multiple Level Precedence and Preemption?     Station as Virtual Extension? n
                       Multiple Locations?      Survivable Remote Processor? n
                                               System Management Data Transfer? n

         Personal Station Access (PSA)? y
                       Posted Messages? n              Tenant Partitioning? n
                       PNC Duplication? n      Terminal Trans. Init. (TTI)? y
                  Port Network Support? y              Time of Day Routing? y
             Processor and System MSP? n             Uniform Dialing Plan? y
                    Private Networking? y    Usage Allocation Enhancements? y
                    Processor Ethernet? y      TN2501 VAL Maximum Capacity? y

                         Remote Office? n               Wideband Switching? y
         Restrict Call Forward Off Net? y                         Wireless? n
                 Secondary Data Module? y
```

Multinational Locations

The Multinational Locations feature provides you with the ability to use a single Enterprise Communication Server (ECS) with stations, port networks, remote offices, or gateways in multiple countries..With this feature enabled, you can administer location parameters such as companding, loss plans, and tone generation per location, instead of system-wide.

Multiple Level Precedence and Preemption

Multiple Level Precedence and Preemption (MLPP) provides users the ability to assign levels of importance to callers, and when activated, to give higher-priority routing to individual calls based on the level assigned to the caller.

Multiple Locations

Allows you to establish numbering plans and time zone and daylight savings plans that are specific for each cabinet in a port network.

Personal Station Access (PSA)

Provides basic telecommuting package capability for Personal Station Access.

Posted Messages

Supports users being able to post messages, which they select from among a set of as many as 30 (15 fixed, 15 administrable), to be shown on display phones.

PNC Duplication

This field only appears on Avaya DEFINITY R. If set to **y**, the **Enable Operation of PNC (Port Network Connectivity) Duplication** field appears on the **System Parameters Duplication** screen. The **Enable Operation of PNC Duplication** field is set with the **Enable Operation of SPE (Switch Processing Element) Duplication** field to provide non-standard reliability levels (high, critical, or ATM PNC Network Duplication).

Port Network Support

Indicates that the server is operating as a stand-alone Internal Communications Controller (ICC) when set to **n** and is used to disable traditional port networking. Set to **y** to indicate that traditional Avaya DEFINITY port networks are in use.

Private Networking

Upgrades PNA or ETN software RTU purchased with earlier systems.

Processor Ethernet

Appears only on S8700 or S8710 IP-Connect, S8700 or S8710 Multi-Connect, and S8300 Media Server. Used to enable use of the ethernet card resident in the processor cabinet for use by the DEFINITY Call Processing software in place of a C-LAN card (located in a port network).

Processor and System MSP

Allows the customer administrator or technician to maintain processor and system circuit packs.

Remote Office

Allows administration of a remote office.

Restrict Call Forward Off Net

The system can monitor the disposition of an off-call and, if it detects busy, bring the call back for further processing, including call coverage.

Secondary Data Module

Provides ability to use any data module as a secondary data module.

Station and Trunk MSP

Provides the customer administrator or technician to maintain station and trunk circuit packs.

Station as Virtual Extension

Allows **virtual** to be entered in the **Type** field of the **Station** screen, which allows multiple virtual extensions to be mapped to a single physical analog telephone. The user can also administer a specific ringing pattern for each virtual extension. Useful in environments such as college dormitories, where three occupants can have three different extensions for one physical telephone.

Survivable Remote Processor

This field can be set to **y** only if, on the **System-Parameters Maintenance** screen, the **CPE Alarm Activation Level** field is **none**. This applies only to DEFINITY SI systems.

System Management Data Transfer

Indicates Communication Manager is accessible by Network Administration.

Tenant Partitioning

Provides for partitioning of attendant groups and/or stations and trunk groups. Typically this is used for multiple tenants in a building or multiple departments within a company or organization.

Terminal Trans. Init. (TTI)

Allows administrators of Terminal Translation Initialization (TTI) to merge an station administered with **X** in the **Port** field, to a valid port by dialing a system-wide TTI security code and the extension from a terminal connected to that port. Must be set to **y** for Automatic Customer Telephone Rearrangement.

Time of Day Routing

Provides AAR and ARS routing of calls based on the time of day and day of the week. You can take advantage of lower calling rates during specific times.

TN2501 VAL Maximum Capacity

If this is enabled, you have the Enhanced offer, which allows up to 60 minutes storage capacity per pack and multiple integrated announcement circuit packs.

Uniform Dialing Plan

Provides 3- to 7-digit Uniform Dial Plan (UDP) and 1 to 7 digit steering. Also allows you to use Extended Trunk Access and Extension Number Portability features.

Usage Allocation Enhancements

Provides for assigning ISDN-PRI or ISDN-BRI Services/Features for Usage Allocation Plans. To use this enhancement, first set either the **ISDN-PRI** or **ISDN-BRI Trunks** fields to **y**.

Wideband Switching

Provides wideband data software for switching video or high-speed data. You can aggregate DSO channels up to the capacity of the span. Wideband supports H0, H11, and H12 standards, where applicable, as well as customer-defined data rates.

Wireless

Provides right to use for wireless applications in certain Network Systems sales. You may purchase it from Avaya Network Wireless Systems.

Field descriptions for Call Center Optional Features

Figure 516: Call Center Optional Features screen

```
  display system-parameters customer-options                          page 6 of x

                        CALL CENTER OPTIONAL FEATURES

                          Call Center Release:

                                  ACD? y     PASTE (Display PBX Data on Phone)? n
                        BCMS (Basic)? y                          Reason Codes? n
                                                       Service Level Maximizer? y
          BCMS/VuStats Service Level? n            Service Observing (Basic)? y
                    Business Advocate? n     Service Observing (Remote/By FAC)? n
                      Call Work Codes? y              Service Observing (VDNs)?
       DTMF Feedback Signals For VRU? y                           Timed ACW?
                    Dynamic Advocate? n                   Vectoring (Basic)? y
        Expert Agent Selection (EAS)? y                Vectoring (Prompting)? y
                             EAS-PHD? n              Vectoring (G3V4 Enhanced)?
                    Forced ACD Calls? n     Vectoring (ANI/II-Digits Routing)? n
                Least Occupied Agent?       Vectoring (G3V4 Advanced Routing)? n
            Lookahead Interflow (LAI)?                    Vectoring (CINFO)? n
  Multiple Call Handling (On Request)? n      Vectoring (Best Service Routing)? n
      Multiple Call Handling (Forced)? n                 Vectoring (Holidays)?
```

ACD

Automatic Call Distribution (ACD) automatically distributes incoming calls to specified splits or skills. Provides the software required for the Call Center Basic, Plus, Deluxe, and Elite features for the number of agents specified. Cannot be set to **n** if the **Call Work Codes** field is **y**.

BCMS (Basic)

Provides real-time and historical reports about agent, ACD split, Vector Directory Number (VDN) and trunk group activity.

BCMS/VuStats Service Level

Allows you to set up hunt groups or Vector Directory Numbers (VDNs) with an acceptable service level. An acceptable service level defines the number of seconds within which a call must be answered to be considered acceptable.

Business Advocate

Software that provides an integrated set of advanced features to optimize call center performance. If set to **n**, the **Least Occupied Agent** field displays. For information on CentreVu Advocate, contact your Account Executive.

Call Center Release

Displays the call center release installed on the system.

Call Work Codes

Allows agents to enter digits for an ACD call to record customer-defined events such as account codes or social security numbers. Cannot be set to **y** if the **ACD** field is **n**.

DTMF Feedback Signals For VRU

Provides support for the use of C and D Tones to VRUs.

Dynamic Advocate

Software that provides an integrated set of advanced features to optimize call center performance.

EAS-PHD

Increases the number of skills an agent can log in to from four to 20. Increases the number of agent skill preference levels from two to 16.

Expert Agent Selection (EAS)

Provides skills-based routing of calls to the best-qualified agent.

Forced ACD Calls

See Multiple Call Handling.

Least Occupied Agent

Appears only if the **Business Advocate** field is **n**. Allows call center calls to be routed to the agent who has been the least busy, regardless of when the agent last answered a call. Cannot be set to **y** if the **Expert Agent Selection (EAS)** field is **n**.

Lookahead Interflow (LAI)

Provides Look-Ahead Interflow to balance the load of ACD calls across multiple locations. Cannot be set to **y** if the **Vectoring (Basic)** field is **n**.

Multiple Call Handling (On Request)

Allows agents to request additional calls when active on a call.

Multiple Call Handling (Forced)

Forces an agent to be interrupted with an additional ACD call while active on an ACD call. Splits or skills can be one forced, one per skill, or many forced. Cannot be set to **y** if the **ACD** field is **n** and the **Forced ACD Calls** field is **y**.

PASTE (Display PBX Data on Phone)

Provides an interface between the display of a DCP telephone set and PC-based applications.

Reason Codes

Allows agents to enter a numeric code that describes their reason for entering the AUX work state or for logging out of the system. Cannot be set to **y** if the **Expert Agent Selection (EAS)** field is **n**.

Service Level Maximizer

Allows an administrator to define a service level whereby X% of calls are answered in Y seconds. When Service Level Maximizer (SLM) is active, the software verifies that inbound calls are matched with agents in a way that ensures that the administered service level is met. SLM is used with Expert Agent Selection (EAS), and without Business Advocate. Call Center Release must be 12 or later.

Service Observing (Basic)

Allows a specified user to observe an in-progress call on a listen-only or listen-and-talk basis.

Service Observing (Remote/By FAC)

Allows users to service observe calls from a remote location or a local station using this feature’s access codes.

Service Observing (VDNs)

Provides the option of observing and/or monitoring another user's calls.

Timed ACW

Places an auto-in agent in ACW for an administered length of time after completion of the currently active ACD call.

Vectoring (Basic)

Provides basic call vectoring capability.

Vectoring (Prompting)

Allows flexible handling of incoming calls based on information collected from the calling party or from an ISDN-PRI message.

Vectoring (G3V4 Enhanced)

Allows the use of enhanced comparators, wildcards in digit strings for matching on collected digits and ANI or II-digits, use of Vector Routing Tables, multiple audio/music sources for use with wait-time command and priority level with the oldest-call-wait conditional.

Vectoring (ANI/II-Digits Routing)

Provides for ANI and II-Digits vector routing.

Vectoring (G3V4 Advanced Routing)

Provides for Rolling Average Speed of Answer Routing, Expected Wait Time Routing, and VDN Calls Routing.

Vectoring (CINFO)

Provides the ability to collect ced and cdpd from the network for vector routing. To use this enhancement, first set either the **ISDN-PRI** or **ISDN-BRI Trunks** fields to **y**.

Vectoring (Best Service Routing)

Enables the Best Service Routing feature. Through special vector commands, Best Service Routing allows you to compare splits or skills at local and remote locations and queue a call to the resource that will give the caller the best service.

Vectoring (Holidays)

Enables the Holiday Vectoring feature. It simplifies vector writing for holidays.

Field descriptions for Call Center Optional Features

Figure 517: Call Center Optional Features screen

```
display system-parameters customer-options                    Page 7 of x
                      CALL CENTER OPTIONAL FEATURES

     VDN of Origin Announcement? n                                VuStats? n
         VDN Return Destination? n              VuStats (G3V4 Enhanced)? n









                                   Used
           Logged-In ACD Agents: 500
      Logged-In Advocate Agents: 500
  Logged-In IP Softphone Agents: 500
```

Logged-In ACD Agents

Number of ACD Agents contracted for. This field limits the number of logged-in ACD agents to a number no more than the maximum purchased. The value of this field indicates the total of ACD agents that can be logged-in simultaneously.

The limit applies to ACD agents on ACD and EAS calls. Auto-Available Split (AAS) agent ports are counted when they are assigned. AAS split or skill members are also counted. If the port for an AAS split/skill member is logged out, (for example, when a ringing call is redirected) the logged-in agent count is not updated. These counts are updated only during administration.

Logged-In Advocate Agents

Appears when the **CentreVu Advocate** field is **y**. Number of CentreVu Advocate Agents contracted for.

The total number of logged-in CentreVu Advocate agents must be equal to or less than the number allowed in the **Logged-In ACD Agents** field. The number of logged-in CentreVu Advocate agents counts towards the total number of logged-in ACD agents.

Logged-In IP Softphone Agents

Number of IP Softphone Agents contracted for. This field limits the number of logged-in IP Softphone agents to a number no more than the maximum purchased. The value of this field indicates the total of IP Softphone agents that can be logged-in simultaneously.

VDN of Origin Announcement

Provides a short voice message to an agent indicating the city of origin of the caller or the service requested by the caller based on the VDN used to process the call.

VDN Return Destination

Allows an incoming trunk call to be placed back in vector processing after all parties, except the originator, drop.

VuStats

Allows you to present BCMS statistics on telephone displays.

VuStats (G3V4 Enhanced)

Allows you to use the G3V4 VuStats enhancements including historical data and thresholds.

Field descriptions for QSIG Optional Features

Figure 518: QSIG Optional Features screen

```
display system-parameters customer-options                      Page 8 of x
                            QSIG OPTIONAL FEATURES

                                        Basic Call Setup? n
                             Basic Supplementary Services? n
                                    Centralized Attendant? n
                                    Interworking with DCS? n
                      Supplementary Services with Rerouting? n
                                Transfer into QSIG Voice Mail? n
                                          Value-Added (VALU)? n
```

Basic Call Setup

Provides basic QSIG services: basic connectivity and calling line ID number. To use this enhancement, either the **ISDN-PRI** or **ISDN-BRI Trunks** fields must be **y**.

Basic Supplementary Services

To use this enhancement, either the **ISDN-PRI** or **ISDN-BRI Trunks** fields must be **y**. Provides the following QSIG Supplementary Services:

- Name ID
- Transit Capabilities; that is, the ability to tandem QSIG information elements
- Support of Notification Information Elements for interworking between QSIG and non-QSIG tandemed connections
- Call Forwarding (Diversion) by forward switching. No reroute capabilities are provided
- Call Transfer by join. No path replacement capabilities are provided.
- Call Completion (also known as Automatic Callback)

Centralized Attendant

Can be enabled only if the **Supplementary Services with Rerouting** field is **y**. Cannot be set to **y** if the **CAS Main** and **CAS Branch** fields are **y**. Allows all attendants in one location to serve users in multi locations. All signaling is done over QSIG ISDN lines. If this field is **y**, the **IAS** fields on the **Console Parameters** screen do not display.

Interworking with DCS

Allows the following features to work between a user on a DCS-enabled media server or switch in a network and a QSIG-enabled media server or switch:

- Calling/Called/Busy/Connected Name
- Voice Mail/Message Waiting
- Leave Word Calling

This field cannot be set to **y** if the **DCS (Basic)** field is **n**.

Supplementary Services with Rerouting

Provides the following QSIG Supplementary Services:

- Transit Capabilities; that is, the ability to tandem QSIG information elements.
- Support of Notification Information Elements for interworking between QSIG and non-QSIG tandemed connections.
- Call Forwarding (Diversion) by forward switching. In addition, reroute capabilities are provided.
- Call Transfer by join. In addition, path replacement capabilities are provided.

Transfer Into QSIG Voice Mail

Can be enabled only if the **Basic Supplementary Services** field is **y** and either the **ISDN-PRI Trunk** or **ISDN-BRI Trunk** field is **y**. Allows transfer directly into the voice-mail box on the voice-mail system when a QSIG link connects Avaya Communication Manager and the voice-mail system.

Value Added (VALU)

Provides additional QSIG functionality, including the ability to send and display calling party information during call alerting. See *Administration for Network Connectivity for Avaya Communication Manager* for more information.

Field descriptions for ASAI

Figure 519: ASAI Features screen when the ASAI Link Plus Capabilities field is y

```
change system-parameters customer options                    Page 8 of X

                                  ASAI FEATURES

                                CTI Stations? n
                               Phantom Calls? n



                           ASAI PROPRIETARY FEATURES

                                Agent States? n
```

Agent States

Appears when the **Computer Telephony Adjunct Links** field is **y**. The **Agent States** field provides proprietary information used by Avaya applications. For more information, contact your Avaya technical support representative.

Note:

The **Agent States** field only applies to links administered as type adjlk. This field was previously named **Proprietary Applications**.

CTI Stations

Appears when the **ASAI Link Plus Capabilities** field is **y**. This field needs to be enabled for any application (using a link of Type ASAI) that uses a CTI station to receive calls.

For more information see the *Avaya MultiVantage CallVisor ASAI Technical Reference*.

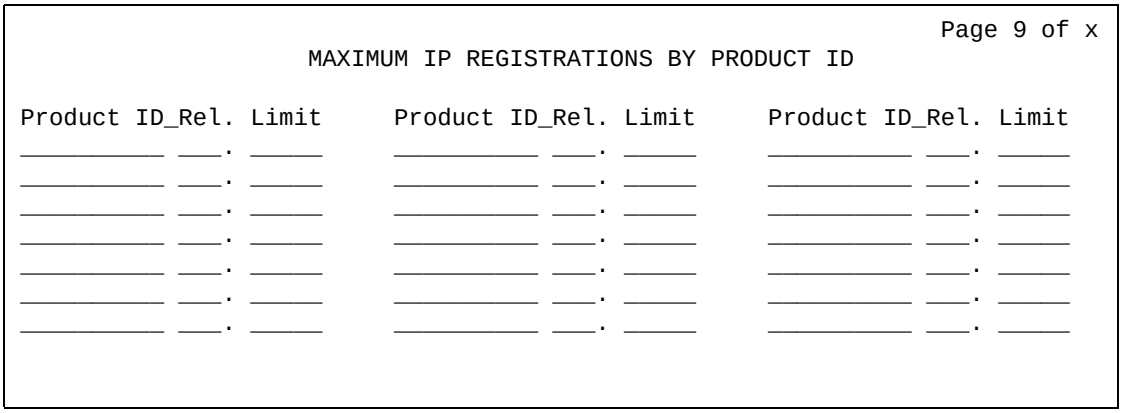
Phantom Calls

Appears when the **ASAI Link Plus Capabilities** field is **y**. Enables phantom calls. The **Phantom Calls** field only applies to links administered as type ASAI.

For more information see the *Avaya MultiVantage CallVisor ASAI Technical Reference*.

Field descriptions for Maximum IP Registrations by Product ID

Figure 520: Maximum IP Registrations by Product ID screen



```
                                                              Page 9 of x
                    MAXIMUM IP REGISTRATIONS BY PRODUCT ID

Product ID_Rel. Limit     Product ID_Rel. Limit     Product ID_Rel. Limit
__________ ___. _____    __________ ___. _____    __________ ___. _____
__________ ___. _____    __________ ___. _____    __________ ___. _____
__________ ___. _____    __________ ___. _____    __________ ___. _____
__________ ___. _____    __________ ___. _____    __________ ___. _____
__________ ___. _____    __________ ___. _____    __________ ___. _____
__________ ___. _____    __________ ___. _____    __________ ___. _____
__________ ___. _____    __________ ___. _____    __________ ___. _____
```

Limit

Maximum number of IP registrations allowed.

Valid displays	Usage
1000 or **5000**, depending on your server configuration	Maximum number of IP registrations allowed. For Avaya R300 Remote Office Communicator, defaults to the maximum allowed value for the **Concurrently Registered Remote Office Stations** on page 1 of this screen.

Product ID

Identifies the product using the IP (internet protocol) registration.

Valid displays	Usage
Avaya_IR	Interactive Response product
IP_Agent	IP Agents
IP_eCons	SoftConsole IP attendant
IP_Phone	IP Telephones
IP_ROMax	R300 Remote Office phones
IP_Soft	IP Softphones

Rel

Release number of the IP endpoint.

Valid displays	Usage
0 to **99** or blank	Release number of the IP endpoint

System Parameters OCM Call Classification

This screen enters the tone characteristics for your country for Outbound Call Management (OCM) applications. It is not required for United States OCM applications. If you cannot access this screen, contact your Avaya technical support representative.

This screen appears when **Global Call Classification** field on the **System-Parameters Customer-Options** screen is set to **y**, or when the **Enable Busy Tone Disconnect for Analog loop-start Trunks** field on the **System Parameters Country Options** screen is set to **y**. This screen defines the busy tone and cadence and can be administered with up to 4 on and off steps, which is four valid cycles to determine busy tone.

Avaya recommends that you use a minimum of two on and off steps to determine a valid busy tone. If the cadence is administered with one on and off step, any time the classifier hears the cadence it is considered BTD signal.

Field descriptions for page 1

Figure 521: System Parameters OCM Call Classification screen

```
                                                              Page 1 of 2
                  SYSTEM PARAMETERS OCM CALL CLASSIFICATION


TONE DETECTION PARAMETERS
  Global Classifier Adjustment (dB): ___
              USA Default Algorithm? _n_
                  USA SIT Algorithm? ___
```

Global Classifier Adjustment (dB)

Enter a number to specify the dB loss adjustment.

Valid entries	Usage
0 to **15**	**0** is the least and **15** the most adjustment.

USA Default Algorithm

Valid entries	Usage
y/n	To use the default United States tone detection, set this field to **y**. If you enter **n**, the **US Special Information Tones (SIT) Algorithm** field appears.

USA SIT Algorithm

Valid entries	Usage
y	To use the United States (SIT) tone characteristics for SIT tone detection.
n	The system treats tones with the administered tone name "intercept" as if they were SIT VACANT, and treats tones with the administered tone name "information" as if they were SIT UNKNOWN.

Field descriptions for page 2

Figure 522: System Parameters OCM Call Classification screen

```
                                                          Page 2 of x
                SYSTEM PARAMETERS OCM CALL CLASSIFICATION

Tone Name  Instance      Tone       Cadence   Duration     Duration
                      Continuous   Step      Minimum      Maximum
__________   _____         ___     1.  on     _____        _____
                                   2.  off    _____        _____
                                   3.  on     _____        _____
                                   4.  off    _____        _____
                                   5.  on     _____        _____
                                   6.  off    _____        _____
                                   7.  on     _____        _____
                                   8.  off    _____        _____
```

Cadence Step

A display-only field identifying the number of each tone cadence step and indicating whether the tone is on or off during this cadence step.

Duration Maximum

Specifies the upper limit in milliseconds of the tone duration.

Note:

On the **Feature-Related System Parameters** screen, set the **Off-Premises Tone Detect Timeout Interval** field to its maximum value.

Valid entries	Usage
75 to **6375**	Enter in increments of 25 msec.

Duration Minimum

Specifies the lower limit in milliseconds (msec) of the tone duration.

Valid entries	Usage
75 to **6375**	Enter in increments of 25 msec.

Instance

Enter the instance number of the tone. If the system identifies a tone that matches the characteristics defined on more than one page of this screen the system applies the tone definition from the earlier page.

Valid entries	Usage
1 to **8**	The number distinguishes tones that have the same name but more than one definition of silence and tone-on characteristics.

Tone Continuous

Valid entries	Usage
y	Indicates a continuous tone. If you enter **y**, you cannot enter data in the **Duration Minimum** or **Duration Maximum** fields.
n	Indicates a non-continuous tone.

Tone Name

This field is required for tone definition outside of the U.S. and Canada.

If the **Global Call Classification** field on the **System-Parameters Customer-Options** screen is **n**, only busy can be entered into this **Tone Name** field. If **Busy Tone Disconnect** is enabled, only busy can be entered into this field.

Valid entries	Usage
busy **information** **intercept** **reorder** **ringback**	Enter the name of the tone that you are adding or modifying. Enter **busy** for **Busy Tone Disconnect**.

System Parameters - SCCAN

Field descriptions for page 1

Figure 523: SCCAN-Related System Parameters screen

```
change system-parameters sccan
                      SCCAN - RELATED SYSTEM PARAMETERS

                 MM(WSM) Route Pattern: _________
                           H1 Handover: _________
                           H2 Handover: _________
                          Announcement: _________
              Special Digit Conversion? _________
```

MM (WSM) Route Pattern

Enter a route pattern number that is SCCAN-enabled. Partition route pattern indexes, RHNPA indexes, deny, or nodes are not allowed.

Valid entries	Usage
blank	Default value. If this field is left blank, the feature is turned off. To enable this feature, you must enter an acceptable value.
digits	Right-click on the field on the SAT screen to see valid entries for your system.

H1 Handover

Valid entries	Usage
unassigned extension	The primary handover number called to facilitate handover of a cellular call to the WAN or WLAN. Depending on whether the user is entering or exiting the Enterprise space, Communication Manager replaces the active call with the new call made using the hand-off H1 or H2 number.

H2 Handover

Valid entries	Usage
unassigned extension	A secondary handover number used when no acknowledgement is received from the H1 Handover number.

Announcement

Valid entries	Usage
assigned announcement extension	Enter the extension of the announcement you want to play during call handin or handout.

Special Digit Conversion

This field allows a user to call a cellular telephone number and get the same treatment as calling an extension that is running Communication Manager.

Valid entries	Usage
y	ARS checks the dialed string to determine if the dialed string is a SCCAN telephone number. If the number is a SCCAN telephone number, the cellular telephone number is replaced with the extension number that the cellular telephone is mapped to.
n	Default value. The feature is turned off.

Telecommuting Access

This screen allows the System Administrator to administer the extension which allows remote users to use the feature.

Field descriptions for page 1

Figure 524: Telecommuting Access screen

```
add telecommuting-access
                          TELECOMMUTING ACCESS

                Telecommuting Access Extension: ________
```

Telecommuting Access Extension

This only allows remote access to the Telecommuting Access feature.

Valid entries	Usage
Unassigned extension of 1-7 digits or blank	Enter an extension that conforms to your system's dial plan and is not assigned to any other system object.

Related Topics

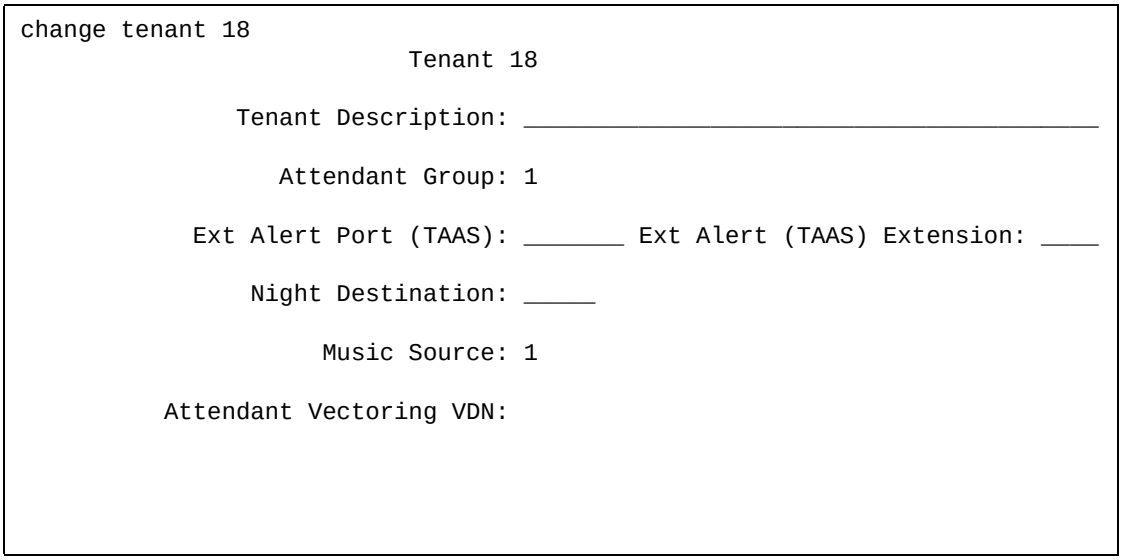
See Configuring Avaya Communication Manager for Telecommuting on page 367 for information about setting up telecommuting.

Tenant

This screen defines tenants to the system. If your server running Communication Manager uses tenant partitioning, see "Tenant Partitioning" in *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205, for more information.

Field descriptions for page 1

Figure 525: Tenant screen



Attendant Group

This required information relates a tenant to an attendant group.

Note:

The default for the system is that all attendant groups exist. However, the attendant group will be empty if no consoles are assigned to it.

Valid entries	Usage
1 to **28**	See *Hardware Guide for Avaya Communication Manager* for your system's range of allowable attendant group numbers.

Attendant Vectoring VDN

This field appears only if, on the **System-Parameters Customer-Options** screen, the **Attendant Vectoring** field is **y** and the **Tenant Partitioning** field is **n**. Enter the assigned Attendant VDN extension or blank. When set to **y**, the **VDN** and **Vector** screens display.

Ext Alert Port (TAAS)

Enter the necessary characters. Enter Trunk Answer Any Station (**TAAS**) alert port information, if any. The port type and the object type must be consistent, and the port can be assigned to only one tenant.

Valid entries	Usage
A valid port address or X **01** to **44** (DEFINITY R) or **01** to **03** (DEFINITY CSI, SI) or **1** to **64** (S8700 or S8710 IP-Connect / S8700 or S8710 Multi-Connect / S8300 Media Server)	First and second characters are the cabinet number
A to **E**	Third character is the carrier
0 to **20**	Fourth and fifth character are the slot number
01 to **04** (Analog TIE trunks) **01** to **31** **1** to **80** (DEFINITY R, CSI) or **1** to **10** (DEFINITY SI) or **1** to **250** (S8700 or S8710 IP-Connect / S8700 or S8710 Multi-Connect / S8300 Media Server)	Six and seventh characters are the circuit number Gateway
V1 to **V9**	Module
01 to **31**	Circuit

Ext Alert (TAAS) Extension

This field appears only if you entered an **x** in the **Ext Alert Port (TAAS)** field. A system installer can then use the Terminal Translation Initialization (TTI) feature from a telephone plugged into any port to assign this extension number to that port. Doing so makes that port the external alert TAAS port.

Valid entries	Usage
A valid extension	Assign an extension as the external alert TAAS extension.

Music Source

Valid entries	Usage
1 to **20**	Enter the music/tone source for this partition. These sources are defined on the **Music Sources** screen.

Night Destination

Valid entries	Usage
A valid extension	Enter the night service station extension, if you want night service for this tenant.

Tenant

This is a display only field. It contains the tenant number that you entered on the command line.

Tenant Description

Valid entries	Usage
40 alpha-numeric characters or blank	You may leave the description field blank, but future administration will be easier if you provide descriptive information.

Field descriptions for page 2

Figure 526: Tenant screen

```
change tenant 18
                                  Tenant 18


CALLING PERMISSION (Enter y to grant permission to call specified Tenant)


 1? y    11? n    21? n    31? n    41? n    51? n    61? n    71? n    81? n    91? n
 2? n    12? n    22? n    32? n    42? n    52? n    62? n    72? n    82? n    92? n
 3? n    13? n    23? n    33? n    43? n    53? n    63? n    73? n    83? n    93? n
 4? n    14? n    24? n    34? n    44? n    54? n    64? n    74? n    84? n    94? n
 5? n    15? n    25? n    35? n    45? n    55? n    65? n    75? n    85? n    95? n
 6? n    16? n    26? n    36? n    46? n    56? n    66? n    76? n    86? n    96? n
 7? n    17? n    27? n    37? n    47? n    57? n    67? n    77? n    87? n    97? n
 8? n    18? y    28? n    38? n    48? n    58? n    68? n    78? n    88? n    98? n
 9? n    19? n    29? n    39? n    49? n    59? n    69? n    79? n    89? n    99? n
10? n    20? n    30? n    40? n    50? n    60? n    70? n    80? n    90? n   100? n
```

Calling permissions

The system default allows each tenant to call only itself and Tenant 1. If you want to change that, you can do that on this screen.

Valid entries	Usage
y/n	Enter **y** to establish calling permission between the tenant number that you entered on the command line and any other tenant.

Tenant

This is a display only field. It contains the tenant number that you entered on the command line.

Terminal Parameters

This screen administers system-level parameters and audio levels for the 603 CALLMASTER telephones and the 4600-series, 6400-series, 8403, 8405B, 8405B+, 8405D, 8405D+, 8410B, 8410D, 8411B, 8411D, 8434D, and 2420/2410 phones. Only authorized Avaya personnel can administer this screen.

Note:

With the Multinational Locations feature enabled, you can administer terminal parameters per location, rather than system-wide.

Figure 527: 603/302 Terminal Parameters screen

```
hange terminal-parameters                                    Page 1 of 3
                     302/603/606-TYPE TERMINAL PARAMETERS

          Base Parameter Set: 1             Customize Parameters? _
     Note: Location-parameters forms assign terminal parameter sets.

   OPTIONS*
                Display Mode: _*               DLI Voltage Level: _________*


   PRIMARY LEVELS*
            Voice Transmit (dB): _____*          Voice Sidetone (dB): _____*
             Voice Receive (dB): _____*     Touch Tone Sidetone (dB): _____*
       Touch Tone Transmit (dB): _____*
```

Figure 528: 6400/607A1/4600/2420 Type Terminal Parameters screen

```
change terminal-parameters                                    Page 2 of 3

                6400/607A1/4600/2420-TYPE TERMINAL PARAMETERS

       Base Parameter Set: 1         Customize Parameters? y
 Note: Location-parameters forms assign terminal parameter sets.
 Note: LEVELS do not apply to the 4600 terminals.*

OPTIONS*
                Display Mode:_*       Handset Expander Enabled?
       Volume for DCP Types:_*
        Volume for IP Types:_*
PRIMARY LEVELS*
         Voice Transmit (dB):__*          Voice Sidetone (dB):__*
          Voice Receive (dB):__*      Touch Tone Sidetone (dB):__*
    Touch Tone Transmit (dB):__*


BUILT-IN SPEAKER LEVELS*
         Voice Transmit (dB):__*           Voice Receive (dB):__*
                                     Touch Tone Sidetone (dB):__*
6402 BUILT-IN SPEAKER LEVELS*
          Voice Receive (dB):__*      Touch Tone Sidetone (dB):__*
```

Figure 529: 8400-Series Terminal Parameters screen

```
change terminal-parameters                                    Page 3 of 3
                          8400-TYPE TERMINAL PARAMETERS

          Base Parameter Set: __          Customize Parameters? _
     Note: Location-parameters forms assign terminal parameter sets.

OPTIONS*
                  Display Mode: _*               DLI Voltage Level: _________*

   PRIMARY LEVELS*
            Voice Transmit (dB): _____*          Voice Sidetone (dB): _____*
             Voice Receive (dB): _____*     Touch Tone Sidetone (dB): _____*
        Touch Tone Transmit (dB): _____*
   ADJUNCT LEVELS*+
            Voice Transmit (dB): _____*          Voice Receive (dB): _____*
            Voice Sidetone (dB): _____*    Touch Tone Sidetone (dB): _____*
   BUILT-IN SPEAKER LEVELS
            Voice Transmit (dB): _____*          Voice Receive (dB): _____*
                                           Touch Tone Sidetone (dB): _____*

   8403 BUILT-IN SPEAKER LEVELS
             Voice Receive (dB): _____*    Touch Tone Sidetone (dB): _____*
```

Base Parameter Set

Determines which default set of telephone options and levels will be used. This field corresponds to the country codes. For the country code listing, see the Country code table on page 1465.

Customize Parameters

Indicates whether the administrator wishes to change one or more of the default parameters.

Note:

Beginning with the May 2004 2.1 Release of Communication Manager, when the **Customize Parameters** field on the **Terminal Parameters n** screen is set to **y**, all Base Parameter Set default values display in the parameter fields. You must change the values in fields for which the default is not desired. To change just a few parameters back to default values, temporarily set the **Customize Parameters** field on the **Terminal Parameters n** screen to **n**, but do not submit the screen (do not press **Enter**). Make note of the default values for the specific fields you want to change, then set the Customize Parameters field back to **y**, and enter the default values in the fields. .

Valid entries	Usage
y	If this field is **y** (yes), the **Option** and **Level** fields appear and all fields can be edited.
n	If this field is **n** (no), the system uses all default parameters associated with the **Base Parameter Set** field and all fields are display-only.

OPTIONS

Display Mode

Determines how the #) and ~ characters appear on the telephone's display.

Valid entries	Usage
1	If this field is set to **1**, the # and ~ do not change.
2	If this field is set to **2**, the telephone displays a # as a British pound sterling symbol and a ~ as a straight overbar.

DLI Voltage Level

Determines whether DCP Line Voltage used by the telephones is forced high, forced low, or allowed to automatically adjust.

Handset Expander Enabled

Determines whether the telephone will reduce noise on the handset.

Valid entries	Usage
y	If the field is **y**, the telephone reduces background noise.

Primary levels

The system displays the default setting from the Base Parameter Set for all fields. Also, these fields all require the same input; valid entries are from **-44.0** db through **+14.0** db in 0.5 increments (for example, -44.0, -43.5, -43.0 and son on).

Volume for DCP Types

This field allows the DCP telephone volume to be adjusted while the call is in progress.

Valid entries	Usage
default speaker, handset unchangeable	The speaker resets to the default settings while the adjusted handset setting is retained.
default settings used to begin each call	No adjusted handset and speaker settings are retained.
retain handset and speaker between calls	The adjusted handset and speaker settings are retained.
retain speaker, handset unchangeable	Only the adjusted speaker setting is retained.

Volume for IP Types

This field allows the IP telephone volume to be adjusted while the call is in progress.

Note:

If you use this field, Avaya recommends that you not change any values in the **PRIMARY LEVELS** or **BUILT-IN SPEAKER LEVELS** areas.

Valid entries	Usage
default speaker, handset unchangeable	The speaker resets to the default settings while the adjusted handset setting is retained.
default settings used to begin each call	No adjusted handset and speaker settings are retained.
retain handset and speaker between calls	The adjusted handset and speaker settings are retained.
retain speaker, handset unchangeable	Only the adjusted speaker setting is retained.

PRIMARY LEVELS

Touch Tone Sidetone (dB)

Determines the touchtone volume fed back from the telephone when a users presses a button.

Touch Tone Transmit (dB) —

Determines the touchtone volume fed outbound from the telephone.

Voice Receive (dB)

Determines the volume of voice inbound to the telephone.

Voice Sidetone (dB)

Determines the volume of voice fed back from the handset voice microphone to the user’s ear.

Voice Transmit (dB)

Determines the volume of voice outbound from the telephone.

Note:

You cannot administer all five of the **Primary Level** fields to +14.0 dB. If you attempt to submit the **Terminal Parameters** screen with all **Primary Levels** set to +14.0 dB, you receive an error message.

ADJUNCT LEVELS

Touch Tone Sidetone (dB)

Determines the touchtone volume fed back from the telephone when a users presses a button.

Voice Receive (dB)

Determines the volume of voice inbound to the adjunct.

Voice Sidetone (dB)

Determines the volume of voice fed back from the handset voice microphone to the user’s ear.

Voice Transmit (dB)

Determines the volume of voice outbound from the adjunct.

Terminating Extension Group

This screen defines a Terminating Extension Group (TEG). Any telephone can be assigned as a TEG member; however, only a multi-appearance telephone can be assigned a **TEG** button with associated status lamp. The **TEG** button allows the telephone user to select a TEG call appearance for answering or for bridging onto an existing call.

The TEG members are assigned on an extension number basis. Call reception restrictions applicable to the group are specified by the group class of restriction (COR). The group COR takes precedence over an individual member's COR. The members could all be termination restricted but still receive calls if the group is not restricted.

The system allows for as many as 32 TEGs with up to 4 members each. An extension number can be assigned to more than one TEG but can have only one appearance of each group.

Field descriptions for page 1

Figure 530: Terminating Extension Group screen

```
change term-ext-group 1                                    Page   1 of   1
                      TERMINATING EXTENSION GROUP

    Group Number: 1                          Group Extension:
      Group Name:                              Coverage Path:
   Security Code:                                       COR: 1
                                                         TN: 1
ISDN Caller Disp:                                       LWC Reception: none
      AUDIX Name:

GROUP MEMBER ASSIGNMENTS

    Ext      Name (first 26 characters)    Ext       Name (first 26 characters)
 1: 51001    26 character name sta 51001   3:
 2:                                           4: 51002   26 character name sta 51002
```

AUDIX Name

Name of the AUDIX machine as it appears in the **Node Names** screen. Only appears for an DEFINITY R configuration.

Valid entries	Usage
Audix machine description	Unique identifiers for adjunct equipment.

COR

Valid entries	Usage
0 to **95**	Enter the class of restriction (COR) number that reflects the desired restrictions.

Coverage Path

Enter a number for the call coverage path for this group. A TEG cannot serve as a coverage point; however, calls to a TEG can redirect to coverage.

Valid entries	Usage
1 to **9999**	path number
t1 to **t999**	time of day table
blank	

Group Extension

Enter the extension of the terminating extension group.

Valid entries	Usage
1 to 7 digits	Unused extension number (may not be a VDN extension). Do not leave blank.

Group Name

Enter the name used to identify the terminating extension group.

Group Number

A display-only field when the screen is accessed using an administration command such as **add** or **change**.

ISDN Caller Disp

This field is required if, on the **System-Parameters Customer-Options** screen, the **ISDN-PRI** or **ISDN-BRI Trunks** field is **y**.

Valid entries	Usage
grp-name	Specify whether the TEG group name or member name (member of TEG where call terminated) will be sent to the originating user.
mbr-name	Specify whether the TEG group name or member name (member of TEG where call terminated) will be sent to the originating user.
blank	If the **ISDN-PRI** or **ISDN-BRI Trunks** field is **n**, leave blank.

LWC Reception

Defines the source for Leave Word Calling (LWC) messages.

Valid entries	Usage
audix	If LWC is attempted, the messages are stored in AUDIX.
spe	If LWC is attempted, the messages are stored in the system processing element (spe).
none	If LWC is attempted, the messages are not stored.

Security Code

Valid entries	Usage
3 to 8 digit security code	This code is used for the Demand Print feature.

TN

Valid entries	Usage
1 to **100**	Enter the Tenant Partition number.

GROUP MEMBER ASSIGNMENTS

Ext

Enter the extension number (may not be a VDN extension) assigned to a station.

Valid entries	Usage
1 to 7 digits	An extension number of 1 to 7 digits.

Name

This display-only field shows the name assigned to the preceding extension number when the TEG member’s telephone is administered.

TFTP Server

The **Trivial File Transfer Protocol** screen allows specification of the TFTP server that Avaya Communication Manager uses to get download files.

Figure 531: TFTP Server screen

```
change tftp-server                                          Page 1 of 1
                        TFTP Server Configuration

      Local Node Name:
TFTP Server Node Name:
     TFTP Server Port: 69
     File to Retrieve:

          File Status:
            File Size:
   Filename in Memory:
```

Filename in Memory

A display-only field showing the name of the file currently in Communication Manager memory.

File Size

A display-only field showing the number of bytes transferred.

File Status

A display-only field showing Download In Progress, Download Failed, File Not Found, or Download Completed.

File to Retrieve

Valid entries	Usage
up to 32 alpha-numeric, case sensitive, characters	Enter the name of the file you are going to retrieve using up to 32 alpha-numeric, case sensitive, characters for identification.

Local Node Name

The local node name must be a valid entry from the **IP Node Names** screen. The node must be assigned to a CLAN ip-interface or **procr** (processor CLAN).

Valid entries	Usage
1 to 15 characters procr	Valid entry from the **IP Node Names** screen. Processor CLAN for S8300 Media Server, S8700 or S8710 Multi-Connect, and S8700 or S8710 IP-Connect.

TFTP Server Node Name

Valid entries	Usage
1 to 15 characters	The TFTP server node name must be a valid entry from the **IP Node Names** screen.

TFTP Server Port

Valid entries	Usage
1 to **64500**	Enter a number for the remote TCP port.

Time of Day Coverage Table

This screen allows up to five different coverage paths, associated with five different time ranges, for each day of the week. Only one coverage path can be in effect at any one time.

Field descriptions for page 1

Figure 532: Time of Day Coverage Table screen

```
change coverage time-of-day 3
                  TIME OF DAY COVERAGE TABLE 3___

           Act   CVG     Act   CVG     Act   CVG     Act   CVG     Act   CVG
           Time  PATH    Time  PATH    Time  PATH    Time  PATH    Time  PATH

  Sun      00:00 _____   __:__ _____   __:__ _____   __:__ _____   __:__ _____
  Mon      00:00 _____   __:__ _____   __:__ _____   __:__ _____   __:__ _____
  Tue      00:00 _____   __:__ _____   __:__ _____   __:__ _____   __:__ _____
  Wed      00:00 _____   __:__ _____   __:__ _____   __:__ _____   __:__ _____
  Thu      00:00 _____   __:__ _____   __:__ _____   __:__ _____   __:__ _____
  Fri      00:00 _____   __:__ _____   __:__ _____   __:__ _____   __:__ _____
  Sat      00:00 _____   __:__ _____   __:__ _____   __:__ _____   __:__ _____
```

Act Time

Specify the activation time of the coverage path administered in the next **CVG PATH** field. Enter the information in 24-hour time format.

Valid entries	Usage
00:01 to **23:59**	If there are time gaps in the table, there will be no coverage path in effect during those periods. The first activation time for a day is set to 00:00 and cannot be changed. Activation times for a day must be in ascending order from left to right.

CVG Path

Enter the coverage path number.

Valid entries	Usage
1 to **9999** or blank	For the DEFINITY R configurations and S8700 or S8710 IP-Connect
1 to **999**	For the DEFINITY SI configurations

Time of Day Coverage Table

A display-only field when the screen is accessed using an administration command. Specifies the Time of Day Coverage Table number. Up to 999 can be administered.

Time of Day Routing Plan

Use this screen to set up Time of Day Routing Plans. You can route AAR and ARS calls based on the time of day each call is made. You can design up to 8 Time of Day Routing Plans, each scheduled to change up to 6 times a day for each day in the week.

Match the Time of Day Routing Plan PGN# with the **PGN#** field on the Partition Routing Table for the route pattern you want to use.

Note:

Automatic Route Selection (ARS) or **Private Networking**, **AAR/ARS Partitioning**, and **Time of Day Routing** must be enabled on the **System-Parameters Customer-Options** screen before you can use Time of Day Routing.

Figure 533: Time Of Day Routing Plan screen

```
change time-of-day
                              TIME OF DAY ROUTING PLAN ___            Page 1 of 1
     Act    PGN   Act   PGN   Act    PGN   Act    PGN   Act    PGN   Act    PGN
     Time    #    Time   #    Time    #    Time    #    Time    #    Time    #
Sun  00:00   1   __:__   _    __:__   _    __:__   _    __:__   _    __:__   _
Mon  00:00   1   __:__   _    __:__   _    __:__   _    __:__   _    __:__   _
Tue  00:00   1   __:__   _    __:__   _    __:__   _    __:__   _    __:__   _
Wed  00:00   1   __:__   _    __:__   _    __:__   _    __:__   _    __:__   _
Thu  00:00   1   __:__   _    __:__   _    __:__   _    __:__   _    __:__   _
Fri  00:00   1   __:__   _    __:__   _    __:__   _    __:__   _    __:__   _
Sat  00:00   1   __:__   _    __:__   _    __:__   _    __:__   _    __:__   _
```

Act Time

Specifies the time of day the route pattern (identified by PGN) begins.

Valid entries	Usage
00:00 to **23:59**	Time is represented using a 24 hour clock. List times for the same day in increasing order. There must be at least one entry for each day.

PGN

Identifies the route pattern for activation time listed.

Valid entries	Usage
1 to **8**	Enter a PGN that matches the PGN and route pattern on the Partition Routing Table. There must be at least one entry for each day.

Time of Day Routing Plan

Displays the Time of Day Routing Plan number (1 through 8).

Toll Analysis

Note:

The **Toll List** field on this screen does not interact with or relate to the ARS Toll Table.

This screen associates dialed strings to the system's Restricted Call List (RCL), Unrestricted Call List (UCL), and Toll List. You can force users to dial an account code if you associate dialed strings with CDR Forced Entry of Account Codes.

To maximize system security, Avaya recommends that toll calling areas be restricted as much as possible through the use of the **RCL (Restricted Call List)** and **Toll List** fields on this screen.

Figure 534: Toll Analysis screen

```
change toll 1                                                  Page 1 of 1
                                TOLL ANALYSIS                 Percent Full:_
                               Location:
                      Total          Toll  CDR  <--Unrestricted Call List-->
   Dialed String      Min  Max  RCL  List  FEAC  1  2  3  4  5  6  7  8  9 10
_____________________ ___ ___  _    _     _    _  _  _  _  _  _  _  _  _  _
_____________________ ___ ___  _    _     _    _  _  _  _  _  _  _  _  _  _
_____________________ ___ ___  _    _     _    _  _  _  _  _  _  _  _  _  _
_____________________ ___ ___  _    _     _    _  _  _  _  _  _  _  _  _  _
_____________________ ___ ___  _    _     _    _  _  _  _  _  _  _  _  _  _
_____________________ ___ ___  _    _     _    _  _  _  _  _  _  _  _  _  _
_____________________ ___ ___  _    _     _    _  _  _  _  _  _  _  _  _  _
_____________________ ___ ___  _    _     _    _  _  _  _  _  _  _  _  _  _
_____________________ ___ ___  _    _     _    _  _  _  _  _  _  _  _  _  _
_____________________ ___ ___  _    _     _    _  _  _  _  _  _  _  _  _  _
_____________________ ___ ___  _    _     _    _  _  _  _  _  _  _  _  _  _
_____________________ ___ ___  _    _     _    _  _  _  _  _  _  _  _  _  _
```

CDR FEAC

Valid entries	Usage
x	Enter **x** to require an account code from a call whose facility COR requires a Forced Entry of Account Code.

Dialed String

Valid entries	Usage
digits **0** to **9** (up to 18 characters)	Enter the dialed string you want Avaya Communication Manager to analyze.
***, x, X**	wildcard characters

Location

Display-only field.

Valid display	Usage
1 to **64**	Defines the server's location for this Toll Analysis Table. On the **System-Parameters Customer-Options** screen, the **ARS** field and the **Multiple Locations** field must be set to **y** for values other than **all** to appear.
all	Indicates that this Toll Analysis Table is the default for all port network (cabinet) locations.

Max

Valid entries	Usage
Min to **28**	Enter the maximum number of user-dialed digits the system collects to match to this dialed string.

Min

Valid entries	Usage
1 to **Max**	Enter the minimum number of user-dialed digits the system collects to match to this dialed string.

Percent Full

Display-only field showing the percentage (**0** to **100**) of the system's memory resources that have been used by AAR/ARS. If the figure is close to 100%, you can free-up memory resources.

RCL

Enter **x** to assign the Dialed String to the Restricted Call List (RCL).

Valid entries	Usage
x	All entries of **x** and their associated dialed strings are referred to as the System's Restricted Call List. The RCL can be assigned to any COR. A call attempt from a facility whose COR is marked as being associated with the RCL and whose dialed string matches a RCL dialed string field will be denied. The caller receives intercept treatment.

Toll List

Valid entries	Usage
x	Enter **x** to assign the Dialed String to the Toll List.

Dialed String	Min	Max	Toll List
0	1	23	x
1	4	23	x
20	10	10	x
21	10	10	x
30	10	10	x
31	10	10	x
40	10	10	x
41	10	10	x
50	10	10	x
51	10	10	x
60	10	10	x
61	10	10	x
70	10	10	x

1 of 2

Dialed String	Min	Max	Toll List
71	10	10	x
80	10	10	x
81	10	10	x
90	10	10	x
91	10	10	x
			2 of 2

Unrestricted Call List

Valid entries	Usage
x	Enter **x** to assign the dialed string to one of the system's Unrestricted Call Lists (UCL).

Tone Generation

The **Tone Generation** screen allows you to administer the tone characteristics that parties on a call hear under various circumstances.

Note:

With the Multinational Locations feature enabled, tone generation can be administered per location, rather than system-wide.

Figure 535: Tone Generation screen

```
change tone-generation 2                                    Page   1 of  X
                            TONE GENERATION 2

                                                 Base Tone Generator Set: 1
          440Hz PBX-dial Tone? n              440Hz Secondary-dial Tone? n
```

Field descriptions for page 1

440Hz PBX-dial Tone

Specifies whether the switch (primary) dial tone will be changed to a continuous 440Hz/-17 tone.

Valid entries	Usage
y/n	A value of **n** implies the tone will either be administered on a later page of this screen or, if no individual definition is administered, as defined in **Base Tone Generation Set**.

440Hz Secondary-dial Tone

Specifies whether the Secondary (CO) dial tone will be changed to a continuous 440Hz/-17 tone.

Valid entries	Usage
y/n	A value of **n** implies the tone will either be administered on a later page of this screen or, if no individual definition is administered, as defined in **Base Tone Generation Set**.

Base Tone Generator Set

The country code identifies the base tone generation set to be used. For information on the appropriate tone-generation hardware to use in a specific country, see *Hardware Guide for Avaya Communication Manager,* 555-245-207.

Valid entries	Usage
1 to **25**	See the Country code table at the beginning of the **System-Parameters Country-Code** screen description.

Field descriptions for page 2

Figure 536: Tone Generation screen

```
change tone-generation                                        Page   2 of  X
                      TONE GENERATION CUSTOMIZED TONES

        Tone Name        Cadence        Tone
                          Step    (Frequency/Level)
                            1:
                            2:
                            3:
                            4:
                            5:
                            6:
                            7:
                            8:
                            9:
                           10:
                           11:
                           12:
                           13:
                           14:
                           15:
```

Cadence Step

Display-only fields that identify the number of each tone cadence step.

Valid entries	Usage
1 to **15**	Identifies the number of each tone cadence step.

Tone Name

Each entry uses one of the keywords below to indicate which of the individually administrable tones this screen modifies.

Valid entries	Usage
blank	If this field is blank, all entries are ignored in the corresponding **Tone (Frequency/Level)** field.
1-call-wait **2-call-wait** **3-call-wait** **busy** **busy-verify** **call-wait-ringback** **conference** **confirmation** **disable-dial** **hold** **hold-recall** **immed-ringback** **intercept** **intrusion** **PBX-dial** **recall-dial** **recall-dont-ans** **redirect** **reorder** **rep-confirmation** **reset-shift** **ringback** **secondary-dial** **special-dial** **whisper-page** **zip**	

Tone (Frequency/Level)

Valid entries	Usage
silence	An entry of **silence** means no tone. A final step of **silence** with an infinite duration will be added internally to any tone sequence that does not end in a **goto**.
goto	An entry of **goto** means to repeat all or part of the sequence, beginning at the specified cadence step.
350/-17.25 **350+425/-4.0** **350+440/-13.75** **375+425/-15.0** **404/-11.0** **404/-16.0** **404+425/-11.0** **404+450/-11.0** **425/-4.0** **425/-11.0** **425/-17.25** **440/-17.25** **440+480/-19.0** **450/-10** **480/-17.25** **480+620/-24.0** **525/-11.0** **620/-17.25** **697/-8.5** **770/-8.5** **852/-8.5** **941/-8.5** **1000/0.0** **1000/+3.0** **1004/0.0** **1004/-16.0** **1209/-7.5** **1336/-7.5** **1400/-11.0** **1477/-7.5** **1633/-7.5** **2025/-12.1** **2100/-12.1** **2225/-12.1** **2804/-16.0**	Specifies the frequency and level of the tone.

Trunk Group

Use the **Trunk Group** screen to set basic characteristics for every type of trunk group and to assign ports to the group. This section lists and describes all the fields you might see on the screen. Many fields are dependent on the settings of other fields and only appear when certain values are entered in other fields on the screen. For example, the entry in the **Group Type** field may significantly change the content and appearance of the **Trunk Group** screen.

For more information on administering trunk groups, see Managing Trunks on page 423.

This section does not cover ISDN trunks. For descriptions of the screens and fields that are unique to ISDN trunks, see ISDN Trunk Group on page 1170.

Field descriptions for page 1

The figure below is only an example, and is intended to show most of the fields that might appear on page 1 of the **Trunk Group** screen. This example may might not show all fields, or might show fields that normally do not appear together; it is not intended to reflect a specific trunk group type. Your own screen mary vary from this example according to specific field and system settings. The list of field descriptions that follow the figure is in alphabetical order for quick reference. This list is intended to be comprehensive, and may include information on fields that are not shown in the example. The field descriptions identify fields that are specific to particular trunk group types.

Figure 537: Trunk Group screen - page 1

```
add trunk-group next                                          Page 1 of x
                                TRUNK GROUP

 Group Number: ___                  Group Type: _____      CDR Reports: _
   Group Name: _____________________________ COR: __       TN: ___      TAC: ____
    Direction: ________       Outgoing Display? _    Trunk Signaling Type: ____
 Dial Access? _                 Busy Threshold: ___         Night Service: ________
 Queue Length: ___                                   Incoming Destination: ________
    Comm Type: _____                 Auth Code? _
                                   Trunk Flash? _
          BCC: _                            ITC? ____
 TRUNK PARAMETERS
    Trunk Type (in/out): _____________     Incoming Rotary Timeout(sec): __
     Outgoing Dial Type: _________                  Incoming Dial Type: ________
                                             Disconnect Timing(msec): ____
        Digit Treatment: __________                          Digits: ____
                                                  Sig Bit Inversion: ____
      Analog Loss Group: ___                      Digital Loss Group: ___
     Incoming Dial Tone? _
Incoming Calling Number - Delete: ___ Insert: _______ Format: _______
               Bit Rate: _____        Synchronization: _____    Duplex: ____
  Disconnect Supervision - In? _  Out? _
  Answer Supervision Timeout: ___           Receive Answer Supervision? _
```

Analog Gain

Use this field to reduce the strength of incoming signals on TN2199 ports if users regularly experience echo, distortion, or unpleasantly loud volume. Experiment to find the best setting for your situation. This field appears if the **Country** field is **15** and the **Trunk Type (in/out)** field is **2-wire-ac**, **2-wire-dc**, or **3-wire**.

Valid entries	Usage
a	Reduces the incoming signal by -3dB.
b	Reduces the incoming signal by -6dB.
c	Reduces the incoming signal by -8dB.
none	No reduction. Don’t change this setting unless the trunk group’s sound quality is unacceptable.

Analog Loss Group

This field determines which administered 2-party row in the loss plan applies to this trunk group if the call is carried over an analog signaling port in the trunk group.

Valid entries	Usage
1 to **17**	Shows the index into the loss plan and tone plan.

Answer Supervision Timeout

If the **Receive Answer Supervision** field is **n**, use this field to set the answer supervision timer for outgoing and two-way trunks. During a cut-through operation, timing begins after each outgoing digit is sent by Communication Manager and timing ceases after the far-end sends answer supervision. If the timer expires, Communication Manager acts as if it had received answer supervision. On senderized operation, the timer begins after the last digit collected is sent.

Valid entries	Usage
0 to **250**	Enter the number of seconds you want Communication Manager to wait before it acts as though answer supervision has been received from the far-end. Set this field to **0** if **Receive Answer Supervision** is **y**.

Note:

This field's setting does not override answer supervision sent from the network or from DS1 port circuit timers. To control answer supervision sent by DS1 firmware, set the **Outgoing End of Dial (sec)** field on the **Administrable Timers** page of the **Trunk Group** screen.

Auth Code

This field affects the level of security for tandemed outgoing calls at your server running Communication Manager. This field appears if the **Direction** field is incoming or two-way, and it can only be **y** if the **Authorization Codes** field is **y** on the **System-Parameters Customer-Options** screen.

Valid entries	Usage
y/n	Enter **y** to require callers to enter an authorization code in order to tandem a call through an AAR or ARS route pattern. The code will be required even if the facility restriction level of the incoming trunk group is normally sufficient to send the call out over the route pattern.

Auto Guard

This field appears if the **Group Type** field is **co** or **fx**. This field controls ports only on TN438B, TN465B, and TN2147 circuit packs. TN438B ports have hardware support for detecting a defective trunk. TN465B and TN2147 ports consider a trunk defective if no dial tone is detected on an outgoing call, and the **Outpulse Without Tone** field is **n** on the **Feature-Related System Parameters** screen.

Valid entries	Usage
y/n	Enter **y** to prevent repeated seizures of a defective trunk. Communication Manager will do a maintenance busy-out on these trunks.

BCC

Generalized Route Selection uses the BCC to select the appropriate facilities for routing voice and data calls. Far-end tandem servers/switches also use the BCC to select outgoing routing facilities with equivalent BCC classes.The entry in the **Bearer Capability Class** field is used to select the appropriate facilities for incoming ISDN calls. Avaya Communication Manager compares the entry in the **BCC** field to the value of the Bearer Capability information element for the incoming call and routes the call over appropriate facilities. For example, a call with BCC 4 will only be connected through facilities that support 64 kbps data transmission.

The **Bearer Capability Class** field appears when all of the following are true:

- Either the **ISDN-BRI Trunks** field or the **ISDN-PRI** field on the **System Parameters Customer-Options** screen is **y**.
- The **Group Type** field is **access**, **co**, **fx**, **tandem**, **tie**, or **wats**.
- The **Comm Type** field is **data**, **avd**, or **rbavd**.

Valid entries	Usage
0	For voice and voice-grade data
1	For 56 kbps synchronous data transmitted with robbed-bit signaling
2	Less than 19.2 kbps synchronous or asynchronous data
3	For 64 kbps data and LAPD protocol
4	For 64 kbps data on unrestricted channels

Bit Rate

This field specifies the baud rate to be used by pooled modems. This field appears when the **Comm Type** field is **avd** or **rbavd**. It also appears if the **Comm Type** field is **data**, but only if the **ISDN-PRI** field is **y** on the **System-Parameters Customer-Options** screen.

Valid entries	Usage
300	Enter the speed of the fastest modem that will use this trunk group.
1200	
2400	
4800	
9600	
19200	

Busy Threshold

Use this field if you want attendants to control access to outgoing and two-way trunk groups during periods of high use. When the threshold is reached and the warning lamp for that trunk group lights, the attendant can activate trunk group control: internal callers who dial out using a trunk access code will be connected to the attendant, and the attendant can prioritize outgoing calls for the last remaining trunks. Calls handled by AAR and ARS route patterns go out normally.

Valid entries	Usage
1 to **255** (DEFINITY R, S8700 or S8710 IP-Connect)	Enter the number of trunks that must be busy in order to light the warning lamp on the Attendant Console. For example, if there are 30 trunks in the group and you want to alert the attendant whenever 25 or more are in use, enter **25**.

Call Still Held

This field appears if the **Group Type** field is **co** or **fx**. This field is used when the receiving end media server or switch initiates the disconnection of incoming calls. It effectively extends the Incoming Glare Guard timer by 140 seconds. This field affects only TN438B, TN465B, and TN2147 ports and is used primarily when the Country Code field is **2**.

Valid entries	Usage
y/n	Enter **y** to prevent glare by delaying an outgoing seizure of a trunk for at least 140 seconds after it is released from an incoming call.

CDR Reports

Valid entries	Usage
y	All outgoing calls on this trunk group will generate call detail records. If the **Record Outgoing Calls Only** field on the **CDR System Parameters** screen is **n**, then incoming calls on this trunk group will also generate call detail records.
n	Calls over this trunk group will not generate call detail records.
r (ring-intvl)	CDR records will be generated for both incoming and outgoing calls. In addition, the following ringing interval CDR records are generated: • Abandoned calls: The system creates a record with a condition code of "H," indicating the time until the call was abandoned. • Answered calls: The system creates a record with a condition code of "G," indicating the interval from start of ring to answer. • Calls to busy stations: The system creates a record with a condition code of "I," indicating a recorded interval of 0.

Note:

For ISDN trunk groups, the **Charge Advice** field affects CDR information. For CO, DIOD, FX, and WATS trunk groups, the **PPM** field affects CDR information.

CESID I Digits Sent

For emergency 911 service, Communication Manager may send Caller's Emergency Service Identification (CESID) information to the central office or E911 tandem server/switch. This digit string is part of the E911 signaling protocol. This field appears when **Group Type** is **cama**.

Valid entries	Usage
1 to 3 digits	Determine the correct entry for this field by talking to your E911 provider.

Comm Type

Use this field to define whether the trunk group carries voice, data, or both.

Note:

Comm Types of **avd**, **rbavd** and **data** require trunk member ports on a DS1 circuit pack.

Valid entries	Usage
avd	Enter **avd** for applications that mix voice and Digital Communication Protocol data, such as video conferencing applications. The receiving end media server or switch discriminates voice calls from data calls and directs each to an appropriate endpoint. Neither originating nor terminating ends insert a modem pool for any calls when **Comm Type** is **avd**. The **Signaling Mode** field on the **DS1 Circuit Pack** screen must be set for either **common-chan** or **CAS** signaling.
data	Enter **data** only when all calls across the trunk group originate and terminate at Avaya Communication Manager digital data endpoints. Public networks don't support **data**: supported by Avaya's DCP protocol, this entry is used almost exclusively for the data trunk group supporting DCS signaling channels. The **Signaling Mode** field on the **DS1 Circuit Pack** screen may be set to **robbed-bit** or **common-chan**.
rbavd	For digital trunk groups that carry voice and data with robbed-bit signaling. The **Signaling Mode** field on the **DS1 Circuit Pack** screen must be set to **robbed-bit** unless mixed mode signaling is allowed on the DS1 circuit pack. In that case, the **Signaling Mode** field may be **isdn-ext** or **isdn-pri**.
voice	For trunk groups that carry only voice traffic and voice-grade data (that is, data transmitted by modem). Analog trunk groups must use **voice**.

COR

Valid entries	Usage
0 to **95**	Enter a class of restriction (COR). Classes of restriction control access to trunk groups, including trunk-to-trunk transfers.

Tip:

Remember that facility restriction levels (FRL) are assigned to classes of restriction. Even if 2 trunk groups have classes of restriction that allow a connection, different facility restriction levels may prevent operations such as off-net call forwarding or outgoing calls by remote access users.

CO Type

This field appears when the **Country** field is **14** and is used only by trunk group members administered on a TN464D vintage 2 or later DS1 circuit pack.

Valid entries	Usage
analog	This field specifies whether the trunk group is connected to analog or digital facilities at the central office.
digital	

Country

This field is administered at installation and sets numerous parameters to appropriate values for the public network in which the server running Communication Manager operates. For example, the value of this field, with the values of the **Trunk Termination** and the **Trunk Gain** fields, determines the input and trans-hybrid balance impedance requirements for ports on TN465B, TN2146, and TN2147 circuit packs.

This field appears for the trunk groups that connect Avaya Communication Manager to a central office in the public network — CO, DID, DIOD, FX, and WATS trunk groups.

CAUTION:

Do not change this field. If you have questions, contact your Avaya technical support representative.

Note:

For DID trunk types, country code 19 is not accepted in the **Trunk Group** screen in Communication Manager. This will be supported at a later date.

Valid entries	Usage
1 to **25**	Set at installation.
11	If the **Country** field is **11**, Avaya Communication Manager is administered for Public Network Call Priority (Call Retention and Re-ring).
14	If the **Country** field is **14** and the **Group Type** is **DID** or **DIOD**, the **CO Type** field appears.
15	If the **Country** field is **15**, Communication Manager is administered for Public Network Call Priority (Intrusion and Re-ring). Also, the **Protocol Type** field appears for **Group Type DID** or **DIOD**.
18	If the **Country** field is **18**, Avaya Communication Manager can be administered for Public Network Call Priority (Mode of Release Control, Forced Disconnect, and Re-ring).
23	If the **Country** field is **23** and **Group Type** field is either **CO** or **DID**, Communication Manager is administered for Block Collect Calls.

Cut-Through

This field appears when the **Outgoing Dial Type** field is either **rotary** or **tone**.

SECURITY ALERT:
Entering **y** in this field will reduce your ability to prevent toll fraud.

Valid entries	Usage
y	Enter **y** to allow users to get dial tone directly from the central office. Outgoing calls over this trunk group will bypass AAR/ARS (if you're using it) and any of your administered restrictions (such as COR or FRL).
n	Enter **n** and the user will receive switch dial tone. Instead of digits being sent to the central office, they will be collected and checked against administered restrictions. If no restrictions apply, the digits are sent to the central office.

Cyclical Hunt

When a call is offered to a trunk group, Communication Manager searches for an available trunk. This field, which appears when the **Direction** field is **two-way** and the **Trunk Type** field is **loop-start**, controls the starting point of this search.

You can change this field from **n** to **y** at any time. To change from **y** to **n**, however, all the trunks in the group must be idle or busied out.

Valid entries	Usage
y	Enter **y** to have Communication Manager start its search from the last trunk seized. This method is faster, and thus better suited for high-traffic trunk groups.
n	Enter **n** to have Communication Manager start each search at member 1 (the first trunk administered on the Group Member Assignments page).

Dial Access

This field controls whether users can route outgoing calls through an outgoing or two-way trunk group by dialing its trunk access code. Allowing dial access does not interfere with the operation of AAR/ARS.

SECURITY ALERT:
Calls dialed with a trunk access code over WATS trunks bypass AAR/ARS and aren't restricted by facility restriction levels. For security, you may want to leave the field set to n unless you need dial access to test the trunk group.

Valid entries	Usage
y	Allows users to access the trunk group by dialing its access code.
n	Does not allow users to access the trunk group by dialing its access code. Attendants can still select this trunk group with a **Trunk Group Select** button.

Dial Detection

Applies only to TN2199 ports. The **Country** field must be **15**.

Valid entries	Usage
A-wire	
B-wire	

Digit Absorption List

This field assigns a digit absorption list, when used, to a trunk group that terminates at a step-by-step central office.

Valid entries	Usage
0 to **4** or blank	Enter the number of the digit absorption list this trunk group should use.

Digits

This field is used with the **Digit Treatment** field, and its meaning depends on the entry in that field. If the **Digit Treatment** field is **absorption**, this field specifies *how many* digits are deleted. If the **Digit Treatment** field is **insertion**, this field identifies the *specific digits* that are added.

Valid entries	Usage
1 to **5**	Enter the number of digits to be deleted (absorbed).
Up to 4 digits, including * and #	Enter the actual digits to be added (inserted).
blank	This field can be blank only if the **Digit Treatment** field is blank.

Digit Treatment

Use this field to modify an incoming digit string (as on DID and tie trunks, for example) by adding or deleting digits. You'll need to do this if the number of digits you receive doesn't match your dial plan.

Valid entries	Usage
blank	The incoming digit string is not changed.
absorption	Deletes digits, starting at the beginning of the string.
insertion	Adds digits, starting at the beginning of the string.

If you enter absorption or insertion, then you must enter a value in the **Digits** field.

Note:

In a DCS network, DCS features that use the **remote-tgs** button (on phones at a remote end) do not work when the incoming trunk group at your end deletes or inserts digits on incoming calls. The **remote-tgs** button on a remote server/ switch, for example, tries to dial a TAC on your switch. If your end adds or deletes digits, it defeats this operation. If you need to manipulate digits in a DCS network (for example, to insert an AAR feature access code), do it on the outgoing side based on the routing pattern.

Digital Loss Group

This field determines which administered 2-party row in the loss plan applies to this trunk group if the call is carried over a digital signaling port in the trunk group.

Valid entries	Usage
1 to **19**	Shows the index into the loss plan and tone plan.

Direction

Enter the direction of the traffic on this trunk group. The entry in this field affects which timers appear on the **Administrable Timers** page. This field appears for all trunk groups except DID and CPE.

Valid entries	Usage
incoming	
outgoing	
two-way	Enter two-way for Network Call Redirection.

Disconnect Supervision-In

This field indicates whether Avaya Communication Manager receives disconnect supervision for incoming calls over this trunk group. It appears when the **Direction** field is **incoming** or **two-way**. (If the **Direction** field is **outgoing**, Communication Manager internally sets this field to **n**.)

The entry in this field is crucial if you allow trunk-to-trunk transfers. (To allow trunk-to-trunk transfers involving trunks in this group, this field must be **y** and the **Trunk-to-Trunk Transfer** field on the **Feature-Related System Parameters** screen must be **y**).

If a user connects 2 trunks through conference or transfer, and neither far-end media server or switch on the resulting connection provides disconnect supervision, the trunks involved will not be released because Avaya Communication Manager cannot detect the end of the call. Communication Manager will not allow trunk-to-trunk transfers unless it believes that at least one party on the call can provide disconnect supervision. Therefore, setting this field incorrectly may cause trunks to become unusable until the problem is detected and the trunks are reset.

Valid entries	Usage
y	Enter **y** to allow trunk-to-trunk transfers involving trunks in this group. Enter **y** if the far-end server/switch sends a release signal when the calling party releases an incoming call, and you want to make the far-end server/switch responsible for releasing the trunk. Enter **y** to enhance Network Call Redirection.
n	Enter **n** if the far-end server/switch doesn't provide a release signal, if your hardware can't recognize a release signal, or if you prefer to use timers for disconnect supervision on incoming calls. Entering **n** prevents trunk-to-trunk transfers involving trunks in this group.

CAUTION:

In general, U.S. central offices provide disconnect supervision for incoming calls but not for outgoing calls. Public networks in most other countries do not provide disconnect supervision for incoming or outgoing calls. Check with your network services provider.

Disconnect Supervision-Out

This field indicates whether Communication Manager receives disconnect supervision for outgoing calls over this trunk group. It appears when the **Direction** field is either **outgoing** or **two-way**. (If the **Direction** field is **incoming**, Communication Manager internally sets this field to **n**.)

The entry in this field is crucial if you allow trunk-to-trunk transfers. (To allow trunk-to-trunk transfers involving trunks in this group, this field must be **y** and the **Trunk-to-Trunk Transfer** field on the **Feature-Related System Parameters** screen must be **y**). If a user connects 2 trunks through conference or transfer, and neither far-end server/switch on the resulting connection provides disconnect supervision, the trunks involved won't be released because Communication Manager can't detect the end of the call. Communication Manager will not allow trunk-to-trunk transfers unless it believes that at least one party on the call can provide disconnect supervision. Therefore, setting this field incorrectly may cause trunks to become unusable until the problem is detected and the trunks are reset.

Also, remember that Avaya Communication Manager must receive *answer* supervision on outgoing analog CO, FX, WATS, Tie, Tandem, and Access trunks before it will recognize a disconnect signal. If this trunk group does not receive *answer* supervision from the far-end server/switch, and you enter **y** in this field, Avaya Communication Manager will internally set the field to **n**.

Valid entries	Usage
y	Enter **y** to allow trunk-to-trunk transfers involving trunks in this group. Enter **y** if the far-end sends a release signal when the called party releases a call an outgoing call, and you want to make the far-end responsible for releasing the trunk. The **Answer Supervision Timeout** field must be **0** and the **Receive Answer Supervision** field must be **y** for the switch to recognize a **y** entry. Enter **y** to enhance Network Call Redirection.
n	Enter **n** if the far-end server/switch doesn't provide a release signal, if your hardware can't recognize a release signal, or if you prefer to use timers for disconnect supervision on outgoing calls. Entering **n** prevents trunk-to-trunk transfers involving trunks in this group.

CAUTION:

Do not set this field to **y** unless you are certain that the far-end server/switch will provide answer supervision and disconnect supervision. Most public networks do not provide disconnect supervision over analog trunks. Check with your network services provider.

Disconnect Timing (msec)

This field specifies the minimum time in milliseconds that the central office or far-end media server or switch requires to recognize that your end has disconnected from a call. This timer does not affect ports on a circuit pack that uses the administrable Incoming Disconnect and Outgoing Disconnect timers; in fact, settings on those two timers override this field.

Valid entries	Usage
140 to **2550** ms in increments of 10	The default of **500** is an industry standard and you shouldn't change it. If you set this field too high, your server/switch won't disconnect sometimes when it should; too low, and it will disconnect when it shouldn't.

Disconnect Type

This field indicates which side or user controls the disconnect, where A refers to the calling party and B refers to the called party. Appears only if the **Country** field is **15** and the **Trunk Type** field is **2-wire-ac**, **2-wire-dc**, or **3-wire**.

This applies *only* to the TN2199 port.

Valid entries	Usage
AandB	
AorB	

Drop Treatment

This field only applies to DID trunks. It determines what the calling party hears when the called party terminates an incoming call.

Valid entries	Usage
intercept	Select one. For security reasons, it's better to apply a tone: silence could provide an opening for hackers.
busy	
silence	

Note:

In Italy, the **Drop Treatment** field must be administered as **intercept** for all DID trunk groups.

Duplex

This field specifies whether a two-way trunk group allows simultaneous transmission in both directions. This field appears when the **Comm Type** field is **avd** or **rbavd**. It also appears if the **Comm Type** field is **data**, but only if the **ISDN-PRI** field is enabled on the **System-Parameters Customer-Options** screen.

Note:

Even if the trunk group supports full-duplex transmission, other equipment in a circuit may not.

Valid entries	Usage
full	Enter **full** in most cases: this allows simultaneous two-way transmission, which is most efficient.
half	Enter **half** to support only one transmission direction at a time.

End-to-End Signaling

Auxiliary equipment such as paging equipment and music sources may be connected to Avaya Communication Manager by auxiliary trunks. Communication Manager may send DTMF signals (touch tones) to these devices. This field, which appears when the **Group Type** field is **cpe** (customer-provided equipment trunk groups), sets the duration of these tones.

Valid entries	Usage
60 to **360** ms in increments of 10	Use this field to set the duration in milliseconds of the touch-tone signal that is sent to the connected equipment.

Note:

For trunks that do not receive real answer supervision, a "connect" Event report is sent when the Answer Supervision Timeout occurs.

Expected Digits

Note:

Set this field to **blank** if the **Digit Treatment** field is set to **insert** and the **Digits** field contains a feature access code (for example, AAR or ARS) followed by digits. In this case, the number of digits expected are set on the AAR and ARS Digit Analysis Table and AAR and ARS Digit Conversion Table.

Valid entries	Usage
1 to **18**	Enter the number of digits that the far-end media server or switch sends for an incoming connection. If your end is absorbing digits on this trunk group, the entry in this field must be larger than the entry in the **Digits** field.
	If you leave this field blank, you cannot administer digit absorption.

Extended Loop Range

This field appears only for a DID trunk group and is used only with the TN459A circuit pack.

Valid entries	Usage
y/n	Enter **y** or **n** depending on the distance between the central office and your server. If greater than the required distance, then the field should be **y**.

Group Name

Valid entries	Usage
1 to 27 characters	Enter a unique name that provides information about the trunk group. Do not use the default entry or the group type (DID, WATS) here. For example, you might use names that identify the vendor and function of the trunk group: USWest Local; Sprint Toll, etc.

Note:

For 4620SW or 4610SW telephones, the **Group Name** field is supported by Unicode language display. For more information on Unicode language display, see Administering Unicode display on page 181.

Group Number

This field displays the group number assigned when the trunk group was added.

Note:

For trunk groups connecting 2 media servers or switches in Distributed Communication System networks, Avaya recommends that you assign the same group number on both servers or switches.

Group Type

Enter the type of trunk group. The fields that are displayed and available may change according to the trunk group type selected. Busy-out the trunk group before you change the group type. Release the trunk group after you make the change. For more information about busying out and releasing trunk groups, see your system's Maintenance manual.

For more information about ISDN trunk groups, see the ISDN Trunk Group section of the Screen Reference chapter of this guide.

Valid entries	Usage
Access	Use access trunks to connect satellite media servers or switches to the main switch in Electronic Tandem Networks (ETN). Access trunks do not carry traveling class marks (TCM) and thus allow satellite callers unrestricted access to out-dial trunks on the main server. Allows Inband ANI.
APLT	Advanced Private Line Termination (APLT) trunks are used in private networks. APLT trunks allow inband ANI.
CAMA	CAMA trunks route emergency calls to the local community's Enhanced 911 systems.
CO	CO trunks typically connect Communication Manager to the local central office, but they can also connect adjuncts such as external paging systems and data modules.
CPE	Use CPE trunks to connect adjuncts, such as paging systems and announcement or music sources, to the server running Communication Manager.
DID	Use DID trunks when you want people calling your organization to dial individual users directly without going through an attendant or some other central point. Allows Inband ANI.

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Valid entries	Usage
DIOD	DIOD trunks are two-way trunks that transmit dialed digits in both directions. In North America, use tie trunks for applications that require two-way transmission of dialed digits. Allows Inband ANI.
DMI-BOS	Digital Multiplexed Interface - Bit-Oriented Signaling (DMI-BOS) trunks allow communication with systems using DMI-BOS protocol. DMI-BOS trunks allow inband ANI.
FX	An FX trunk is essentially a CO trunk that connects your server running Communication Manager directly to a central office outside your local exchange area. Use FX trunks to reduce long distance charges if your organization averages a high volume of long-distance calls to a specific area code.
ISDN	Use ISDN trunks when you need digital trunks that can integrate voice, data, and video signals and provide the bandwidth needed for applications such as high-speed data transfer and video conferencing. ISDN trunks can also efficiently combine multiple services on one trunk group. Use ISDN for Network Call Transfer. For more information, see *Avaya MultiVantage Call Center Software Release 11 — Network Call Redirection*. You cannot enter isdn unless the **ISDN-PRI** field, the **ISDN-BRI Trunks** field, or both have been enabled on the **System-Parameters Customer-Options** screen.
RLT	Release Link trunks work with Centralized Attendant Service in a private network.
SIP	Use SIP trunks to connect a server running Communication Manager to a SIP service provider’s server or switch.
Tandem	Tandem trunks connect tandem nodes in a private network. Tandem trunks allow inband ANI.

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Valid entries	Usage
Tie	Use tie trunks to connect a server running Communication Manager to a central office or to another server or switch in a private network. Tie trunks transmit dialed digits with both outgoing and incoming calls, and allow inband ANI.
WATS	Use WATS trunks to reduce long-distance bills when your organization regularly places many calls to a specific geographical area in North America. Outgoing WATS service allows calls to certain areas ("WATS bands") for a flat monthly charge. Incoming WATS trunks allow you to offer toll-free calling to customers and employees.

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Incoming Calling Number - Delete

The Incoming **Calling Number - Delete**, **Insert**, and **Format** fields are the administrable fields for the Calling Line Identification Prefix feature. They appear when the **Direction** field is **incoming** or **two-way**.

Valid entries	Usage
1 to **15**, **all**, or blank	Enter the number of digits, if any, to delete from the calling party number for all incoming calls on this trunk group.

Incoming Calling Number - Insert

Valid entries	Usage
Enter up to 15 characters (0 to 9), **all**, or blank	Enter up to specific digits, if any, to add to the beginning of the digit string of incoming calls when the calling party is a member of this trunk group.

Incoming Calling Number - Format

This field indicates the TON/NPI encoding applied to CPN information modified by the CLI Prefix feature. This encoding does not apply to calls originating locally. The **Numbering Format** field on page 2 of this screen applies to calls originated from this server running Communication Manager.

If this field is blank, Avaya Communication Manager passes on the encoding received in the incoming setup message. If the incoming setup message did not contain CPN information and digits are added, the outgoing message will contain these digits. If the **Format** field is blank in this case, the value defaults to pub-unk.

If the **Format** field on page 2 of this screen is also administered as **unknown**, the trunk group is modified to "unk-unk" encoding of the TON/NPI. Therefore, this field also must contain a value other than **unknown**.

Note:

The values for this field map to the Type of Numbering (TON) and Numbering Plan Identifier (NPI) values shown below.

Valid entries	Type of numbering (TON)	Numbering plan identifier (NPI)
blank	incoming TON unmodified	incoming NPI unmodified
natl-pub	national(2)	E.164(1)
intl-pub	international(1)	E.164(1)
locl-pub	local/subscriber(4)	E.164(1)
pub-unk	unknown(0)	E.164(1)
lev0-pvt	local(4)	Private Numbering Plan - PNP(9)
lev1-pvt	Regional Level 1(2)	Private Numbering Plan - PNP(9)
lev2-pvt	Regional Level 2(1)	Private Numbering Plan - PNP(9)
unk-unk	unknown(0)	unknown(0)

Incoming Destination

Use this field to set the destination for all incoming calls on trunk groups such as CO, FX, and WATS that must terminate at a single destination. The destination you enter here is also the default night service destination unless you enter a different destination in the **Night Service** field. Appears when the **Direction** field is **incoming** or **two-way**.

Valid entries	Usage
blank	Leave this field blank if the **Trunk Type** (in/out) field is not **auto/….**
an extension number	Calls go to the extension you enter. You may enter any type of extension, though typically the extension entered here identifies a VDN, a voice response unit, or a voice messaging system. Night service overrides this setting when it's active. **Note:** If entering a Multi-Location Dial Plan shortened extension, note the following: When entering a Multi-Location Dial Plan shortened extension in a field designed for announcement extensions, certain administration end validations that are normally performed on announcement extensions are not done, and resultant warnings or submittal denials do not occur. The shortened extensions also do not appear in any display or list that shows announcement extensions. Extra care should be taken to administer the correct type of announcement for the application when assigning shortened extensions..
attd	Calls go to the attendant and are recorded as Listed Directory Number (LDN) calls on call detail records.

Incoming Dial Tone

Indicates whether or not your server running Communication Manager will give dial tone in response to far-end seizures of the trunk group.

Valid entries	Usage
y	Enter **y** if the incoming trunk group transmits digits. For example, you would enter **y** for two-way, dial-repeating tie trunks that users select by dialing a trunk access code.
n	Enter **n** for trunks that aren't sending digits, such as tandem or incoming CO trunks.

Incoming Dial Type

Indicates the type of pulses required on an incoming trunk group. Usually, you should match what your central office provides. See "Trunks and Trunk Groups" in *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205, for more information. This field appears for Access, APLT, DID, DIOD, DMI-BOS, FX, RLT, Tandem, and WATS trunk groups. It also appears for Tie trunk groups when the **Trunk Signaling Type** field is blank, **cont**, or **dis**.

Valid entries	Usage
tone	Enter **tone** to use Dual Tone Multifrequency (DTMF) addressing, also known as "touch tone" in the U.S. Entering **tone** actually allows the trunk group to support both DTMF and rotary signals. Also, if you're using the Inband ANI feature, enter **tone**. For pulsed and continuous E&M signaling in Brazil and for discontinuous E&M signaling in Hungary, use **tone**.
rotary	Enter **rotary** if you only want to allow the dial pulse addressing method used by non-touch tone phones. Though the tone entry supports rotary dialing as well, it's inefficient to reserve touch tone registers for calls that don't use DTMF.
mf	Enter **mf** if the **Trunk Signaling Type** field is blank. The **Multifrequency Signaling** field must be enabled on the **System-Parameters Customer-Options** screen in order for you to enter **mf** here. You may not enter **mf** if the **Used for DCS** field (field descriptions for page 2) is **y**. For pulsed and continuous E&M signaling in Brazil and for discontinuous E&M signaling in Hungary, use **mf**.

Incoming Rotary Timeout (sec)

Call setup at central offices that still use older switching equipment, such as step-by-step technology, is considerably longer then at central offices with more modern media servers. If you're receiving digits with incoming calls from a central office that uses less efficient switching technology, your media server or switch needs to allow more time to ensure it receives all the incoming digits. When the **Incoming Dial Type** field is **rotary**, use this field to set the maximum time your end will wait to receive all incoming digits from the far-end switch.

Valid entries	Usage
5 to **99** or blank	If the system is connected to a step-by-step central office, or any CO using older switching technology, enter at least **18** seconds; if not, enter at least **5** seconds.

ITC

The Generalized Route Selection feature, part of the automatic routing technology used in Avaya Communication Manager, compares the line coding of available digital facilities and selects appropriate routes for voice and data calls. The **Information Transfer Capability** field appears when the **Comm Type** field is **data**, **avd**, or **rbavd** and the **BCC** field is not **0**.

Valid entries	Usage
rest(ricted)	Restricted trunks use **ami-basic** or **ami-zcs** line coding and can carry only restricted calls.
unre(stricted)	Unrestricted trunks use **b8zs**, **hdb3,** or **cmi** line coding and can carry restricted or unrestricted calls. A trunk group with an unrestricted ITC may have only unrestricted trunks as members.

Line Length

This field appears only when the **Group Type** field is **tie** and the **Trunk Signaling Type** field is **tge**, **tgi**, or **tgu**.

Valid entries	Usage
short	
long	

Note:

Unless one or more trunk members are administered, the administered value is not saved when you submit the screen (press **Enter**).

Night Service

This field sets the destination to which incoming calls go when **Night Service** is in operation. If a **Night** field on the **Group Member Assignments** page is administered with a different destination, that entry will override the group destination for that trunk. CPE, DID, and DIOD trunk groups do not support night service.

Tip:

Whenever possible, use a night service destination on your switch: otherwise some features won't work correctly, even over a DCS network.

Valid entries	Usage
blank	Leave this field blank if the **Trunk Type (in/out)** field is not auto/….
An extension number (can be a VDN)	Enter the extension of your night service destination. **Note:** If entering a Multi-Location Dial Plan shortened extension, note the following: When entering a Multi-Location Dial Plan shortened extension in a field designed for announcement extensions, certain administration end validations that are normally performed on announcement extensions are not done, and resultant warnings or submittal denials do not occur. The shortened extensions also do not appear in any display or list that shows announcement extensions. Extra care should be taken to administer the correct type of announcement for the application when assigning shortened extensions.
attd	Calls go to the attendant and are recorded as Listed Directory Number (LDN) calls on call detail records.

Number of Members

This field displays the number of trunk groups assigned to this trunk group.

Numbering Format

This field appears if the **Send Calling Number** field is **y** or **r** or the **Send Connected Number** field is **y** or **r**. This specifies the encoding of Numbering Plan Indicator for identification purposes in the Calling Number and/or Connected Number IEs, and in the QSIG Party Number. Valid entries are **public**, **unknown**, **private**, and **unk-pvt**. **Public** indicates that the number plan according to CCITT Recommendation E.164 is used and that the **Type of Number** is national. **Unknown** indicates the **Numbering Plan Indicator** is unknown and that the **Type of Number** is unknown. **Private** indicates the **Numbering Plan Indicator** is PNP and the **Type of Number** is determined from the **ISDN Private-Numbering** screen. An entry of **unk-pvt** also determines the **Type of Number** from the **ISDN Private-Numbering** screen, but the **Numbering Plan Indicator** is unknown.

Outgoing Dial Type

This field sets the method used to transmit digits for an outgoing call. Usually, you should match what your central office provides. See "Trunks and Trunk Groups" in *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205, for more information. This field appears for Access, APLT, CO, DIOD, DMI-BOS, FX, RLT, and WATS trunk groups. It also appears for Tie trunk groups when the **Trunk Signaling Type** field is blank, **cont**, or **dis**.

DIOD trunks support pulsed and continuous E&M signaling in Brazil and discontinuous E&M signaling in Hungary.

Valid entries	Usage
tone	Enter **tone** to use Dual Tone Multifrequency (DTMF) addressing, also known as "touch tone" in the U.S. Entering **tone** actually allows the trunk group to support both DTMF and rotary signals. For pulsed and continuous E&M signaling in Brazil and for discontinuous E&M signaling in Hungary, use **tone** or **mf**.
rotary	Enter **rotary** if you only want to allow the dial pulse addressing method used by non-touch tone phones. If you have a full touch tone system internally and a connection to a central office that only supports rotary dialing, for example, it would be appropriate to enter **rotary**.
r1mf	Enter **r1mf** for CAMA trunk groups. It is the only outgoing dial type allowed on CAMA trunk groups. Enter **r1mf** to allow Russian MF Packet Signaling on outgoing trunks. Russian MF Packet Signaling carries calling party number and dialed number information. **Group type** field must be set to **co**.

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Valid entries	Usage
mf	Enter **mf** if the **Trunk Signaling Type** field is blank. The **Multifrequency Signaling** field must be enabled on the **System-Parameters Customer-Options** screen in order for you to enter **mf** here. You may not enter mf if the **Used for DCS** field (Field descriptions for page 2) is **y**. For pulsed and continuous E&M signaling in Brazil and for discontinuous E&M signaling in Hungary, use **tone** or **mf**.
automatic	Enter **automatic** for tie trunks if the **Trunk Signaling Type** field is blank. This provides "cut-through" operation to outgoing callers who dial a trunk access code, connecting them directly to central office dial tone and bypassing any toll restrictions administered on Communication Manager.

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Outgoing Display

This field allows display phones to show the name and number of the trunk group used for an outgoing call before the call is connected. This information may be useful to you when you're trying to diagnose trunking problems.

Valid entries	Usage
y	Displays the trunk group name and number.
n	Displays the digits the caller dials.

Prefix-1

Use this field for outgoing and two-way trunk groups handling long distance service. This field appears for CO, FX, and DIOD trunk groups.

Valid entries	Usage
y/n	Enter **y** to add the prefix "1" to the beginning of the digit string for outgoing calls. Do not enter **y** for trunk groups in AAR or ARS route patterns.

Protocol Type

This field specifies the type of line signaling protocol used for DID and DIOD trunk groups. This field appears when the **Country** field is **15** and is used only by trunk group members administered on a TN2199 or TN464D vintage 3 or later circuit pack.

Valid entries	Usage
inloc (Incoming local)	Enter the protocol the central office is using for this trunk group. Only the inloc protocol provides ANI.
intol (Incoming toll)	

Queue Length

Outgoing calls can wait in a queue, in the order in which they were made, when all trunks in a trunk group are busy. If you enter 0, callers receive a busy signal when no trunks are available. If you enter a higher number, a caller hears confirmation tone when no trunk is available for the outgoing call. The caller can then hang up and wait: when a trunk becomes available, Communication Manager will call the extension that placed the original call. The caller will hear 3 short, quick rings. The caller doesn't need to do anything but pick up the handset and wait: Communication Manager remembers the number the caller dialed and automatically completes the call.

This field appears when the **Direction** field is **outgoing** or **two-way**.

Valid entries	Usage
0 to **100**	Enter the number of outgoing calls that you want to be held waiting when all trunks are busy.
0	Enter **0** for DCS trunks.

Receive Answer Supervision

Use this field to specify whether the network provides answer supervision for a trunk group.

For Outbound Call Management applications, set this field to **y** for trunks supporting network answer supervision. For trunks that do not receive a real answer, this field determines when the CallVisor Adjunct-Switch Application Interface (ASAI) connect event is sent.

Valid entries	Usage
y	Enter **y** if the network provides answer supervision. Set the **Answer Supervision Timeout** field to **0**.
n	Enter **n** if the network does not provide answer supervision, and set the **Answer Supervision Timeout** field. Also enter **n** for incoming trunk groups.

Note:

When you set this field to **y**, the **Outgoing End of Dial (sec)** field is not displayed. The firmware timeout on circuit packs controlled by the **Outgoing End of Dial (sec)** field is automatically set to **0**.

Receive Release Ack

This field appears when the **Trunk Signaling Type** field is **cont** or **dis** and only applies to TN2140 ports (used for Italian and Hungarian tie trunks).

Valid entries	Usage
y/n	Enter **y** if Communication Manager will receive a release acknowledgment in response to a forward or backward release signal.

SCCAN

This field appears when the **Group Type** field is **sip** and **Enhanced EC500** on the **System Parameters - Customer Options** screen is set to **y**.

Valid entries	Usage
y/n	Enter **y** to indicate that this trunk group provides support for incoming SCCAN calls. Default is **n**.

Send Answer Supervision

This field appears when the **Trunk Signaling Type** field is **cont** or **dis** and only applies to TN2140 ports.

Valid entries	Usage
y/n	Enter **y** to make Communication Manager signal the calling media server or switch when an incoming call is answered. You can only set this field to **y** if the **Direction** field is **incoming** or **two-way**.

Send Release Ack

This field appears when the **Trunk Signaling Type** field is **cont** or **dis** and only applies to TN2140 ports (used for Italian and Hungarian tie trunks).

Valid entries	Usage
y/n	Enter **y** to send a release acknowledgment in response to a forward or backward release signal.

Service Type

Indicates the service for which this trunk group is dedicated. The following table provides a listing of predefined entries. In addition to the Services/Features listed in this table, any user-defined Facility Type of **0** (feature) or **1** (service) on the **Network Facilities** screen is allowed.

As many as 10 (for Avaya DEFINITY Server SI or CSI) or 200 (for DEFINITY Server R) ISDN trunk groups can have this field administered as **cbc**.

Valid entries	Usage
access	A tie trunk giving access to an Electronic Tandem Network.
accunet	ACCUNET Switched Digital Service — part of ACI (AT&T Communications ISDN) phase 2.
cbc	Call-by-Call service — provides different dial plans for different services on an ISDN trunk group. Indicates this trunk group is used by the Call-By-Call Service Selection feature.
dmi-mos	Digital multiplexed interface — message oriented signaling.

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Valid entries	Usage
i800	International 800 Service — allows a subscriber to receive international calls without a charge to the call originating party.
inwats	INWATS — provides OUTWATS-like pricing and service for incoming calls.
lds	Long-Distance Service — part of ACI (AT&T Communications ISDN) phase 2.
megacom	MEGACOM Service — an AT&T communications service that provides unbanded long-distance services using special access (switch to 4ESS switch) from an AT&T communications node.
mega800	MEGACOM 800 Service — an AT&T communications service that provides unbanded 800 service using special access (4ESS switch to switch) from an AT&T communications node.
multiquest	AT&T MULTIQUEST Telecommunications Service — dial 700 service. A terminating-user's service that supports interactive voice service between callers at switched-access locations and service provides directly connected to the AT&T Switched Network (ASN).
operator	Network Operator — provides access to the network operator.
outwats-bnd	OUTWATS Band — WATS is a voice-grade service providing both voice and low speed data transmission capabilities from the user location to defined service areas referred to as bands; the widest band is 5.
public-ntwrk	Public network calls — It is the equivalent of CO (outgoing), DID, or DIOD trunk groups. If Service Type is public-ntwrk, Dial Access can be set to **y**.
sddn	Software Defined Data Network — provides a virtual private line connectivity via the AT&T switched network (4ESS switches). Services include voice, data, and video applications. These services complement the SDN service. Do not use for DCS with Rerouting.
sdn	Software Defined Network (SDN) — an AT&T communications offering that provides a virtual private network using the public switched network. SDN can carry voice and data between customer locations as well as off-net locations.
sub-operator	Presubscribed Common Carrier Operator — provides access to the presubscribed common carrier operator.
tandem	Tandem tie trunks integral to an ETN

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Valid entries	Usage
tie	Tie trunks — general purpose
wats-max-bnd	Maximum Banded Wats — a WATS-like offering for which a user's calls are billed at the highest WATS band subscribed to by users.

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Signaling Group

This field displays the group number assigned when the group was added.

Sig Bit Inversion

When transmission facilities use bit-oriented signaling (such as CAS), 2 bits are used to transmit seizure and release signals for calls. Called the A-bit and the B-bit, their meaning can vary. For example, in the A-bit a "1" might mean on-hook and a "0" might mean off-hook. The entry in the **Country Protocol** field on the **DS1 Circuit Pack** screen sets the default meaning of these bits.

For trunk ports on TN2242 and TN464B and later circuit packs, this field allows you to invert the A- and B-bits as necessary so that the far-end server/switch can understand seizure and release signals from Communication Manager. If the far-end media server or switch, such as a central office, on this trunk group interprets the A- and B-bits differently from the default, you may need to invert one or both bits — to change "1" to "0" and vice-versa in the A-bit, for example.

Valid entries	Usage
A	For the TN464B and later circuit packs, indicate which bits, if any, should be inverted.
B	
A&B	
none	
A and **none**	For the Japanese 2Mbit trunk circuit pack, indicate which bits, if any, should be inverted.

Supplementary Service Protocol

Appears only when trunk group Type is ISDN.

Valid entries	Usage
a	Allows ASAI Flexible Billing. AT&T, Bellcore, Nortel. When the **Country Code** field on the **DS1** screen is **1A**, SSA selects AT&T custom supplementary services. When the **Country Code** field on the **DS1** screen is **1B**, SSA selects Bellcore Supplementary Services. When the **Country Code** field on the **DS1** screen is **1C**, SSA selects Nortel Proprietary Supplementary Services.
b	QSIG; also used for SBS signaling trunk groups when full QSIG functionality is needed.
c	ETSI Use c protocol for Network Call Deflection.
d	ECMA QSIG
e	Allows ASAI Flexible Billing. Allows DCS with rerouting. **DCS with Rerouting** must be **y**, and the **Used for DCS** field on the **Trunk Group** screen must be **y**.
f	Feature Plus
g	ANSI. Available only if, on the **System-Parameters Customer-Options** screen, the **ISDN-PRI** or **ISDN-BRI** field is **y** or the **Used for DCS** field is **y**. Use **g** protocol for Network Call Transfer.

Synchronization

This field determines whether the trunk group will use synchronous or asynchronous communications. This field appears if:

- the **Group Type** field is **dmi-bos** or **isdn**, or
- the **Group Type** field is **access**, **co**, **fx**, **tandem**, **tie**, or **wats** and the **Comm Type** field is **avd** or **rbavd**, or
- the **Group Type** field is **access**, **co**, **fx**, **tandem**, **tie**, or **wats**, the **Comm Type** field is **data**, and the **ISDN-PRI** field or the **ISDN-BRI Trunks** field is **y** on the **System-Parameters Customer-Options** screen.

Valid entries	Usage
async	Do not change this field without the assistance of Avaya or your network service provider.
sync	

TAC

Enter the trunk access code (TAC) that must be dialed to access the trunk group. A different TAC must be assigned to each trunk group. CDR reports use the TAC to identify each trunk group.

Valid entries	Usage
1 to 4 digit number	Enter any number that fits the format for trunk access codes or dial access codes defined in your dial plan.
*****, #	* and # may be used as the first character in a TAC.

TN

Valid entries	Usage
1 to **100** (DEFINITY R, S8700 or S8710 IP-Connect)	Enter a Tenant Partition number to assign this trunk group to the partition.

Tip:

Double-check your entry. If you accidentally enter an unassigned tenant partition number, the system will accept the entry but no calls will go to the trunk group.

Toll Restricted

Valid entries	Usage
y	Enter **y** to prevent toll-restricted users from using a trunk access code to make restricted outgoing calls over this trunk group.
n	Enter **n** if the field is automatic or if you don't want to restrict access.

Tip:

To find out what kind of line coding a trunk group member uses, check the **Line Coding** field on the **DS1 Circuit Pack** screen for the DS1 port to which the member is assigned.

Trunk Flash

Trunk Flash enables multifunction telephones on Avaya Communication Manager to access central office customized services that are provided by servers at the far-end or Central Office (CO). These central office customized services are electronic features, such as conference and transfer, that are accessed by a sequence of flash signal and dial signals from the Communication Manager telephone on an active trunk call.

Valid entries	Usage
y/n	Enter **y** to allow trunk flash.

Trunk Gain

This field specifies the amplification applied to the trunks in this group. With the values of the **Trunk Termination** and **Country** fields, the value in this field also determines the input and trans-hybrid balance impedance for TN465B, TN2146, TN2147, and TN2184 ports. All other CO and DID circuit packs are set automatically to high

Valid entries	Usage
high	Enter **high** if users complain of low volume.
low	Enter **low** if users complain of squeal or feedback.

Trunk Hunt

Avaya Communication Manager performs a trunk hunt when searching for available channels within a facility in an ISDN trunk group. With both **ascend** and **descend**, all trunks within an ISDN trunk group are selected based on this field and without regard to the order in which trunks are administered within the trunk group. When using ISDN-BRI interfaces, only **cyclical** is allowed

Valid entries	Usage
ascend	Enter to enable a linear trunk hunt search from the lowest to highest numbered channels.
cyclical	Enter to enable a circular trunk hunt based on the sequence the trunks were administered within the trunk group.
descend	Enter for a linear trunk hunt search from the highest to lowest numbered channels.

Note:

The cyclical option cannot be set if the trunk group using ISDN-PRI interfaces is to be used for Wideband operations (the **Wideband Support** field set to **y**).

The search can be administered per ISDN-PRI trunk group, but it infers the direction of search within all ISDN-PRI facilities (or portions of those facilities) administered within the trunk group.

Trunk Signaling Type

This field controls the signaling used by members in private network trunk groups, mainly in Italy, Brazil, and Hungary. This field also controls the signaling used by members in public network digital trunk groups. This field displays if the **Group Type** field is **access**, **aplt**, **rlt**, **tandem**, or **tie**. Entries in this field affect which timers appear on the **Administrable Timers** page.

Valid entries	Usage
cont (continuous)	E&M trunks in Italy, Brazil, and Hungary can use either continuous or discontinuous signaling. Each entry specifies a set of signals and available timers used in the process of setting up and releasing connections. The type of signaling you select on Communication Manager must match the signaling type administered on the far-end media server or switch. Use these values only when all trunk group members are assigned to ports on a TN464F, TN2464, or TN2140 circuit pack. Entering one of these values causes the **Send Release Ack**, **Receive Release Ack**, and **Send Answer Supervision** fields to appear. See Trunk Type (in/out) on page 1572 for more information.
dis (discontinuous)	
Use the following entries for tie trunks in Main-Satellite/Tributary networks. Each entry defines a function of the trunk group in the network. Use these values only when all trunk group members are assigned to a TN497 circuit pack.	
tgu (for outgoing trunks)	Enter **tgu** at the main server running Communication Manager to administer a tie trunk group connected to a satellite server. (This same group should be administered as **tge** at the satellite.)
tge (for incoming trunks)	Enter **tge** at a satellite server to administer a tie trunk group connected to the main server running Communication Manager. (This same group should be administered as **tgu** at the main server.)
tgi (for internal trunks)	Enter **tgi** at to administer a two-way tie trunk group between 2 satellites or between the main server and a satellite. (This trunk group should be administered as **tgi** on both servers.)

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Valid entries	Usage
DIOD trunks support pulsed and continuous E&M signaling in Brazil and discontinuous E&M signaling in Hungary. Use the following entries for DIOD trunks. Use these values only when all trunk group members are assigned to a TN464F (or later version) or TN2464 circuit pack.	
cont **pulsed** **discont**	Enter **cont** for continuous E&M signaling. Enter **pulsed** for pulsed E&M signaling. Leave blank forR2 signaling. Hungary uses discontinuous E&M signaling when this field is **dis**. Brazil E&M trunks use continuous and pulsed E&M.
	2 of 2

Trunk Termination

This field adjusts the impedance of the trunk group for optimal transmission quality. Check with your service provider if you're not sure of the distance to your central office.

Valid entries	Usage
600ohm	Enter **600ohm** when the distance to the central office or the media server or switch at the other end of the trunk is less than 3,000 feet.
rc	Enter **rc** when the distance to the central office or the media server or switch at the other end of the trunk is more than 3,000 feet.

Trunk Type

Use this field to control the seizure and start-dial signaling used on this trunk group. Entries in this field vary according to the function of the trunk group and *must* match the corresponding setting on the far-end server or switch. This field appears for CO, DID, FX, and WATS trunk groups.

Procedures in Tips for Working with Trunk Groups on page 423 give specific suggestions for signaling to use with different types of trunk groups.

Valid entries	Usage
ground-start	Use ground-start signaling for two-way trunks whenever possible: ground-start signaling avoids glare and provides answer supervision from the far end.
loop-start	In general, try to use loop-start signaling only for one-way trunks. Loop-start signaling is susceptible to glare and does not provide answer supervision.
auto/auto **auto/delay** **auto/immed** **auto/wink**	The term before the slash tells Communication Manager how and when it will receive incoming digits. The term after the slash tells Communication Manager how and when it should send outgoing digits. • auto — Used for immediate connection to a single preset destination (incoming CO trunks, for example). No digits are sent, because all calls terminate at the same place. • delay — The sending server running Communication Manager does not send digits until it receives a delay dial signal (an off-hook signal followed by an on-hook signal) from the far-end server or switch, indicating that it's ready to receive the digits. • wink — The sending server running Communication Manager does not send digits until it receives a a wink start (momentary off-hook) signal from the far-end server or switch, indicating that it's ready to receive the digits. • immed — The sending server running Communication Manager sends digits without waiting for a signal from the far-end server or switch.
2-wire-ac **2-wire-dc** **3-wire**	These entries are used with CO trunks in Russia: enter the type of connection to your central office. Check with you network service provider if you don't know what type of connection they're using. To use these entries, the **Country** field must be **15** and the CO trunks must use ports on a TN2199 circuit board.

Trunk Type (in/out)

Use this field to control the seizure and start-dial signaling used on this trunk group. The setting of the **Trunk Signaling Type** field can affect the entries allowed in this field. In addition, settings may differ for incoming and outgoing trunks.

Valid entries	Usage
auto **cont** **delay** **disc** **immed** **wink**	There are numerous valid entries for this field: use the online help in Communication Manager to view all the possible combinations. Here are what the elements used in those combinations mean: • auto — Used for immediate connection to a single preset destination (incoming CO trunks, for example). No digits are sent, because all calls terminate at the same place. • cont — Continuous signaling is used with Italian E&M tie trunks. The server/switch seizes a trunk by sending a continuous seizure signal for at least the duration specified by the Incoming Seizure Timer. See Trunk Hunt on page 1568 for more information. • delay — The sending switch does not send digits until it receives a delay dial signal (an off-hook signal followed by an on-hook signal) from the far-end switch, indicating that it's ready to receive the digits. • disc —Discontinuous signaling is used with Italian tie trunks that use E&M signaling. The media server or switch can seize a trunk by sending a single, short signal for the duration specified by the **Normal Outgoing Seize Send** field. However, with the Three-Way Seizure option, the calling end can also send routing information to the called end by sending one or a series of brief seizure signals. • wink — The sending server or switch does not send digits until it receives a a wink start (momentary off-hook) signal from the far-end server or switch, indicating that it's ready to receive the digits. • immed — The sending server or switch sends digits without waiting for a signal from the far-end server or switch.
2-wire-ac **2-wire-dc** **3-wire**	These entries are used with CO trunks in Russia: enter the type of connection to your central office. Check with you network service provider if you don't know what type of connection they're using. To use these entries, the **Country** field must be **15** and the CO trunks must use ports on a TN2199 circuit board.

Tip:

When incoming trunks use the setting immed/immed, the far-end media server or switch seizes the trunk and sends digits without waiting for acknowledgment from the receiving end. When traffic is heavy, the receiving server or switch may not immediately attach a Touch Tone Receiver to a call and therefore lose digits. Use wink-start trunks or increase the dial-guard timer value on the far-end server or switch to avoid this problem.

Note:

The value in this field affects the appearance of the **Incoming Partial Dial (sec)** field on the **Administrable Timer** page.

Version

Use this field to adjust the signaling on multi-country CO trunk circuit packs. Entries in this field adjust signaling characteristics on these circuit packs to match the signaling characteristics of the public network in a specific country. The field appears only for CO, FX, and WATS trunk groups when the **Country** field is **5**, **16**, or **23**.

Valid entries	Usage
If the **Country** field is **5**, the **Version** field only controls TN2147 ports.	
a	Enter **a** to use standard signaling for the Netherlands public network.
b	Enter **b** to use country 1 (U.S.) signaling. The value **b** is appropriate if Communication Manager is connected to a central office using an Ericcson AXE-10 switch.
If the **Country** field is **16** or **23**, the **Version** field sets the input impedance value and only controls TN465C (vintage 2 or later) ports.	
a	Enter **a** to set input impedance to 600 Ohms.
b	Enter **b** to set input impedance to 900 Ohms. The value **b** is appropriate in Brazil.

Wink Timer (msec)

This field allows you to reduce the risk of glare by controlling part of call setup. Requirements for the United States domestic network specify that the wink signal for wink-start trunks must begin within 5 seconds after a trunk is seized. For trunks with a delay-dial start, the wink must not last longer than 5 seconds. While some circuit packs are hard-coded to allow the full 5 seconds in both cases, other circuit packs allow you reduce the allowed start time and duration, thus reducing the window in which glare could occur.

Unlike other fields on this screen, the **Wink Timer** field therefore controls 2 different variables. What your entry does depends on the outgoing value in the **Trunk Type (in/out)** field.

Setting of the Trunk Type (in/out) field	What the Wink Timer field sets
…/wink	Maximum duration of the wink signal (wait-for-wink-to-end)
…/delay	Maximum interval after trunk seizure for the wink to begin (wait-for-wink-to-start)

This field appears when the "out" side of the entry in the **Trunk Type (in/out)** field is **…/wink** or **…/delay** and the **Group Type** is **tie**, **access**, **aplt**, **dmi-bos**, **rlt**, or **tandem**. The setting in this field only affects trunks administered to ports on TN760C (vintage 4 or later), TN767, TN464C (or later), and TN2242 circuit packs. If the trunk group also contains trunks assigned to ports on other types of circuit packs, those trunks are unaffected.

Valid entries	Usage
300 to **5000** in increments of 50	In general, Avaya recommends that you not change this field. If you do, remember that timing on your server running Communication Manager must be compatible with the timing on the far-end media server or switch.

Field descriptions for page 2

The figure below is only an example, and is intended to show most of the fields that might appear on page 2 of the **Trunk Group** screen. This example may might not show all fields, or might show fields that normally do not appear together; it is not intended to reflect a specific trunk group type. Your own screen mary vary from this example according to specific field and system settings. The list of field descriptions that follow the figure is in alphabetical order for quick reference. This list is intended to be comprehensive, and may include information on fields that are not shown in the example. The field descriptions identify fields that are specific to particular trunk group types.

Figure 538: Trunk Group screen (page 2)

```
add trunk-group next                                          Page 2 of x
                              TRUNK FEATURES
           ACA Assignment? _           Measured: ____     Wideband Support? _
 Long Holding Time(hours): _                             Maintenance Tests? _
  Short Holding Time(sec): _  Data Restriction? _    NCA-TSC Trunk Member: _
  Short Holding Threshold: __         Send Name: _     Send Calling Number: _
             Used for DCS? _

Suppress # Outpulsing? _  Numbering Format: _______
Outgoing Channel ID Encoding: _________     UUI IE Treatment: ______
                                       Maximum Size of UUI IE Contents: ___
                                            Replace Restricted Numbers? _
                                           Replace Unavailable Numbers? _
                                                 Send Connected Number: _
                                             Hold/Unhold Notifications? _

        Send UUI IE? _
          Send UCID? _                BRS Reply-best DISC Cause Value: __
                                                Ds1 Echo Cancellation? _

                                      US NI Delayed Calling Name Update? _

                          Network (Japan) Needs Connect Before Disconnect? _

Time (sec) to Drop Call on No Answer:_
Outgoing ANI:_
R2 MFC Signaling:_

DSN Term? n       Precedence Incoming _______   Precedence Outgoing _______
```

CAUTION:

Customers: Do not change fields on this page without assistance from Avaya or your network service provider.

Abandoned Call Search

Use this field when the **Trunk Type** field is **ground-start**. Abandoned Call Search is designed to work with analog ground-start CO trunks that do not provide disconnect supervision. Your central office must support Abandoned Call Search for the feature to work properly. If your central office provides disconnect supervision, you do not need to use the Abandoned Call Search feature.

Valid entries	Usage
y/n	Enter **y** if this trunk group will conduct an Abandoned Call Search to identify ghost calls.

ACA Assignment

Valid entries	Usage
y/n	Enter **y** if you want Automatic Circuit Assurance (ACA) measurements to be taken for this trunk group. If **y** is entered, complete the **Long Holding Time**, **Short Holding Time**, and **Short Holding Threshold** fields.

Charge Conversion

Avaya Communication Manager multiplies the number of charge units by the value of this field and displays it as a currency amount. If there is no value in this field, Avaya Communication Manager displays the number of charge units without converting it to currency. This field appears for CO, DIOD, FX, and WATS trunk groups when the **Direction** field is **outgoing** or **two-way**. For ISDN trunk groups, it appears when the **Charge Advice** field is *not* **none**.

Valid entries	Usage
1 to **64**, **500**	Enter the value of a charge unit in terms of the currency you use.

Charge Type

Entries in this field are text strings you use to describe charges related to a telephone call. This field appears for CO, DIOD, FX, and WATS trunk groups when the **Direction** field is **outgoing** or **two-way**. For ISDN trunk groups, it appears when the **Charge Advice** field is *not* **none**.

Valid entries	Usage
1 to 7 characters (embedded spaces count as characters) or blank	Enter the words or characters you want to appear on telephone displays after the charge amount. Most likely you will use either the currency symbol or the charge type, bur not both.

Connected to CO

This field appears when the **Group Type** field is **tie**.

Valid entries	Usage
y/n	Enter **y** to allow overlap sending to a Central Office.

Currency Symbol

This field appears for CO, DIOD, FX, and WATS trunk groups when the **Direction** field is **outgoing** or **two-way**. For ISDN trunk groups, it appears when the **Charge Advice** field is *not* **none**.

Valid entries	Usage
1 to 3 characters (leading and embedded spaces count as characters) or blank	Enter the symbol you want to appear on telephone displays before the charge amount.

Data Restriction

If **y**, whisper page is denied on this trunk.

Valid entries	Usage
y/n	Enter **y** to prevent features from generating tones on a data call that would cause erroneous data transmission.

Decimal Point

This field appears for CO, DIOD, FX, and WATS trunk groups when the **Direction** field is **outgoing** or **two-way**. For ISDN trunk groups, it appears when the **Charge Advice** field is *not* **none**.

Valid entries	Usage
comma	Chose the appropriate representation for a decimal point as it will appear on telephone displays. Entering **comma** or **period** in this field divides the charge value by 100.
period	
none	

DS1 Echo Cancellation

Reduces voice call echo.

Note:

Changes to the **DS1 Echo Cancellation** field do not take effect until one of the following occurs:

- Port is busied-out/released
- Trunk group is busied-out/released
- SAT command test trunk group is performed
- Periodic maintenance runs

Valid entries	Usage
y/n	Enter **y** to allow echo cancellation on a per port basis.

DSN Term

Valid entries	Usage
y/n	Use the **DSN Term** field to identify the trunk group as a DSN termination telephone. The default is **n**.

Glare Handling

This field determines what Communication Manager will do when glare occurs. This field appears when the **Direction** field is **two-way** and the outgoing side of the **Trunk Type** field is either **…/wink** or **…/delay**.

If you enter **control** or **backoff**, and ports for the trunk group are not capable of detecting glare, warnings are generated. The following circuit packs can detect glare: TN767 (all releases), TN760C (or later releases), and TN464C (or later releases).

Valid entries	Usage
control	Communication Manager will seize the trunk and proceed with call setup. The other switch will find another trunk.
backoff	The other server or switch will seize the trunk and proceed with call setup. Your server/switch will find another trunk.
none	

Hold/Unhold Notifications

Appears only when the **Group Type** field is **isdn**. Use this field to indicate whether or not hold/unhold messages are sent over the isdn trunk when a user places a call on hold/unhold.

Valid entries	Usage
y/n	Enter **y** to enable sending of hold/unhold messages over this isdn trunk. Default is **y**.

Incoming Tone (DTMF) ANI

This field appears only when the **Incoming Dial Type** field is **tone**. Digits received through Automatic Number Identification (ANI) are printed on a CDR record, passed to the INTUITY AUDIX and ASAI interfaces, and displayed on the telephone (and on tandem calls if the outgoing trunk requires ANI). Then the digits are sent to the outgoing trunk.

Valid entries	Usage
ANI*DNIS	If 555-3800 calls extension 81120, the trunk group receives *55538000*81120*. The telephone displays Call from 555-3800.
ANI*DNIS*	If 555-3800 calls extension 81120, the trunk group receives 55538000*81120*. The telephone displays Call from 555-3800.
no	

Internal Alert

Valid entries	Usage
y/n	Enter **y** if internal ringing and coverage will be used for incoming calls.

Long Holding Time (hours)

This field appears only when the **ACA Assignment** field is **y**.

Valid entries	Usage
0 to **10**	Enter the length of time (in hours) that the system will consider as being a long holding time. If you enter **0**, the system will not consider long holding calls.

Maintenance Tests

Appears when the **Group Type** field is **aplt** or **tie**.

Valid entries	Usage
y/n	Enter **y** if hourly maintenance tests will be made on this trunk group. One or more trunk members must be administered for this entry to be saved.

Measured

Indicates if the system will transmit data for this trunk group to the Call Management System (CMS). You cannot use **internal** and **both** unless either the **BCMS (Basic)** or the **VuStats** field is **y** on the **System-Parameters Customer-Options** screen. If the **ATM** field is set to **y** on the **System-Parameters Customer-Option** screen, this field accepts only **internal** or **none**. If this field contains a value other than **internal** or **none** when **ATM** is **y**, **none** appears.

Valid entries	Usage
internal	Enter **internal** if the data can be sent to the Basic Call Management System (BCMS), the VuStats data display, or both.
external	Enter **external** to send the data to the CMS.
both	Enter **both** to collect data internally and to send it to the CMS.
none	Enter **none** if trunk group measurement reports are not required.

MF Tariff Free

This field appears for Access, APLT, DID, DIOD, DMI-BOS, and Tandem trunk groups when the **Incoming Dial Type** field is **mf** or the **Group Type** field is **tie**, the **Trunk Signaling Type** field is blank, **cont**, or **dis**, and the **Incoming Dial Type** field is **mf**.

Valid entries	Usage
y/n	Enter **y** to make Communication Manager generate an MFC Tariff-Free Backward Signal (administered on the **Multifrequency-Signaling-Related-System-Parameters** screen) during call setup instead of the "free" signal. This aids CO billing.

Network Call Redirection

For more information, see *Avaya MultiVantage Call Center Software Release 11 — Network Call Redirection*.

Valid entries	Usage
deflect	Use to allow Network Call Deflection.
ANSI-transfer	Use to allow Network Call Transfer for MCI DEX 600 ISDN trunks.
Nortel-transfer	Use to allow Network Call Transfer for MCI DMS 250 switches.
telcordia-tbct	Use to allow Network Call Transfer for Lucent 5ESS or Nortel DMS100 switches.

Numbering Format

This field appears if the **Send Calling Number** field is **y** or **r** or the **Send Connected Number** field is **y** or **r**. This specifies the encoding of Numbering Plan Indicator for identification purposes in the Calling Number and/or Connected Number IEs, and in the QSIG Party Number. Valid entries are **public**, **unknown**, **private**, and **unk-pvt**. **Public** indicates that the number plan according to CCITT Recommendation E.164 is used and that the **Type of Number** is national. **Unknown** indicates the **Numbering Plan Indicator** is unknown and that the **Type of Number** is unknown. **Private** indicates the **Numbering Plan Indicator** is PNP and the **Type of Number** is determined from the **ISDN Private-Numbering** screen. An entry of **unk-pvt** also determines the **Type of Number** from the **ISDN Private-Numbering** screen, but the **Numbering Plan Indicator** is unknown.

Outgoing ANI

If this trunk group is used for an outgoing call with ANI, the entry in this field overrides the normal ANI. The ANI is sent exactly as administered, except for the normal truncation to 7 digits for Russian ANI. This ANI override works both for calls originated in Avaya Communication Manager and calls tandemed through it. This field appears for CO, DIOD, FX, and WATS trunk groups.

Valid entries	Usage
1 to 15 digits	Enter the digit string to be sent in place of normal ANI.
blank	Leave this field blank to allow ANI to work normally.

Path Replacement Method

Appears when the following fields are set on the **Trunk Group** screen: trunk **Group Type** is **ISDN**, **Supplementary Service Protocol** is **b** or **e**, the **Path Replacement with Retention** is **n**, and the **Supplementary Services with Rerouting** field or the **DCS with Rerouting** field on the **System-Parameters Customer-Options** screen is **y**.

Valid entries	Usage
always	Use any QSIG (SSB) trunk group as the replacement trunk group. A new call is always originated, even when the original trunk group is determined to be the replacement trunk group.
BR (better route)	Route pattern preferences help determine trunk group path replacement. The original trunk group is retained if the **Path Replacement with Retention** field is **y**. Path replacement fails (and the original trunk group is retained) if the **Path Replacement with Retention** field is **n**.

Path Replacement with Retention

Appears when the following fields are set on the **Trunk Group** screen: trunk **Group Type** is **ISDN**, **Supplementary Service Protocol** is **b** or **e**, and the **Supplementary Services with Rerouting** field or the **DCS with Rerouting** field on the **System-Parameters Customer-Options** screen is **y**.

Valid entries	Usage
y/n	Enter **y** to retain the original trunk group. Set to **n** to allow path replacement according to setting on the **Path Replacement Method** field.

PBX ID

Appears when the **Used for DCS** field is **y**. This field identifies the remote switch in the network with which the trunk will communicate on a DCS signaling link.

Valid entries	Usage
1 to **63** or blank	Enter the ID of the switch at the other end of this trunk.

Per Call CPN Blocking Code

For Access, APLT, CO, DIOD, FX, tandem, tie, and WATS trunk groups only.

Valid entries	Usage
1 to 4 digit number or blank	
*****, #	May be used as the first digit

Per Call CPN Unblocking Code

For access, APLT, CO, DIOD, FX, tandem, tie, and WATS trunk groups only.

Valid entries	Usage
1 to 4 digit number or blank	
*****, #	May be used as the first digit

Precedence Incoming

The **Precedence Incoming** field defines whether the precedence level for dual-tone multifrequency (DTMF) or tone trunks is received as digits (rotary pulses) or as DTMF signals (touchtones). Appears when the **DSN Term** field is **y** and **Group Type** is **tie**.

Valid entries	Usage
digit	Precedence level is received as digits (rotary pulses).
dtmf (a-d)	Precedence level is received as DTMF signals (touchtones).

Precedence Outgoing

The **Precedence Outgoing** field defines whether the precedence level for dual-tone multifrequency (DTMF) or tone trunks is sent as digits (rotary pulses) or as DTMF signals (touchtones). Appears when the **DSN Term** field is **y** and **Group Type** is **tie**.

Valid entries	Usage
digit	Precedence level is sent as digits (rotary pulses).
dtmf (a-d)	Precedence level is sent as DTMF signals (touchtones).

R2 MFC Signaling

Appears when the **Outgoing Dial Type** field on page 1 of the **Trunk Group** screen is **mf**.

Valid entries	Usage
1 to **8**	Enter the MFC signaling parameters set used by this trunk group.

Receive Analog Incoming Call ID

15 characters of name and number information associated with an incoming call on analog trunks (ICLID, or incoming call line identification information) is stored and displays. This field appears for CO, DID, and DIOD trunk groups when the **Analog Trunk Incoming Call ID** field on the **System-Parameters Customer-Options** screen is **y** and the **Direction** field is **incoming** or **two-way**.

Valid entries	Usage
Bellcore	Used to collect ICLID information in the U.S.
NTT	Used to collect ICLID information in Japan.
disabled	Stops the collection of ICLID information on analog trunks.

Replace Unavailable Numbers

Appears when the **Group Type** field is **isdn** or **sip**. Indicates whether to replace unavailable numbers with administrable strings for incoming and outgoing calls assigned to the specified trunk group. This field applies to BRI/PRI and SIP trunks.

Valid entries	Usage
y/n	Enter **y** for the display to be replaced regardless of the service type of the trunk.

Request Category

This field appears when the **Country** field is **15** and the **Shuttle** field is **y**.

Valid entries	Usage
y/n	Enter **y** if Communication Manager should request a call category from the central office.

Seize When Maintenance Busy

This field only affects ports on TN760C (or later release), TN767, and TN464C (or later release) circuit packs. It indicates whether this media server or switch generates an outgoing seizure when a trunk in this trunk group is maintenance busied and whether the far-end server or switch is administered to do likewise. This supports the Electronic Tandem Network Busyout feature, which is intended to prevent a far-end server or switch from reporting problems with a trunk that has been removed from service on your end. This field's setting does not affect the behavior of the far-end server or switch; it controls the behavior of your media server or switch and defines the expected far-end behavior.

For DIOD trunks using TN464F (or later release) or TN2464, displays only when the **Group Type** field is **diod** and the **Trunk Signaling Type** field is **pulsed**, **cont**, or **dis**.

Valid entries	Usage
near-end	Enter **near-end** if Communication Manager generates an outgoing seizure when a trunk is maintenance busied, but the far-end server or switch does not. The seizure is maintained until the maintenance busyout is released.

1 of 2

Valid entries	Usage
far-end	Enter **far-end** if the far-end server or switch generates an outgoing seizure when a trunk is maintenance busied, but this server running Communication Manager does not.
both-ends	Enter **both-ends** if both this server running Communication Manager and the far-end server or switch generate an outgoing seizure when a trunk is maintenance busied.
	2 of 2

If a media server or switch generates an outgoing seizure when a trunk is busied out, the seizure will probably cause alarms at the far-end server or switch, perhaps leading to a far-end maintenance busy out, unless the far-end server or switch is administered to expect this behavior.

If the administered value of this field is either **far-end** or **both-ends**, any abnormally long incoming seizure (including failure to drop from a completed call) is assumed to be the result of a far-end maintenance busy condition. Note that this assumption may be incorrect, since the abnormally long seizure may actually be due to failure of the trunk circuit. Regardless of the cause of the abnormally long seizure, your server running Communication Manager does the following things:

1. Generates a warning alarm indicating that the trunk is assumed to be maintenance busy at the far-end
2. Removes the trunk from service
3. Keeps the trunk out of service until a far-end disconnect is received

Allowable values depend on the entry in the **Direction** field: check the online help in Avaya Communication Manager.

Send Calling Number

Specifies whether the calling party's number is sent on outgoing or tandemed ISDN calls. Valid entries are **y**, **n**, or **r** (restricted). If **y** is entered, the **ISDN Numbering - Public/Unknown Format** screen is accessed to construct the actual number to be sent, or the **ISDN Numbering - Private** screen (based on the **Numbering Format** field) is used. If the value is **r**, the calling number is sent "presentation restricted.".

Note:

The **ISDN Numbering - Public/Unknown Format** screen can override the **Send Calling Number** field entry for any administrable block of extensions.

Send Called/Busy/Connected Number

Appears if the **QSIG Value-Added** field on the **Trunk Group** screen is **y**. Specifies if the dialed number, whether called (ringing), busy (busy tone), or connected (answered) is sent on incoming or tandemed ISDN calls. Valid entries are **y**, **n**, or **r** (restricted). If **y** is entered, the **ISDN Numbering - Public/Unknown Format** screen is accessed to construct the actual number sent, or the **ISDN Numbering - Private** screen (based on the **Numbering Format** field) is used. If the value is **r**, the connected number is sent "presentation restricted." The **Send Called/Busy/Connected Number** field must be set to **y** in order for the Calling Party Number of an incoming ISDN call to display at the transferred-to station after a QSIG transfer operation.

Send Connected Number

Appears if the **QSIG Value-Added** field on the **Trunk Group** screen is **n**. Specifies if the connected party's number is sent on incoming or tandemed ISDN calls. Valid entries are **y**, **n**, or **r** (restricted). If **y** is entered, the **ISDN Numbering - Public/Unknown Format** screen is accessed to construct the actual number sent, or the **ISDN Numbering - Private** screen (based on the **Numbering Format** field) is used. If the value is **r**, the connected number is sent "presentation restricted." The **Send Connected Number** field must be set to **y** in order for the Calling Party Number of an incoming ISDN call to display at the transferred-to station after a QSIG transfer operation.

Note:

The AT&T Switched Network Protocol does not support restricted displays of connected numbers. Therefore, if you administer the 1a country-protocol/ protocol-version combination on the **DS1** screen, you should not administer the **Send Connected Number** field to **r** (restricted) on the **ISDN Trunk Group** screen, as this causes display problems.

Note:

The **ISDN Numbering - Public/Unknown Format** screen overrides the **Send Connected Number** field entry for any administrable block of extensions.

Send Name

Appears only when the **Group Type** field is **isdn** or **sip**. Specifies whether the calling/ connected/called/busy party's administered name is sent to the network on outgoing/incoming calls. Valid entries are **y**, **n**, or **r** (restricted). The value **r** indicates that the calling/connected name will be sent by Avaya Communication Manager, but will be marked "presentation restricted." This value is valid only if the **Supplementary Service Protocol** field is **a** (national supplementary service), **b** (for called/busy only) or **d** for the QSIG Global Networking Supplementary Service Protocol. When the **Supplementary Service Protocol** field is **e** (DCS with Rerouting), only values of **y** and **n** are permitted.

Note:

If name information is not administered for the calling station or the connected/ called/busy station, the system sends the extension number in place of the name.

Short Holding Threshold

This field appears when the **ACA Assignment** field is **y**.

Valid entries	Usage
0 to **30**	Enter the number of times the system will record a short holding call before alerting an attendant to the possibility of a faulty trunk. Enter **0** for no short holding calls.

Short Holding Time (seconds)

This field appears when the **ACA Assignment** field is **y**.

Valid entries	Usage
0 to **160**	Enter the length of time (in seconds) that the system considers as being a short holding time. If **0** is entered, the system will not consider short holding calls.

Shuttle

This field appears when the **Group Type** field is **co**, **fx**, or **wats**, the **Country** field is **15**, and the **Outgoing Dial Type** field is **rotary**. It can be administered on TN464D (or later release) or TN2199 circuit packs.

Valid entries	Usage
y/n	Enter **y** to enable MF shuttle signaling.

Signaling Group

This field displays the group number assigned when the group was added.

Start B Signal

This field appears when the **Country** field is **15** and the **Shuttle** field is **y**. Enter **1** to **3** to indicate which B-signal should be used to start a call. The value administered in this field must be coordinated with your central office. See Start Position on page 1589.

Valid entries	Usage
1	Start calls with signal B1 (first digit)
2	Start calls with signal B2 (next digit)
3	Start calls with signal B3 (previous digit)

Start Position

The value administered in this field must be coordinated with your central office. This field appears when the **Country** field is **15** and the **Shuttle** field is **y**.

Valid entries	Usage
1 to **9**	Indicate which digit in the digit string is considered to be the "previously sent" digit (see Start B Signal on page 1589).

Suppress # Outpulsing

Valid entries	Usage
y/n	Enter **y** if end-to-end signaling begins with (and includes) "#". The final "#" is suppressed in cases where the system would normally outpulse it. This field should be **y** when the Central Office (for example, rotary) or any other facility treats "#" as an error.

Time (sec) to Drop Call on No Answer

This field appears if the **Group Type** field is **co** or **diod** and the **Outgoing Dial Type** field is **mf**, or if the **Group Type** field is **co**, **diod**, **fx**, or **wats** and the **Country** field is **15**.

Valid entries	Usage
0 to **1200**	Enter the duration (in seconds) Communication Manager should wait for outgoing calls to be answered. If the call is not answered in the specified number of seconds, the call drops. If this field is **0**, the timer is not set and no calls drop.

Used for DCS

Valid entries	Usage
y/n	Enter **y** if this trunk group will send and receive messages on a DCS signaling link.

Note:

This field cannot be activated if the trunk group number is greater than 255 or if the Trunk Access code is more than 3-digits long.

If this field is **y**, you can administer ISDN-BRI trunk groups unless the **DCS Signaling** field is **d-chan**. In that case, remove the BRI trunks or set the **DCS Signaling** field to **bx.25** before submitting the screen.

Field descriptions for Administrable Timers page

This screen may not appear for all trunk group types. The figure below is only an example, and is intended to show most of the fields that might appear on this page of the **Trunk Group** screen. This example may might not show all fields, or might show fields that normally do not appear together. Your own screen mary vary from this example according to specific field and system settings. The list of field descriptions that follow the figure is in alphabetical order for quick reference. This list is intended to be comprehensive, and may include information on fields that are not shown in the example. The field descriptions identify fields that are specific to particular trunk group types.

Figure 539: Administrable Timers for Trunk Group screen

```
add trunk-group next                                          Page 3 of x
                          ADMINISTRABLE TIMERS
               Send Incoming/Outgoing Disconnect Timers to TN465 Ports? _
                                          Outgoing Dial Guard(msec): _____
  Incoming Glare Guard(msec): _____     Outgoing Glare Guard(msec): _____

                                Outgoing Rotary Dial Interdigit (msec): ____
      Ringing Monitor(msec): _____            Incoming Seizure(msec): ____
   Outgoing End of Dial(sec): ___      Outgoing Seizure Response(sec): ___
 Programmed Dial Pause(msec): _____      Disconnect Signal Error(sec): ___
         Flash Length(msec): ____
                                                Busy Tone Disconnect?

END TO END SIGNALING
    Tone (msec): ____    Pause (msec): 150

OUTPULSING INFORMATION
     PPS: 10    Make(msec): 40    Break(msec): 60  PPM? y  Frequency: 50/12k
```

CAUTION:

Customers: Do not change fields on this page without assistance from Avaya or your network service provider.

Answer Send (msec)

This field appears only if the **Trunk Type** field is **dis** and the trunk group **Direction** field is **incoming** or **two-way**. Only TN2140 and TN2199 ports receive this timer.

Valid entries	Usage
10 to **2550** in increments of 10	Enter the duration of the answer signal pulse.

Busy Tone Disconnect

The field appears when **Enable Busy Tone Disconnect for Analog loop-start Trunks** is **y** on the **System Parameters Country-Options** screen.

Valid entries	Usage
y/n	Enter **y** to allow Communication Manager to recognize a busy tone signal as a disconnect on this trunk group.

Cama Outgoing Dial Guard (msec)

This field appears when **Group Type** is **cama** (the trunk group type used for emergency 911 service).

Valid entries	Usage
25 to **6375** in increments of 25	Enter the minimum interval between the seizure acknowledgment on the receiving server or switch and the outpulsing of digits by this server.

Cama Wink Start Time (msec)

This field appears when **Group Type** is **cama**.

Valid entries	Usage
20 to **5100** in increments of 20	Specifies the duration (the wait-for-wink-to-end time) for a wink-start CAMA trunk. The wink must begin before the Outgoing Seizure Response timer expires.

Disconnect Signal Error (sec)

This field appears for ground-start trunk groups.

Valid entries	Usage
1 to **255** in increments of 1	Enter the maximum interval that Communication Manager will wait to receive a disconnect signal from the far-end after the local party (a telephone or tie trunk) goes on-hook. If the timer expires, Communication Manager assumes a disconnect failure and takes appropriate action, such as creating an error message.

Flash Length (msec)

This timer is sent to TN436B, TN459B, TN464C (or later), TN465B (or later), (TN753 if Country is **23**), TN2146, TN2147, TN2184, and TN2199 circuit boards.

Valid entries	Usage
10 to **2550** in increments of 10	Enter the duration of a flash signal generated toward the central office.

Glare

This field is only administrable if the **Trunk Type** field is **cont** and the trunk group **Direction** field is **two-way** or **outgoing**. Only TN2140 ports receive this timer.

Valid entries	Usage
40 to **100** in increments of 10	Enter the minimum acceptable interval (in msec) between the moment your server running Communication Manager sends an outgoing seizure and the moment it receives a seizure acknowledgment. If acknowledgment is received before the timer expires, glare is assumed.

Incoming Dial Guard (msec)

Valid entries	Usage
10 to **2550** in increments of 10	Enter the minimum acceptable interval between the detection of an incoming seizure and the acceptance of the first digit. Communication Manager will not accept digits before this timer expires.

Note:

This timer is never sent to TN429 ports.

Incoming Disconnect (msec)

The field appears only when the **Direction** field is **incoming** or **two-way** and the **Trunk Type** field is either blank or **cont**.

Valid entries	Usage
50 to **2550** in increments of 10	Enter the minimum valid duration of a disconnect signal for an incoming call. Avaya Communication Manager will not recognize shorter disconnect signals. This field cannot be blank. For Brazil pulsed E&M signaling, use **600**.

Incoming Disconnect Send (msec)

This field is only administrable if the **Trunk Type** field is **dis** and the trunk group **Direction** field is **incoming** or **two-way**. Only TN2140 ports receive this timer.

Valid entries	Usage
500 to **1200** in increments of 100	Enter the duration of the backward release signal your server running Communication Manager sends at the end of an incoming call.

Incoming Glare Guard (msec)

This field only appears when the trunk group **Direction** field is **two-way**.

Valid entries	Usage
100 to **25500** in increments of 100	Enter the minimum interval that must elapse between a trunk's release from an incoming call and its seizure for an outgoing call. This field cannot be blank. This delay gives the far-end time to release all equipment after the trunk is released.

Incoming Incomplete Dial Alarm (sec)

Only the TN436 (all), TN459 (all), TN464C (or later), TN767, TN2140, TN2146, TN2184, TN2199, and TN2242 circuit packs use this timer.

Valid entries	Usage
1 to **255** in increments of 1	Enter the maximum acceptable interval between an incoming seizure and receipt of all digits. Intervals greater than this limit generate an inline error.

Incoming Partial Dial (sec)

This timer appears only if the **Incoming Dial Type** field is **rotary**.

Valid entries	Usage
5 to **255** in increments of 1	Enter the maximum time allowed between incoming rotary digits.

Note:

This timer is never sent to TN429 ports.

Incoming Seizure (msec)

This field appears when the **Direction** field is **incoming** or **two-way**, and, when applicable, the **Trunk Type** field is **cont**. Only TN429, TN438 (any release), TN 447, TN464C (or later), TN465 (any release), TN767, TN2138, TN2140, TN2147, TN2184, and TN2199 ports receive this timer. For DID trunks, only TN2199 and TN429D (or later) receive this timer.

Valid entries	Usage
20 to **2550** in increments of 10	Enter the duration of the shortest incoming seizure signal your server running Communication Manager can recognize. For ICLID, set this field to 120. The field cannot be blank.

Normal Outgoing Seize Send (msec)

This field appears only if the **Trunk Type** field is **dis** and the trunk group **Direction** field is **two-way** or **outgoing**. Only TN2140 ports receive this timer.

Valid entries	Usage
10 to **990** in increments of 10	Enter the duration of the signal your server running Communication Manager sends for an outgoing seizure.

Outgoing Dial Guard (msec)

Valid entries	Usage
100 to **25500** in increments of 100	Enter the minimum interval between seizure acknowledgment of a trunk and the outpulsing of digits. This field cannot be blank. For trunks that do not provide seizure acknowledgment, the timer specifies the minimum time between seizure and the outpulsing of digits. Any digit the caller dials after they lift the receiver, but before the timer expires, is not outpulsed until the timer expires.

Outgoing Disconnect (msec)

Valid entries	Usage
50 to **2550** in increments of 10	Enter the minimum valid duration of a disconnect signal for an outgoing call. Communication Manager will not recognize shorter disconnect signals. This field cannot be blank. This timer begins timing when a disconnect signal is detected on an outgoing call and resets when the signal is no longer detected. If the timer expires, the trunk drops. For Brazil pulsed E&M signaling, use **600**.

Outgoing Disconnect Send (msec)

This field is administrable only if the **Trunk Type** field is **dis** and the trunk group **Direction** field is **two-way** or **outgoing**. Only TN2140 ports receive this timer.

Valid entries	Usage
100 to **9900** in increments of 100	Enter the duration of the forward release signal your server running Communication Manager sends at the end of outgoing calls.

Outgoing End of Dial (sec)

This field controls firmware answer supervision timers on circuit packs that have them. It appears when the **Direction** field is **outgoing** or **two-way** and the **Receive Answer Supervision** field is **n**. If the **Receive Answer Supervision** field is **y**, this field does not appear and the firmware timer on the appropriate circuit pack is automatically disabled.

Note:

The **Answer Supervision Timeout** field on the **Trunk Group** screen provides timed answer supervision for circuit packs without administrable timers. Since trunk groups may contain ports on more than one circuit pack, it's possible you may need to use both timers with the same trunk group. If so, set the **Outgoing End of Dial** field and the **Answer Supervision Timeout** field to the same value.

Valid entries	Usage
1 to **254** in increments of 1	Enter the maximum time, in seconds, that Communication Manager will wait to receive answer supervision for outgoing calls on the ports controlled by firmware timers. For Brazil pulsed E&M signaling, use **40**.

During a cut-through operation, timing begins after Communication Manager sends each outgoing digit and ceases when answer supervision is received. If the timer expires, Communication Manager acts as if it has received answer supervision. On senderized operation, the timer begins after the switch sends the last digit collected. The timer ceases when answer supervision is received. If the timer expires, Communication Manager acts as if it has received answer supervision.

Outgoing Glare Guard (msec)

This field only appears for **outgoing** and **two-way** trunk groups.

Valid entries	Usage
100 to **25500** in increments of 100	Enter the minimum interval that must elapse between a trunk's release from an outgoing call and its seizure for another outgoing call. This field cannot be blank. This delay gives the far-end time to release all equipment after the outgoing trunk is released.

Outgoing Last Digit (sec)

This field is only administrable if the **Trunk Type** field is **dis** or **cont** and the trunk group **Direction** field is **two-way** or **outgoing**. Only TN497 and TN2140 ports receive this timer.

Valid entries	Usage
1 to **40** in increments of 1	Enter the maximum time that Communication Manager will wait for the next digit dialed. After the timer expires, no more digits are accepted by the circuit pack.

Outgoing Rotary Dial Interdigit (msec)

This field only appears when:

1. the trunk group **Group Type** field is **access**, **aplt**, **co**, **diod**, **dmi-bos**, **fx**, **rlt**, **tandem**, or **wats** and the **Outgoing Dial Type** field is **rotary**.
2. the **Group Type** field is **tie**, the **Trunk Type** field is blank, **cont**, or **dis**, and the **Outgoing Dial Type** field is **rotary**.
3. the **Group Type** field is **tie**, and the **Trunk Type** field is **tge**, **tgi**, or **tru** (the **Outgoing Dial Type** field does not appear but is implied to be **rotary**).

Valid entries	Usage
150 to **2550** in increments of 10	Enter the minimum time between outpulsed digits on outgoing rotary trunks.

Outgoing Seizure (msec)

Appears when the **Country** field is **15**, the **Direction** field is **outgoing** or **two-way**, and the **Trunk Type** field is **2-wire-ac**, **2-wire-dc**, or **3-wire**. This timer is sent only to the TN2199 circuit pack.

Valid entries	Usage
20 to **2550** in increments of 10	Enter the duration of the outgoing seizure signal.

Outgoing Seizure Response (sec)

This timer is sent to the TN438B, TN439, TN447, TN458, TN464B (or later), TN465B (or later), TN767, TN2140, TN2147, TN2184, TN2199, and TN2242 circuit packs.

Valid entries	Usage
1 to **255** in increments of 1	Enter the maximum interval that Communication Manager should wait after sending a seizure signal to receive seizure acknowledgment from the far-end. If the acknowledgment is not received in this time, a seizure failure response is uplinked. For Brazil pulsed E&M signaling, use **255**.

Programmed Dial Pause (msec)

This timer is administrable for all outgoing and two-way trunk groups. This timer works with the TN464B (or later), TN767, TN458, TN2140, and TN2242 tie circuit packs. All CO circuit packs that accept administrable timers accept this timer.

Valid entries	Usage
100 to **25500** in increments of 100	Set the exact duration of pauses used during abbreviated dialing, ARS outpulsing, and terminal dialing operations.

Release Ack Send (msec)

After your server running Communication Manager receives a forward release signal, it must send a forward release acknowledgment signal. This field appears only if the **Trunk Type** field is **dis** and the trunk group **Direction** field is **incoming** or **two-way**. Only TN2140 ports receive this timer.

Valid entries	Usage
500 to **1200** in increments of 100	Enter the duration of the signal your server running Communication Manager sends for a forward release acknowledgment.

Ringing Monitor (msec)

This timer is sent to TN464C (or later), TN767, TN438 (all), TN447, TN465 (all), TN2138, TN2147, TN2184, and TN2199 CO circuit packs.

Valid entries	Usage
200 to **51000** in increments of 200	Enter the minimum time Communication Manager requires to determine if a trunk disconnects. The field cannot be blank. If the ringing signal disappears for a duration longer than the time specified in this field, Communication Manager assumes the call has been disconnected.

Seize Ack Delay (msec)

This field appears only if the **Trunk Type** field is **dis** and the trunk group **Direction** field is **incoming** or **two-way**. Only TN2140 ports receive this timer.

Valid entries	Usage
40 to **120** in increments of 10	Enter the maximum interval your server running Communication Manager will wait after receipt of an incoming seizure to send seizure acknowledgment.

Seize Ack Send (msec)

This field appears only if the **Trunk Type** field is **dis** and the trunk group **Direction** field is **incoming** or **two-way**. Only TN2140 ports receive this timer.

Valid entries	Usage
10 to **990** in increments of 10	Enter the duration of the seizure acknowledgment signal your server running Communication Manager sends in response to an incoming seizure.

Send Incoming/Outgoing Disconnect Timers to TN465 Ports

The field appears only for a co, fx, or wats trunk group.

Valid entries	Usage
y/n	Enter **y** if you want to send the incoming disconnect and outgoing disconnect timer values to the trunk group ports that are on a TN465 board.

END TO END SIGNALING

Pause (msec)

THis field is administrable only if the **Trunk Type** field is blank. All CO, DIOD, and tie circuit packs that accept administrable timers accept this timer. However, this timer is sent only to the following circuit packs: TN464B (or later), TN767, TN436B, TN459B, TN2146, TN2199, and TN2242, and TN429 and TN2184 ports in a DID trunk group.

Valid entries	Usage
20 to **2550** in increments of 10	Enter the minimum acceptable interval (pause) between DTMF tones sent from a hybrid telephone.

Tone (msec)

This field appears only if the **Trunk Type** field is blank. All CO, DIOD, and Tie circuit packs that accept administrable timers accept this timer. This timer is also sent to the following circuit packs: TN464B (or later), TN767, TN436B, TN459B, TN2146, TN2199, TN429, TN2184 ports in a DID trunk group.

Valid entries	Usage
20 to **2550** in increments of 10	Enter the duration of a DTMF tone sent when a button on a hybrid telephone is pressed.

OUTPULSING INFORMATION

Break (msec)

Valid entries	Usage
Enter the duration of the break interval (the pulse duration) while the system is outpulsing digits using dial pulse signaling. The field cannot be blank.	
20 to **80** in increments of 5	If PPS field is **10**, the sum of the **Make (msec)** and **Break (msec)** fields must equal 100.
10 to **40** in increments of 5.	If the **PPS** field is **20**, the sum of the **Make (msec)** and **Break (msec)** fields must equal 50.

Frequency

This field identifies the PPM pulse frequency(ies) sent by the public network. It appears if the **Direction** field is **outgoing** or **two-way** and **PPM** is **y**. Circuit packs can detect up to three different frequencies, (12kHz, 16kHz, and 50Hz), plus two frequency combinations, (50Hz/12kHz and 50Hz/16kHz). This field controls TN465B, TN2138, and TN2184 circuit packs.

Valid entries	Usage
12k	The TN465B (or later) and TN2184 can only detect 12k and 16kHz PPM. Therefore, if **12k** is administered, the circuit pack will be set to detect 12kHz.
16k	The TN465B (or later) and TN2184 can only detect 12k and 16kHz PPM. Therefore, if **16k** is administered, the circuit pack will be set to detect 16kHz.
50	The TN465B (or later) and TN2184 can only detect 12k and 16kHz PPM. Therefore, if **50** is administered, the circuit pack will be set to detect 16kHz.
50/12k	The TN465B (or later) and TN2184 can only detect 12k and 16kHz PPM. Therefore, if **50/12k** is administered, the circuit pack will be set to detect 12kHz.
50/16k	The TN465B (or later) and TN2184 can only detect 12k and 16kHz PPM. Therefore, if **50/16k** is administered, the circuit pack will be set to detect 16kHz.

Make (msec)

Valid entries	Usage
Enter the duration of the make interval (the pause between pulses) while the system is outpulsing digits using dial pulse signaling. The field cannot be blank.	
20 to **80** in increments of 5	If the **PPS** field is **10**, the sum of the **Make (msec)** and **Break (msec)** fields must equal 100.
10 to **40** in increments of 5	If the **PPS** field is **20**, the sum of the **Make (msec)** and **Break (msec)** fields must equal 50.

PPM

For CO, DIOD, FX, PCOL, and WATS trunks. This field appears when the **Direction** field is **outgoing** or **two-way**.

Valid entries	Usage
y/n	Enter **y** if Periodical Pulse Metering (PPM) pulses should be collected from the public network to determine call cost. If this field is **y**, the **Frequency** field appears.

PPS

Valid entries	Usage
10	Enter the rate (pulses per second) at which outgoing rotary pulses are sent over this trunk group.
20	

Note:

The TN439, TN458, TN497, TN747Bv12 (or later), and TN767 circuit packs only send rotary pulses at 10 pps.

Field descriptions for Incoming Call Handling Treatment page

This screen appears as page 3 for Group Types **sip** and **isdn**. For more information on ISDN trunk groups, see the ISDN Trunk Group section of this chapter.

Figure 540: Incoming Call Handling Treatment screen

```
add trunk-group next                                          Page 3 of 10
                        INCOMING CALL HANDLING TREATMENT
     Service/   Called     Called          Del      Insert       Per Call    Night
     Feature     Len       Number                              CPN/BN      Serv
   _____________  ___   ________________   ___   _____________ __________   _______
   _____________  ___   ________________   ___   _____________ __________   _______
   _____________  ___   ________________   ___   _____________ __________   _______
   _____________  ___   ________________   ___   _____________ __________   _______
   _____________  ___   ________________   ___   _____________ __________   _______
   _____________  ___   ________________   ___   _____________ __________   _______
   _____________  ___   ________________   ___   _____________ __________   _______
   _____________  ___   ________________   ___   _____________ __________   _______
   _____________  ___   ________________   ___   _____________ __________   _______
   _____________  ___   ________________   ___   _____________ __________   _______
   _____________  ___   ________________   ___   _____________ __________   _______
   _____________  ___   ________________   ___   _____________ __________   _______
   _____________  ___   ________________   ___   _____________ __________   _______
   _____________  ___   ________________   ___   _____________ __________   _______
   _____________  ___   ________________   ___   _____________ __________   _______
   _____________  ___   ________________   ___   _____________ __________   _______
   _____________  ___   ________________   ___   _____________ __________   _______
   _____________  ___   ________________   ___   _____________ __________   _______
   _____________  ___   ________________   ___   _____________ __________   _______
   _____________  ___   ________________   ___   _____________ __________   _______
```

Called Len

Specifies the number of digits received for an incoming call. A blank entry is used as a "wild card" entry and, when used, means that any length of digits associated with the specified Service/Feature can match in this field. Valid entries are **0** to **21**, or leave blank.

Called Number

Specifies the leading digits received for an incoming call. A blank entry is used as a "wild card" entry and, when used, means that any number associated with the specified Service/Feature can match in this field. Valid entries are up to 16 digits, or leave blank.

Del

Specifies the number of leading digits to be deleted from the incoming Called Party Number. Calls of a particular type may be administered to be routed to a single destination by deleting all incoming digits and then administering the **Insert** field with the desired extension. Valid entries are **1** to **21**, **all**, or leave blank.

Insert

Specifies the digits to be prepended to the front of the remaining digits after any (optional) digit deletion has been performed. The resultant number formed from digit deletion/insertion is used to route the call, provided night service is not in effect. Valid entries are up to 16 characters consisting of a combination from the following: **0** to **9**, *****, **#**, or leave blank.

Per Call CPN/BN

This field appears only when **Group Type** is **isdn**. For details on this field, see the Field descriptions for Incoming Call Handling Treatment page on page 1604 of the ISDN Trunk Groups section of this chapter.

Night Serv

This field appears only when **Group Type** is **isdn**. For details this field, see the Field descriptions for Incoming Call Handling Treatment page on page 1604 of the ISDN Trunk Groups section of this chapter.

Service/Feature

Specifies the Services/Features for an incoming call type. See the **Service Type** field description for a list of predefined Services/Features that can be received. Or, use **Type 0**, **Type 1**, and **Type 2** user defined services. The identifier other can be used for any Services/Features not explicitly specified.

Field descriptions for ATMS Thresholds page

This screen appears when the **Direction** field on page 1 is **outgoing** or **two-way** and the **ATMS** field is **y** on the **Feature-Related System Parameters** screen.

The figure below shows a common configuration for the **ATMS Thresholds** page of the **Trunk Group** screen. This screen is only an example, and the fields shown below may change or disappear according to specific field settings.

Figure 541: ATMS Thresholds screen

```
add trunk-group next                                          Page 4 of x
                          ATMS THRESHOLDS
TTL Type: _________                  Far End Test No: ____________________
  TTL Vendor: _______________            TTL Contact: ________________________
Trunk Vendor: _____________            Trunk Contact: ______________________
Trunk Length: _____
                          MARGINAL          UNACCEPTABLE
                          Min   Max           Min   Max
           1004 Hz Loss: __     __           __    __

                          -Dev  +Dev          -Dev   +Dev
            404 Hz Loss:   _     _             _      _
           2804 Hz Loss:   _     _             _      _

Maximum C Message Noise:      __                  __
Maximum C Notched Noise:      __                  __
         Minimum SRL-HI:      __                  __
         Minimum SRL-LO:      __                  __
            Minimum ERL:      __                  __

     Allow ATMS Busyout, Error Logging and Alarming? _
  Maximum Percentage of Trunks Which Can Be Removed from Service by ATMS: __
```

CAUTION:

Customers: Do not change fields on this page without assistance from Avaya or your network service provider.

Far-End Test No.

Valid entries	Usage
1 to 16 digits	Enter the access number dialed to reach the terminating test line (TTL).

Trunk Contact

Valid entries	Usage
0 to 25 alphanumeric characters	Enter the name and/or telephone number of someone from the trunk vendor who can be contacted in the event of problems with the trunks.

Trunk Length

This field is not required. Since noise on a trunk increases with the length of the trunk, however, this information may be useful,

Valid entries	Usage
Use this field to record the length of the trunk group in kilometers or miles.	
0 to 4 digits followed by **k**	Shows the length in kilometers.
0 to 4 digits followed **m**	Shows the length in miles.

Trunk Vendor

Valid entries	Usage
0 to 22 alphanumeric characters	Enter the name of the vendor providing service over this trunk group (the company to notify in the event of problems with the trunks in this trunk group).

TTL Contact

Valid entries	Usage
0 to 25 alphanumeric characters	Enter the name and/or telephone number of someone from the TTL vendor who can be contacted in the event of problems with the terminating test line.

TTL Type

Specifies the type of terminating test line (TTL) selected for testing trunks. The TTL type determines what ATMS tests can be completed and thus which threshold values need to be administered.

Valid entries	Usage
105-w-rl	105 with return loss
105-wo-rl	105 without return loss
high-lts	high-level tone source
low-lts	low-level tone source
100	100 type
102	102 type

The following table explains the differences between types of terminating test lines:

Type TTL	**Description**	**Example**
105-w-rl	Full range of 18 measurements or some defaults for return loss used (56A)	TN771B, ZLC12 and SN261B circuit packs and new 56A mini-responder
105-wo-rl	Cannot return default values for far-end return loss	Older 56A mini-responder
high-level-tone	Sends a fixed sequence of tones at 0 dBm	ZLC12 and SN261B circuit packs
low-level-tone	Sends a fixed sequence of tones at -16dBm	SN261B circuit pack
100	Up to 5 measurements that sends a 1004 Hz tone then a quiet termination	
102	One measurement that sends a 1004 Hz tone	

The far-end server or switch containing the TTL may be any of the following:

- System 85 R2 switch, equipped with the Maintenance/Test Board (TN771B)
- System 75 R1V2 and beyond, all of which contain the circuitry required to perform the TTL function
- System 85 R2 switch, equipped with the Analog/Digital Facility Test Circuit (ADFTC, SN261)
- DIMENSION FP8, equipped with the Analog Facility Test Circuit (AFTC, ZLC-12)
- Central Office switches, equipped with various TTL equipment that provide 100, 102, or 105 test line capabilities (56A)

Other vendors' switches may be supported if compatible test lines are provided by these switches.

Four different versions of the ATMS Threshold Administration page can appear depending upon the measurements allowed by the TTL type selected. The four possibilities are:

1. 105-w-rl and 105-wo-rl - All thresholds appear.
2. high-lts and low-lts - All thresholds (except maximum C-notched noise) appear.
3. 100 - All thresholds (except maximum c-notched noise, 404Hz loss, and 2804 Hz loss) appear.
4. 102 - Only 1004 Hz loss threshold appears.

TTL Vendor

Valid entries	Usage
0 to 22 alphanumeric characters	Enter the name of the vendor supplying the terminating test line (TTL).

MARGINAL / UNACCEPTABLE

Allow ATMS Busyout, Error Logging and Alarming

Valid entries	Usage
y/n	Enter **y** to allow ATMS error logging and alarming (subject to filtering depending on the service organization used to deal with alarms).

Marginal Threshold - -Dev - 404 Hz Loss

Valid entries	Usage
0 to **9**	Enter the maximum negative deviation of measured loss at 404 Hz from the 1004 Hz test tone noise level (in dB) allowed before reporting a trunk as out of tolerance. Smaller dB values are more restrictive.

Marginal Threshold - +Dev - 404 Hz Loss

Valid entries	Usage
0 to **9**	Enter the maximum positive deviation of measured loss at 404 Hz from the 1004 Hz test tone loss level (in dB) allowed before reporting a trunk as out of tolerance. Smaller dB values are more restrictive.

Marginal Threshold - -Dev - 2804 Hz

Valid entries	Usage
0 to **9**	Enter the maximum negative deviation of measured loss at 2804 Hz from the 1004 Hz test tone loss level (in dB) allowed before reporting a trunk as out of tolerance. Smaller dB values are more restrictive.

Marginal Threshold - +Dev - 2804 Hz

Valid entries	Usage
0 to **9**	Enter the maximum positive deviation of measured loss at 2804 Hz from the 1004 Hz test tone loss level (in dB) allowed before reporting a trunk as out of tolerance. Smaller dB values are more restrictive.

Marginal Threshold - Max - 1004 Hz Loss

Valid entries	Usage
0 to **21**	Enter the maximum signal loss allowed for a 1004 Hz test tone (in dB) before a trunk is reported as out of tolerance. A smaller dB value is more restrictive.

Marginal Threshold - Min -1004 Hz Loss

Valid entries	Usage
-2 to **21**	Enter the minimum signal loss allowed for a 1004 Hz test tone (in dB) before a trunk is reported as out of tolerance. A larger dB value is more restrictive.

Marginal Threshold - Minimum ERL

Valid entries	Usage
0 to **40**	Enter the minimum low-frequency echo return loss in dB allowed before reporting a trunk as out of tolerance. Larger values are more restrictive.

Marginal Threshold - Maximum C Message Noise

Valid entries	Usage
15 to **55**	Enter the maximum C-message noise telephone as measured within the voice band frequency range (500 to 2500 Hz) allowed before reporting a trunk as out of tolerance. Smaller values are more restrictive.

Marginal Threshold - Maximum C Notched Noise

Valid entries	Usage
34 to **74**	Enter the maximum C-notched signal dependent noise interference in dBmC allowed before reporting a trunk as out of tolerance. Smaller values are more restrictive.

Marginal Threshold - Minimum SRL-HI

Valid entries	Usage
0 to **40**	Enter the minimum high-frequency signaling return loss in dB allowed before reporting a trunk as out of tolerance. Larger values are more restrictive.

Marginal Threshold - Minimum SRL-LO

Valid entries	Usage
0 to **40**	Enter the minimum low-frequency signaling return loss in dB allowed before reporting a trunk as out of tolerance. Larger values are more restrictive.

Maximum Percentage of Trunks Which Can Be Removed From Service by ATMS

Appears when the **Allow ATMS Busyout**, **Error Logging and Alarming** field is **y**.

Valid entries	Usage
0, **25**, **50**, **75**, **100**	Enter the highest percentage of trunks from the trunk group that can be removed from service at one time because of unacceptable transmission measurement results.

Unacceptable Threshold - -Dev - 404 Hz

Valid entries	Usage
0 to **9**	Enter the maximum negative deviation of measured loss at 404 Hz from the 1004 Hz test tone loss level (in dB) allowed before reporting a trunk as unacceptable. Smaller dB values are more restrictive.

Unacceptable Threshold - +Dev - 404 Hz

Valid entries	Usage
0 to **9**	Enter the maximum positive deviation of measured loss at 404 Hz from the 1004 Hz test tone loss level (in dB) allowed before reporting a trunk as unacceptable. Smaller dB values are more restrictive.

Unacceptable Threshold - -Dev - 2804 Hz

Valid entries	Usage
0 to **9**	Enter the maximum negative deviation of measured loss at 2804 Hz from the 1004 Hz test tone loss level (in dB) allowed before reporting a trunk as unacceptable. Smaller dB values are more restrictive.

Unacceptable Threshold - +Dev - 2804 Hz

Valid entries	Usage
0 to **9**	Enter the maximum positive deviation of measured loss at 2804 Hz from the 1004 Hz test tone loss level (in dB) allowed before reporting a trunk as unacceptable. Smaller dB values are more restrictive.

Unacceptable Threshold - Max - 1004 Hz Loss

Valid entries	Usage
0 to **21**	Enter the maximum signal loss allowed for a 1004 Hz test tone (in dB) before a trunk is reported as unacceptable. A smaller dB value is more restrictive.

Unacceptable Threshold - Min - 1004 Hz Loss

Valid entries	Usage
-2 to **21**	Enter the minimum signal loss allowed for a 1004 Hz test tone (in dB) before a trunk is reported as unacceptable. A larger dB value is more restrictive.

Unacceptable Threshold - Maximum C Message Noise

Valid entries	Usage
15 to **55**	Enter the maximum C-message noise interference in dBmC above reference noise terminating on a telephone as measured within the voice band frequency range (500 to 2500 Hz) allowed before reporting a trunk as unacceptable. Smaller values are more restrictive.

Unacceptable Threshold - Maximum C Notched Noise

Valid entries	Usage
34 to **74**	Enter the maximum C-notched signal dependent noise interference in dBmC allowed before reporting a trunk as unacceptable. Smaller values are more restrictive.

Unacceptable Threshold - Minimum ERL

Valid entries	Usage
0 to **40**	Enter the minimum low-frequency echo return loss in dB allowed before reporting a trunk as unacceptable. Larger values are more restrictive.

Unacceptable Threshold - Minimum SRL-HI

Valid entries	Usage
0 to **40**	Enter the minimum high-frequency signaling return loss in dB allowed before reporting a trunk as unacceptable. Larger values are more restrictive.

Unacceptable Threshold - Minimum SRL-LO

Valid entries	Usage
0 to **40**	Enter the minimum low-frequency signaling return loss in dB allowed before reporting a trunk as unacceptable. Larger values are more restrictive.

Field descriptions for Group Member Assignments page

The total number of pages of the **Trunk Group** screen, and the page number of the first page of **Group Member Assignments**, vary depending on whether the **Administrable Timers** and **ATMS Thresholds** pages display. Note that the **Group Member Assignments** screen is repeated, as needed, to allow assignment of all group members.

Figure 542: Group Member Assignments screen

```
add trunk-group next
                                TRUNK GROUP
                                      Administered Members(min/max): xxx/yyy
                                         Total Administered Members: xxx
GROUP MEMBER ASSIGNMENTS
       Port     Code  Sfx  Name         Night          Mode     Type   Ans Delay
    1: _______                _________  _____        ________ _______ ____
    2: _______                _________  _____        ________ _______ ____
    3: _______                _________  _____        ________ _______ ____
    4: _______                _________  _____        ________ _______ ____
    5: _______                _________  _____        ________ _______ ____
    6: _______                _________  _____        ________ _______ ____
    7: _______                _________  _____        ________ _______ ____
    8: _______                _________  _____        ________ _______ ____
    9: _______                _________  _____        ________ _______ ____
   10: _______                _________  _____        ________ _______ ____
   11: _______                _________  _____        ________ _______ ____
   12: _______                _________  _____        ________ _______ ____
   13: _______                _________  _____        ________ _______ ____
   14: _______                _________  _____        ________ _______ ____
   15: _______                _________  _____        ________ _______ ____
```

Administered Members (min/max)

This display-only field shows the minimum and maximum member numbers that have been administered for this trunk group.

Ans Delay

CAUTION:

Customers should not attempt to administer this field. Please contact your Avaya technical support representative for assistance.

Valid entries	Usage
20 to **5100** in increments of 20	Specifies the length of time (in ms) your server running Communication Manager will wait before it sends answer supervision for incoming calls on tie trunks using the TN722A or later, TN760 (B, C, or D), TN767, TN464 (any suffix), TN437, TN439, TN458, or TN2140 circuit packs.
blank	Same as setting the field to zero.

This delay serves two purposes:

- It ensures that the answer supervision signal is valid and not a secondary delay-dial or wink-start signal.
- It ignores momentary off-hook signals resulting from connections made off-network through certain No. 5 Crossbar CCSA switches as the connection is being established. Therefore, calls aren't dropped inappropriately.

Code

This display-only field shows the type of circuit pack physically installed or logically administered at the location to which this member is assigned. If no circuit pack is installed or administered at the port address you enter, the field is blank.

Mode

This field specifies the signaling mode used on tie trunks with TN722A or later, TN760B or later, TN767, TN464 (any suffix), TN437, TN439, TN458, or TN2140 circuit packs. This entry must correspond to associated dip-switch settings on the circuit pack.

CAUTION:

Customers should not attempt to administer this field. Please contact your Avaya technical support representative for assistance.

Valid entries	Usage
e&m	Enter **e&m** for 6-wire connections that pair 2 signaling wires with 4 voice wires. You'll use **e&m** in the vast majority of systems in the U.S.
simplex	Enter **simplex** for 4-wire connections that do not use an additional signaling pair. This configuration is very rare in the U.S.
protected	

Name

Your vendor, as well as Avaya technical support staff, sometimes need to identify specific trunks to work with your system. Therefore, the name you give to a trunk should identify the trunk unambiguously.

Valid entries	Usage
Up to 10 characters	Examples of good names: • The telephone number assigned to incoming trunks • The Trunk Circuit Identification number assigned by your service provider

Night

Use this field only if you want to assign a night service destination to individual trunks that is different from the group destination entered in the **Night Service** field on page 1. Incoming calls are routed to this destination when the system is placed in night service mode.

Valid entries	Usage
a valid extension	Enter the extension of the night destination for the trunk.
attd	Enter **attd** if you want calls to go to the attendant when night service is active.
blank	

Port

If this trunk is registered as an endpoint in an IP application, this field will display T00000.

Valid entries	Usage
1 to **64** (S8700 or S8710 IP-Connect / S8700 or S8710 Multi-Connect / S8300 Media Server)	First and second characters are the cabinet number.
A to **E**	Third character is the carrier.
0 to **20**	Fourth and fifth characters are the slot number.

1 of 2

Valid entries	Usage
01 to **04** (Analog TIE trunks) **01** to **31** (R, CSI, SI, S8700 or S8710 IP-Connect, S8700 or S8710 Multi-Connect)	Six and seventh characters are the circuit number.
1 to **250** (S8700 or S8710 IP-Connect / S8700 or S8710 Multi-Connect / S8300 Media Server)	Gateway
V1 to **V9** (DEFINITY R, CSI, SI, S8700 or S8710 IP-Connect, S8700 or S8710 Multi-Connect)	Module
01 to **31** (DEFINITY R, CSI, SI, S8700 or S8710 IP-Connect, S8700 or S8710 Multi-Connect)	Circuit
	2 of 2

Note:

In DCS networks, trunks must be assigned the same member number at both nodes.

Note:

Members assigned to IP trunk groups will display a value of **T00001**.

Sfx

This display-only field shows the model suffix for the type of circuit pack physically installed at the location to which this member is assigned. If no circuit pack is installed at the port address you enter, the field is blank.

Total Administered Members

This display-only field shows the total number of members administered in the trunk group.

Type

The Type column appears when the **Trunk Type** field is blank or **cont**. The **Type** column does not display if the **Trunk Type** field is **dis**.

This field specifies the signaling type to be used with TN760B (or later release), TN722 (with any suffix), TN767, TN2140 (when the **Trunk Type** field is **cont**), TN437, TN439, TN464 with any suffix, or TN458 circuit packs.

CAUTION:

Customers should not attempt to administer this field. Please contact your Avaya technical support representative for assistance.

Valid entries	Usage
t1-stan	t1-stan (DEFINITY, S8700 or S8710 IP-Connect)
t1-comp	t1-comp (DEFINITY, S8700 or S8710 IP-Connect)
t5-rev	(DEFINITY, S8700 or S8710 IP-Connect) The value of **t5 rev** is allowed only for the TN760D vintage 10 or later. When Type is **t5 rev**, Mode must be **e&m**.
type-5	type-5 (DEFINITY, S8700 or S8710 IP-Connect)

Related topics

See Tips for Working with Trunk Groups on page 423 for instructions on adding and managing trunk groups.

See "Trunks and Trunk Groups" in *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205, for more information on all types of trunk groups except ISDN.

See "ISDN Service" in *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205, for more information.

Uniform Dial Plan Report

Figure 543: Uniform Dial Plan Report screen

```
list uniform-dialplan
                            UNIFORM DIAL PLAN REPORT

 Matching Pattern   Len   Del    Insert Digits     Net     Conv    Node Num
    2                4     0         817           aar      n
    4                5     1         334           aar      n
    43659            5     1         928           aar      y
    6                6     1                       ext      n
    73012            5     1                       enp      n        31
    74100            5     0         81            ars      y
```

Matching Pattern

The number you want Communication Manager to match to dialed numbers.

Len

The number of user-dialed digits the system collects to match to the dialed string.

Del

The number of digits deleted before routing the call.

Insert Digits

The digits that will be inserted at the beginning of the dialed number.

Net

The server or switch network used to analyze the converted number.

Conv

Indicates whether additional digit conversion is allowed.

Node Num

The Extension Number Portability (ENP) node number.

Uniform Dial Plan Table

The **Uniform Dialing Plan** field must be **y** on the **System-Parameters Customer-Options** screen before you can administer this table.

The UDP provides a common 3- to 7-digit dial plan length — or a combination of extension lengths — that can be shared among a group of media servers or switches. Additionally, UDP can be used alone to provide uniform dialing between two or more private switching systems without ETN, DCS, or Main/Satellite/Tributary configurations.

See "Uniform Dial Plan" in *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205, for more information.

Figure 544: Uniform Dial Plan Table screen

```
change uniform-dialplan 0                                          Page 1 of 2
                        UNIFORM DIAL PLAN TABLE
                                                      Percent Full:   2

Matching           Insert          Node     Matching         Insert          Node
Pattern  Len Del   Digits Net Conv Num      Pattern  Len Del Digits Net Conv Num
2         4   0    817    aar  n
4         5   1    334    aar  n
43659     5   1    928    aar  y
623       3   3    5380   aar  n
73012     5   1           enp  n    31
74100     5   0    81     ars  y
8         5   0           ext  n
911       3   0           ars  n
_____     _   _    ____   ___  _    __
_____     _   _    ____   ___  _    __
_____     _   _    ____   ___  _    __
_____     _   _    ____   ___  _    __
_____     _   _    ____   ___  _    __
_____     _   _    ____   ___  _    __
```

Conv

Valid entries	Usage
y/n	Enter **y** to allow additional digit conversion

Del

Valid entries	Usage
0 to **3**	Enter the number of digits to delete before routing the call. This number must be less than or equal to the number entered in the **Len** field.

Insert Digits

Valid entries	Usage
0 to **9** (1 to 4 digits)	Enter the digits that replace the deleted portion of the dialed number. Leave this field blank to simply delete the digits.
L*x* (1 to 5)	The variable ***x*** represents a number of digits taken from the locations prefix on the ***Locations*** screen. These digits are prepended to the dialed string. The value for ***x*** must less than the number of digits in the location prefix. Type a number between 1 and 5 that represents the number of leading x digits that should be prepended to (added to the front of) the dialed string.

Len

Valid entries	Usage
2 to **7**	Enter the number of user-dialed digits the system collects to match to this Matching Pattern. This number must be greater than or equal to the number entered in the **Matching Pattern** field. The value **2** may be used only when **Insert Digits** contains an **L*x*** value, where ***x*** is the number of leading digits to prependfor the location of an originating call..

Matching Pattern

Valid entries	Usage
0 to **9** (1 to 7 digits)	Enter the number you want Communication Manager to match to dialed numbers.

Net

Enter the server or switch network used to analyze the converted number.

Valid entries	Usage
aar, ars, enp, ext	The converted digit-string will be routed either as an extension number or via its converted AAR address, its converted ARS address, or its ENP node number. If you enter **enp**, you must enter the ENP node number in the **Node Num** field. The **Insert Digits** field must be blank, and **Conv** must be **n**.

Node Num

This is the ENP (Extension Number Portability) Node Number.

Valid entries	Usage
1 to **999**	Enter the ENP node number.

Percent Full

Displays the percentage (0 to 100) of the memory resources allocated for the uniform dial plan data that are currently being used.

Vector Directory Number

This screen defines vector directory numbers (VDN) for the Call Vectoring feature. A VDN is an extension number used to access a call vector. Each VDN is mapped to one call vector.

VDNs are software extension numbers (that is, not assigned to physical equipment). A VDN is accessed via direct dial CO trunks mapped to the VDN (incoming destination or night service extension), DID trunks, and LDN calls. The VDN may be Night Destination for LDN.

See the *Avaya Communication Manager Contact Center Call Vectoring and Expert Agent Selection (EAS) Guide* for more information.

Field descriptions for page 1

Figure 545: Vector Directory Number screen

```
change vdn 5000                                              Page  1 of x
                      VECTOR DIRECTORY NUMBER

                           Extension: 5000
                                Name:
                       Vector Number: 234
                  Attendant Vectoring: n
                 Meet-me Conferencing? n
                   Allow VDN Override? n
                                 COR: 59
                                  TN: 1
                            Measured: none
        Acceptable Service Level (sec):
         VDN of Origin Annc. Extension: 301
                            1st Skill:
                            2nd Skill:
                            3rd Skill:
```

1st/2nd/3rd Skill

Only appears when, on the **System-Parameters Customer-Options** screen, the **Expert Agent Selection (EAS)** field is **y** and if, on this screen, the **Meet-me Conferencing** field is **n**.

Valid entries	Usage
1 to **999** or blank	Enter the desired Skill numbers in each field.

Acceptable Service Level (sec)

Only appears when, on the **System-Parameters Customer-Options** screen, the **BCMS/ VuStats Service Level** field is **y** and the **Measured** field is **internal** or **both**.

Valid entries	Usage
0 to **9999** seconds	Enter the number of seconds within which calls to this VDN should be answered. This will allow BCMS to print out a percentage of calls that were answered within the specified time.

Allow VDN Override

This field appears if the **Meet-me Conferencing** field is **n**. This entry affects the operation of an agent's display and certain options/data assigned to the VDN when a call is routed through several VDNs.

For **Expert Agent Selection (EAS)**, if this field is **y** on the original VDN, the Skills of the new VDN will be used. If this field is **n** on the original VDN, the Skills of the original VDN will be used.

For Best Service Routing (BSR), if this field is **y** on the original VDN, the BSR Application and Available Agent Strategy of the new VDN will be used. If this field is **n** on the original VDN, the BSR Application and Available Agent Strategy of the original VDN will be used.

Valid entries	Usage
y	The name of the VDN appearing on the terminating display depends on the administration and chaining of the subsequent VDNs and the AUDIX mail for the last VDN is accessed.
n	The name of this VDN appears on the agent's display and the VDN's AUDIX mail is accessed. If any subsequent VDNs are used to process this call, their names will not appear on the terminating display and the AUDIX mail for the original VDN is accessed.

Attendant Vectoring

This field appears only if, on the **System-Parameters Customer-Options** screen, the **Attendant Vectoring** is **y** and if, on this screen, the **Meet-me Conferencing** field is **n**. This field identifies if the vector you are defining is an attendant vectoring VDN.

Valid entries	Usage
y	Enter **y** so the vector is an attendant vector. This entry will dynamically change the rest of the screen to eliminate field options available with other types of vectors. Default when, on the **System-Parameters Customer-Options** screen, both the **Vectoring (Basic)** and **Vectoring (Prompting)** fields are **n**. No changes allowed to this field.
n	Default when, on the **System-Parameters Customer-Options** screen, the **Vectoring (Basic)** and/or **Vectoring (Prompting)** fields are **y**.

COR

Specifies the class of restriction (COR) of the VDN.

Valid entries	Usage
0 to **95**	Enter a 1 or 2-digit number. This field cannot be blank.

Extension

This is a display-only field when using an administration command such as **add** or **change** to access the screen. The extension is a 1 to 7-digit number that starts with a valid first digit and length as defined by the system's dial plan.

Measured

This field appears if the **Meet-me Conferencing** field is **n**. Used to collect measurement data for this VDN. Data may be collected for reporting by BCMS or CMS.

Note:

On the **System-Parameters Customer-Options** screen, the **BCMS** field must be **y** for the **Measured** field to be set to **internal** or **both**. In addition, the appropriate CMS release must be administered on the **Feature-Related System Parameters** screen if this field is being changed to **external** or **both**.

Valid entries	Usage
internal	Data will be measured internally by BCMS.
external	Data will be measured internally by CMS.
both	Data will be measured internally by both BCMS and CMS.
none	Data will not be measured.

Meet-me Conference

This field appears only if, on the **System-Parameters Customer-Options** screen, the **Enhanced Conferencing** field is **y**. This field determines if the VDN is a Meet-me Conference VDN.

Note:

If the VDN extension is part of your DID block, external users will be able to access the conference VDN. If the VDN extension is not part of your DID block, only internal callers on the your network (including DCS or QSIG) or remote access callers can access the conference VDN.

Valid entries	Usage
y/n	Enter **y** to enable Meet-me Conference for this VDN. If Meet-me Conference is **y**, only **Extension**, **Name**, **Vector Number**, **Meet-me Conference**, **COR**, and **TN** fields display and the fields for page 2 change. Both **Attendant Vectoring** and **Meet-me Conference** cannot be enabled at the same time. If **Enhanced Conferencing** is **y**, but no other vectoring options are enabled, only Meet-me Conference vectors can be assigned.

Note:

If the vector for Meet-Me conferencing allows a new party to join a conference immediately, and that party is joining as an H.323 ip trunk user, the caller may not have talkpath with the others in the conference. To prevent this, include in the vector a short delay before a new party joins the Meet-Me conference, such as a step to collect digits, a 1-second delay, or play an announcement. Since Meet-Me vectors are always configured with announcements and digit collections, this should rarely be an issue.

Name

This is an optional field that need not contain any data. It is the name associated with the VDN.

Valid entries	Usage
Enter up to a 27-character alphanumeric name that identifies the VDN.	The name may be truncated on agents' displays depending on the application. When information is forwarded with an interflowed call, only the first 15 characters are sent

Note:

For 4620SW or 4610SW telephones, the **Name** field is supported by Unicode language display. For more information on Unicode language display, see Administering Unicode display on page 181.

Service Objective

Appears when **Skill** and **Centre Vu Advocate** are **y** on the **Feature Related System Parameters** screen.

Valid entries	Usage
1 to **9999**	Enter the service objective.

TN

Specifies the Tenant Partition number for this VDN.

Valid entries	Usage
1 to **20**	For R servers.
1 to **100**	For S8700 or S8710 IP-Connect.

VDN of Origin Annc. Extension

Data for this field appears only when, on the **System-Parameters Customer-Options** screen, the **VDN of Origin Announcement** field is **y** and, on this screen, the **Meet-me Conferencing** field is **n**.

Valid entries	Usage
VDN extension	Enter the extension number of the VDN of Origin announcement.

Vector Number

Identification number that specifies a particular call vector that is accessed through the VDN.

Valid entries	Usage
1 to **999**	Enter a 1 to 3-digit vector number. This field cannot be blank.

Field descriptions for page 2

Figure 546: Vector Directory Number screen

```
change vdn 5000                                              Page  2 of  x
                      VECTOR DIRECTORY NUMBER

                                AUDIX Name:

                           Return Destination:
                     VDN Timed ACW Interval:
                              BSR Application:
                BSR Available Agent Strategy: 1st-found



                Display VDN for Route-To DAC?
VDN Override for ISDN Trunk ASAI Messages?
```

AUDIX Name

Only appears for DEFINITY R and S8700 or S8710 IP-Connect configurations. If this VDN is associated with the AUDIX vector, enter the name of the AUDIX machine as it appears in the **Node Names** screen.

BSR Application

To use multi-site Best Service Routing with this VDN, specify an application plan for the VDN. This field appears if, on the **System Parameters Customer-Options** screen, the **Lookahead Interflow (LAI)** and **Vectoring (Best Service Routing)** fields are **y**.

Valid entries	Usage
1 to **255** or blank	Enter a 1 to 3-digit number. For DEFINITY R, CSI, SI.
1 to **511** or blank	Enter a 1 to 3-digit number. For S8300 Media Server, S8700 or S8710 IP-Connect, and S8700 or S8710 Multi-Connect.

BSR Available Agent Strategy

The available agent strategy determines how Best Service Routing identifies the "best" split or skill to service a call in an agent surplus situation. To use Best Service Routing with this VDN, enter an agent selection strategy in this field.

This field only appears if, on the **System Parameters Customer-Options** screen, the **Vectoring (Best Service Routing)** field is **y**.

Valid entries	Usage
1st-found	
UCD-LOA	Can be set only if, on the **System-Parameters Customer-Options** screen, the **Least Occupied Agent (LOA)** or **Business Advocate** field is **y**.
UCD-MIA	
EAD-LOA	Can be set only if, on the **System-Parameters Customer-Options** screen, the **Least Occupied Agent (LOA)** or **Business Advocate** field is **y**.
EAD-MIA	Can be set only if, on the **System-Parameters Customer-Options** screen, the **Expert Agent Selection (EAS)** field is **y**.

Display VDN for Route-To DAC

This field can be set to **y** only if, on the **System-Parameters Customer-Options** screen, the **Expert Agent Selection (EAS)** field is **y**.

For more information, see *Avaya Communication Manager Contact Center Call Vectoring and Expert Agent Selection (EAS) Guide*.

Valid entries	Usage
y/n	This field addresses situations where one of the following conditions is in effect: • A **`route-to number or route-to digits`** vector command routes an EAS direct agent call, with coverage option set to **y** • An adjunct routing step routes a direct agent call, with the coverage option set to **y**

Observe an Agent Answer

Valid entries	Usage
y/n	This field allows for a service observer to start observing a call to the VDN when the call is delivered to the agent/ station.

Return Destination

Valid entries	Usage
VDN extension or blank	Enter the VDN extension number to which an incoming trunk call will be routed if it returns to vector processing after the agent drops the call.

VDN Override for ISDN Trunk ASAI Messages

This field appears if, on the **System-Parameters Customer-Options** screen, the **ASAI Link Core Capabilities** field is **y**. This field can be set to **y** only when the **Allow VDN Override** field on this screen is **y**.

Valid entries	Usage
y	When an incoming call routes, the "Called Number" information sent in the "Call Offered," "Altering," "Queued," and "Connect" ASAI events and the "Adjunct Route Request" ASAI message, is the "active VDN" extension associated with the routed call.
n	The "Called Number" information sent for the ASAI event notification and adjunct-request messages does not change for a ISDN-PRI trunk. It is always the number in the Called Number IE sent in the incoming ISDN call’s SETUP message.

VDN Timed ACW Interval

When a value is entered in this field, an agent in auto-in work mode who receives a call from this VDN is automatically placed into After Call Work (ACW) when the call drops. When the administered time is over, the agent automatically becomes available. This field has priority over the **Timed ACW Interval** field on the **Hunt Group** screen.

Valid entries	Usage
1 to **9999** or blank	Enter the number of seconds the agent should remain in ACW following the call.

Field descriptions for page 2

Figure 547: Vector Directory Number screen (if the Meet-me Conference field is y)

```
change vdn 5000                                              Page  2 of 2
                        VECTOR DIRECTORY NUMBER

                      MEET-ME CONFERENCE PARAMETERS
    Conference Access Code:
     Conference Controller:
```

Conference Access Code

This field appears only if the **Meet-me Conference** field is **y**.

SECURITY ALERT:
You should always assign an access code to a Meet-me Conference VDN.

Valid entries	Usage
6-digit number or blank	Enter a 6-digit access code for the Meet-me Conference VDN. If you do not want an access code, leave blank. Once an access code is assigned, an asterisk displays in this field for subsequent change, display, or remove operations by all users except the "init" superuser login.

Conference Controller

This field appears only if the **Meet-me Conference** field is **y**. This field controls which user is allowed to change the access code for a Meet-me Conference VDN using a feature access code. This can be a local user or someone dialing in via remote access trunks.

Valid entries	Usage
extension number or blank	If an extension number is entered, only a user at that extension can change the access code for that VDN using a feature access code. If this field is blank, any station user that is assigned with console permissions can change the access code for that VDN using a feature access code.

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