

Overview for Avaya Communication Manager

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About this book

What is the purpose of this book?

This book provides general information about the features and capabilities of Avaya Communication Manager. It also discusses practical and creative applications for the various platforms that run Communication Manager.

For details about changes for the most current software release, see *What's New in Avaya Communication Manager*, 03-300682.

Who should read this book?

This book is written for those who are considering the purchase of Communication Manager, and for Avaya representatives and distributors who need high-level information about the system and how it can be used.

What is in this book?

This book outlines all Communication Manager features and capabilities that are available world-wide.

Note:

Some products are unavailable in some countries. Please check with your local distributor for further information about what features are available to you.

Conventions

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 Figure 7: Optional Features screen on page 198 for an example of a screen and how it is shown in this book.
- Avaya uses the term “telephone” in this book. Other 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 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 bold 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. 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:

A note calls attention to neutral information or positive information that supplements the main text. A note also calls attention to valuable information that is independent of the main text.

Important:

An important note calls attention to situations that can cause serious inconvenience.

Tip:

A tip calls attention to information that helps you apply the techniques and the procedures that the text describes. A tip can include keyboard shortcuts, or alternative methods that might not be obvious.

CAUTION:

A caution statement calls attention to situations that can result in harm to software, loss of data, or an interruption of service.

WARNING:

A warning statement calls attention to situations that can result in harm to hardware or equipment.

DANGER:

A danger statement calls attention to situations that can result in physical injury to yourself or to other people.

SECURITY ALERT:

A security alert calls attention to situations that can increase the potential for toll fraud or other unauthorized use of your telecommunications system.

ELECTROSTATIC ALERT:

An electrostatic alert calls attention to situations that can result in damage to electronic components from electrostatic discharge (ESD).

Systems, circuit packs, and media modules

- The word “system” is a general term encompassing all references to an Avaya media server running Communication Manager.
- Circuit pack codes (for example, TN780 or TN2182B) are shown with the *minimum acceptable* alphabetic suffix (like the “B” in the code TN2182B). Generally, an alphabetic suffix higher than that shown is also acceptable. However, not every *vintage* of either the minimum suffix or a higher suffix code is necessarily acceptable. A suffix of “P” means that firmware can be downloaded to that circuit pack.
- The term “cabinet” refers to the external casing (shell) of an MCC1, SCC1, CMC1, G600, or G650 Media Gateway. Circuit packs are installed in the cabinet in a specific carrier (row), and in a specific slot within that carrier.
- The designation “***UUCSSpp***” refers to the location (address) of a circuit pack in cabinet-carrier-slot-port order. In this address designation, ***UU*** is the cabinet number, ***C*** is the carrier letter, ***SS*** is the slot number of a specific circuit pack, and ***pp*** (if applicable) is a specific port on the circuit pack. A sample address for port 4 on a circuit pack on an MCC1 Media Gateway might look like this: 02A0704.
- A G350 or G700 Media Gateway uses media modules instead of circuit packs. The media module address is designated as ***XXX*****V*****Spp***, where ***XXX*** is the administered number of the media gateway, **V*****S*** is the slot number of a specific media module location on the media gateway, and ***pp*** (if applicable) is a specific port on the media module. The **V** is not a variable and needs to be included in the command exactly where shown. A sample address for port 4 in slot V3 on an MM711 Media Module on a G700 Media Gateway might look like this: 002V304. If an S8300 Media Server is installed in a G700 Media Gateway, it must be installed in slot number V1.

Trademarks

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Security concerns

Toll fraud is the theft of long distance service. When toll fraud occurs, your company is responsible for charges. For information on how to prevent toll fraud, see the *Avaya Toll Fraud and Security Handbook,* 555-025-600. You can also call the Avaya Security Hotline at 1 800 643 2353, or contact your Avaya representative.

Related books

The *Administrator Guide for Avaya Communication Manager*, 03-300509, explains system features and interactions in greater detail. The *Administrator Guide for Avaya Communication Manager* provides a reference how to plan, operate, and administer your system.

Note:

Prior to April 1997, this same information was in two separate books: the *DEFINITY Implementation* and the *DEFINITY Feature Description* books.

You might also find additional information in the following books:

- *Avaya Toll Fraud and Security Handbook*, 555-025-600
- *Hardware Description and Reference for Avaya Communication Manager*, 555-245-207
- *What's New in Avaya Communication Manager*, 03-300682
- *Feature Description and Implementation of Avaya Communication Manager*, 555-245-205.

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How to get books on the Web

If you have internet access, you can view and download the latest version of the *Overview for Avaya Communication Manager*. To view this book, you must have a copy of Acrobat Reader.

Note:

If you do not have Acrobat Reader, you can get a free copy at http://www.adobe.com.

To get the latest version of this book:

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- Within the United States, click the **Escalation Contacts** link that is located under the **Support Tools** heading. Then click the appropriate link for the type of support that you need.
- Outside the United States, click the **Escalation Contacts** link that is located under the **Support Tools** heading. Then click **International Services**, which includes telephone numbers for the international Centers of Excellence.

You can also access the following services in the USA. You might need to purchase an extended service agreement to use some of these services. Contact your local Avaya authorized dealer for any additional help and questions.

Avaya Communication Manager Helpline (for help with feature administration and system applications)	1 800 225 7585
Avaya National Customer Care Center Support Line (for help with maintenance and repair)	1 800 242 2121
Avaya Toll Fraud Intervention	1 800 643 2353
Avaya Corporate Security	1 800 822 9009

Chapter 1: Overview of Avaya Communication Manager

Avaya Communication Manager organizes and routes voice, data, image and video transmissions. It can connect to private and public telephone networks, ethernet LANs, ATM networks, and the Internet.

Communication Manager seeks to solve business challenges by powering voice communications and integrating with value-added applications. Communication Manager is an open, scalable, highly reliable and secure telephony application. Communication Manager provides user and system management functionality, intelligent call routing, application integration and extensibility, and enterprise communications networking Figure 1: System running Avaya Communication Manager on page 31.

Figure 1: System running Avaya Communication Manager

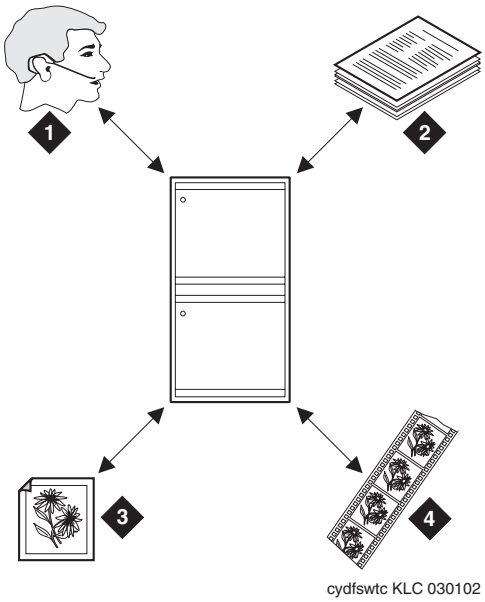

Figure notes:

1.	Voice	**2.**	Data	**3.**	Image	**4.**	Multimedia

Optional software

Various optional packages can enhance the capabilities of your system. Some of the capabilities described in this document require optional software. See your Avaya account representative for more information.

Capacities

System capacities have been expanded for many products and features. However, the most up-to-date system capacity information is not listed in Communication Manager documentation.

For the entire list of updated capacities, see the *Avaya Communication Manager System Capacities Table*, 555-245-601.

To view the system capacity limits:

1. Go to the Avaya Web site at http://www.avaya.com/.
2. In the **Get Support** section, click the word **Documentation**.
3. In the **Search** text box, type **555-245-601** or the words **system capacities table**. Click the arrow.
4. Locate the latest version of the system capacities table document, and then click the title of the document to download the information.

Avaya Installation Wizard

The Avaya Installation Wizard is a tool to be used in installations (not upgrades) of Communication Manager in certain system configurations. The Installation Wizard helps with reduced complexity, time-to-install, and the cost of installation.

The Avaya Installation Wizard does not work with all media gateways and media servers. For more information, contact your Avaya representative.

CAUTION:

The Avaya Installation Wizard and the web installation that is accessible from the Avaya maintenance Web pages should not be run at the same time. Make sure you complete one process before you start the other process.

The Installation Wizard delivers the following installation advantages:

- Intuitive user interface with on-line help
- Auto-discovery, where appropriate
- No assumption of external internet connectivity
- Ease of updating to newest software & firmware
- Ability to import customized name & number list
- Complete record of all settings
- Guided process from beginning to end

The Installation Wizard can guide installers through:

- License file and authentication file setup
- Media server & media gateway configuration
- Telephony, trunk, and endpoint configuration and installation
- Installation log file summary creation

The Installation Wizard for Communication Manager has these features:

- The Installation Wizard supports a stack of up to 10 G700 Media Gateways.
- Technicians are able to load updated media module firmware versions from their laptop as part of the Installation Wizard process.
- Installation of the BRI Media Module is supported.
- The Installation Wizard supports installation of a G700 Media Gateway with a Local Survivable Processor (LSP).
- Remote G700s without an Internal Call Controller (ICC) Media Module can be configured using the Installation Wizard by temporarily installing a spare ICC Media Module in the G700 Media Gateway until the end of the installation process.
- Provide an Electronic pre-Installation Worksheet (EIW) to automate the task of importing selected pre-installation data. This capability is focused on importing IP address information.
- A customization template to allow for the selective customization of defaulted translation data.
- Support configuration of the IA770 module.
- Support for Japan, United Kingdom, and France, including system and trunk level parameters. May be extended to Australia and other countries prior to the next release of Communication Manager.
- Support configuration of IP trunks.

- Support trunk diagnostics.
- Support IP address configuration of distributed G700 gateways through the Gateway Installation Wizard (GIW).

Updates to the Avaya Installation Wizard are available on the Web, and are not necessarily linked to any software release of Communication Manager. The latest version of Avaya Installation Wizard can be downloaded from http://support.avaya.com/avayaiw.

Gateway Installation Wizard

The Gateway Installation Wizard (GIW) is a separate application that assists in installing and configuring certain media gateways. For more information, see your Avaya representative.

Updates to GIW are available on the Web, and are not necessarily linked to any software release of Communication Manager. The latest version of GIW can be downloaded from http://support.avaya.com/avayaiw.

Avaya media servers and media gateways

For information about any of the media servers or media gateways that can run Communication Manager, see the *Hardware Description and Reference for Avaya Communication Manager*, 555-245-207.

Also see the capacities table for the entire list of updated capacities. The most up-to-date system capacity information is not listed in Communication Manager documentation. Instead, this information is available online. See Capacities on page 32 for instructions how to locate the capacities table.

Multi-Tech gateway support

Communication Manager supports a voice over IP (VoIP) gateway from Multi-Tech Systems, Inc., a third-party vendor. Any system that is running Communication Manager can connect and run this Multi-Tech gateway.

For more information, see the *Hardware Description and Reference for Avaya Communication Manager*, 555-245-207.

Chapter 2: Application programming interface

An application programming interface (API) allows numerous software applications to work with Avaya Communication Manager. APIs also allow a client programmer to create their own applications that work with Communication Manager.

Application Enablement Services

Application Enablement Services (AE Services) is a connector that provides connectivity between applications and Communication Manager. This connector allows development of new applications and new features without having to modify Communication Manager or expose its proprietary interfaces.

AE Services provides a single common platform architecture for call control, device control, media control, and management. AE Services enables internal Avaya developers and external partners to create powerful applications that harness the extensive Communication Manager feature set.

Note:

AE Services has its own set of customer documentation, including an overview. This *Overview of Avaya Communication Manager* does not outline the changes to AE Services. To obtain AE Services documentation on the Web, see How to get books on the Web on page 28. After you click the **Documentation** link, click the letter **A**, then click the **Application Enablement Services** link.

Avaya provides two different AE Services deployment options:

- Software-only option
- Bundled server option

The same client applications and software development kits (SDK) can run against both options.

Software-only option

Avaya provides the AE Services software, which is the AE Services connector server software and the AE Services SDK. The customer obtains the prerequisite hardware, platform software, and third-party software. The customer then installs and maintains all software and hardware.

AE Services requires an AE Services license file. The license file can only be accessed by Avaya Services or by an Avaya BusinessPartner.

Bundled server option

Avaya provides the hardware server and all of the necessary software:

- AE Services connector server software
- AE Services SDK
- Platform
- Third-party software

Avaya service technicians install the hardware and software. If the customer buys a maintenance and/or a service contract, Avaya also provides maintenance and/or service for the system. Otherwise, the customer maintains the system.

CVLAN

CallVisor LAN (CVLAN) is an application programming interface (API) that enables applications to communicate with Communication Manager. CVLAN sends and receives ASAI messages over shared ASAI links on TCP/IP. An application can use ASAI messages to monitor and control Communication Manager resources.

CVLAN software consists of a client component and a server component. The CVLAN client can be installed on a server or on a client workstation. The CVLAN client provides clients with access to the switch using the CVLAN server.

Web services

Telephony Service

Telephony Service (TS) is a web service that exposes basic outbound call control features of Communication Manager. Telephony Service enables its clients to originate an outbound call, drop a call, transfer a call, or conference a party into a call.

Telephony Service is one of the web services that resides on the Application Enablement Services platform (AE Services).

System Management Service

System Management Service (SMS) is a web service that exposes management features of Communication Manager to clients. SMS enables its clients to display, list, add, change, and remove specific managed objects on Communication Manager that are available through the OSSI protocol and SAT screens.

SMS is one of the web services that resides on the Application Enablement Services platform (AE Services).

User Service

User Service provides a common way of administering, retrieving, and programmatically operating on user data. User Service provides a common user store and a programmable interface for products and applications with which to integrate. User Service has a common industry-standard data store (LDAP) as the repository for common user profile data.

User Service has web services as the infrastructure. This infrastructure allows products to integrate with User Service at your schedule. User Service exposes a programmatic SOAP interface that allows clients to write third party applications to utilize its functionality.

This integration occurs through the use of software adapters to User Service. The adapter and web services technology allows User Service to publish user events to the product spaces, and the product spaces to publish events to the common user area.

So if an administrator adds a user to the common store, a user event is sent to all participating products with the appropriate information. Likewise, if a product level administrator modifies a user record in its own user system, an event is sent to User Service for the modified data to be stored in the common store. User Service then relays this user event to the other participating product areas.

Device and media control API

Device and media control API provides a connector to Communication Manager that allows clients to develop applications that provide first party call control. Applications can register as IP extensions on Communication Manager and then monitor and control those extensions.

Device and media control API consists of connector server software and a connector client API library. The connector server software runs on a hardware server that is independent from Communication Manager. That is, device and media control API does not run co-resident with Communication Manager.

Ask your Avaya representative for a complete list of device and media control API documentation.

DEFINITY LAN Gateway

DEFINITY LAN Gateway (DLG) is a software service that tunnels the ASAI call control protocol messages onto IP packets for transport between a customer Computer Telephony Integration (CTI) server or application and Communication Manager.

Adjunct switch application interface

Adjunct Switch Application Interface (ASAI) links Communication Manager and adjunct applications. The interface allows adjunct applications to access Communication Manager features and supply routing information to the system.

ASAI is the Avaya recommendation for Computer Telephony Integration (CTI). ASAI is based on the Q.932 protocol.

JTAPI

Java telephony application programming interface (JTAPI) is an open API supported by Avaya computer telephony that enables integration to Communication Manager ASAI. It is an object-oriented programming interfaces favored for the development of multimedia solutions. JTAPI applications are supported on any clients that supports a JAVA virtual machine (this includes Windows, UnixWare, and Solaris platforms), or a Java-compatible Web browser.

TSAPI

Telephony Services Application Programming Interface (TSAPI) is an open API supported by Avaya computer telephony that allows integration to Communication Manager ASAI.

TSAPI is based on international standards for CTI telephony services. Specifically, the European Computer Manufacturers Association (ECMA) CTI standard definition of Computer-Supported Telecommunications Applications (CSTA) is the foundation for TSAPI. The CSTA standard is a technical agreement reached by an open, multi-vendor consortium of major switch and computer vendors. Since CSTA Services and protocol definitions are the basis for TSAPI, TSAPI provides a generic, switch-independent API. CSTA services logically integrate the two most common pieces of equipment on user desktops, the telephone and the personal computer.

Security administration for telephony services allows administrators to restrict user access to TSAPI features in various ways. For example, an administrator might restrict a user to control and monitoring of the telephone at their desktop. Similarly, an administrator can restrict a user to call control and monitoring of the telephone at any desktop where they log in.

Expanded security permissions can increase user control in support of work group or departmental telephony applications. Administrators can expand user permissions even further to include any telephone or device that it is possible to control on a CTI link. An administrator might assign an unrestricted security permission level to a server application that processes calls before call delivery to user desktops in a call center environment. An administrator can assign different users different permissions.

Chapter 3: Attendant features

Avaya Communication Manager contains many exciting features that provide easy ways to communicate through your telephone system attendant (operator). In addition, attendants can connect to their console (switchboard) from other telephones in your system, thereby expanding the attendant capabilities.

Accessing the attendant

Dial access to attendant

The dial access to attendant feature allows you to reach an attendant by dialing an access code. The attendant can then extend the call to a trunk or to another telephone.

Individual attendant access

Individual attendant access allows you to call a specific attendant console. Each attendant console can be assigned an individual extension number.

Recall

This feature allows users to recall the attendant when they are on a two-party call or on an attendant conference call held on the console.

- Single-line users press the recall button or flash the switchhook to recall the attendant.
- Multi-appearance users press the conference or transfer button to recall the attendant and remain on the connection when either button is used.

Attendant backup

The attendant backup feature allows you to access most attendant console features from one or more specially-administered backup telephones. This allows you to answer calls more promptly, thus providing better service to your guests and prospective clients.

When the attendant console is busy, you can answer overflow calls from the backup telephones by pressing a button or dialing a feature access code. You can then process the calls as if you are at the attendant console. The recommended backup telephones are the Avaya models 6408, 6416, or 6424.

Attendant room status

Communication Manager allows an attendant to see whether a room is vacant or occupied, and what the housekeeping status of each room is. This feature is available only when you have enhanced hospitality enabled for your system (see Hospitality on page 89).

This feature combines the property management capabilities of housekeeping status and check-in/check-out, but does not require that you have a property management system (PMS).

Attendant functions using Distributed Communications System protocol

Control of trunk group access

Control of trunk group access allows an attendant at any node in the Distributed Communications System (DCS) to take control of any outgoing trunk group at an adjacent node. This is helpful when an attendant wants to prevent telephone users from calling out on a specific trunk group for any number of reasons, such as reserving a trunk group for incoming calls or for a very important outgoing call.

Direct trunk group selection

Direct trunk group selection allows the attendant direct access to an idle outgoing trunk in a local or remote trunk group by pressing the button assigned to that trunk group. This feature eliminates the need for the attendant to memorize, or look up, and dial the trunk access codes associated with frequently used trunk groups. Direct trunk group selection is intended to expedite the handling of an outgoing call by the attendant.

Inter-PBX attendant calls

Inter-PBX attendant calls allows attendants for multiple branches to be concentrated at a main location. Incoming trunk calls to the branch, as well as attendant-seeking voice-terminal calls, route over tie trunks to the main location.

Call handling

Attendant Intrusion

Use the Attendant Intrusion feature to allow an attendant to intrude on an existing call. The Attendant Intrusion feature is also called Call Offer.

Attendant lockout - privacy

This feature prevents an attendant from re-entering a multiple-party connection held on the console unless recalled by a telephone user. This feature is administered on a system-wide basis. It is either activated or not activated.

Attendant split swap

The attendant split swap feature allows the attendant to alternate between active and split calls. This operation may be useful if the attendant needs to transfer a call but first must talk independently with each party before completing the transfer.

Attendant vectoring

Attendant vectoring provides a highly flexible approach for managing incoming calls to an attendant. For example, with current night service operation, calls redirected from the attendant console to a night station can ring only at that station and will not follow any coverage path.

With attendant vectoring, night service calls will follow the coverage path of the night station. The coverage path could go to another station and eventually to a voice mail system. The caller can then leave a message that can be retrieved and acted upon.

Automated attendant

Automated attendant allows the calling party to enter the number of any extension on the system. The call is then routed to the extension. This allows you to reduce cost by reducing the need for live attendants.

Backup alerting

The backup alerting feature notifies backup attendants that the primary attendant cannot pick up a call. It provides both audible and visual alerting to backup stations when the attendant queue reaches its queue warning level. When the queue drops below the queue warning level, alerting stops.

Audible alerting also occurs when the attendant console is in night mode, regardless of the attendant queue size.

Call waiting

Call waiting allows an attendant to let a single-line telephone user who is on the telephone know that a call is waiting. The attendant is then free to answer other calls. The attendant hears a call waiting ringback tone and the busy telephone user hears a call waiting tone. This tone is heard only by the called telephone user.

Calling of inward restricted stations

A telephone with a class of restriction (COR) that is inward restricted cannot receive public network, attendant-originated, or attendant-extended calls. This feature allows you to override this restriction.

Conference

The conference feature allows an attendant to set up a conference call for as many as six conferees, including the attendant. Conferences from inside and outside the system can be added to the conference call.

Starting with Communication Manager release 3.0, attendants can set up conferences for more than six people using the Enhanced Meet-me Conferencing feature. For more information, see Expanded Meet-me Conferencing on page 77.

Listed directory number

Allows outside callers to access your attendant group in two ways, depending on the type of trunk used for the incoming call. You can allow attendant group access through incoming direct inward dial trunks, or you can allow attendant group access through incoming central office and foreign exchange trunks.

Override of diversion features

The override of diversion feature allows an attendant to bypass diversion features such as send all calls and call coverage by putting a call through to an extension even when these diversion features are on. This feature, together with attendant intrusion, can be used to get an emergency or urgent call through to a telephone user.

Priority queue

Priority queue places incoming calls to the attendant in an orderly queue when these calls cannot go immediately to the attendant. This feature allows you to define twelve different categories of incoming attendant calls, including emergency calls, which are given the highest priority.

Release loop operation

Release loop operation allows the attendant to hold a call at the console if the call cannot immediately go through to the person being called. A timed reminder begins once the call is on hold. If the call is not answered within the allotted time, the call returns to the queue for the attendant. Timed reminders attempt to return the call to the attendant who previously handled it. Only when the original attendant is unavailable are calls returned to the queue.

Selective conference mute

See Selective conference mute on page 79.

Serial calling

The serial calling feature enables an attendant to transfer trunk calls that return to the same attendant after the called party hangs up. The returned call can then transfer to another station within the switch. This feature is useful if trunks are scarce and direct inward dialing services are unavailable. An outside caller may have to redial often to get through because trunks are so busy. Once callers get through to an attendant they can use the same line into the switch for multiple calls. The attendant display shows if an incoming call is a serial call.

Timed reminder and attendant timers

Attendant timers automatically alert the attendant after an administered time interval for the following types of calls:

- Extended calls to be answered or waiting to be connected to a busy single-line telephone
- One-party calls placed on hold on the console
- Transferred calls that have not been answered after transfer

The timed reminder feature informs the attendant that a call requires additional attention. After the attendant reconnects to the call, the user can either choose to try another extension number, hang up, or continue to wait. Communication Manager supports a variety of administrable attendant timers for use in a variety of situations.

Centralized Attendant Service

Centralized Attendant Service (CAS) enables attendant services in a private network to be concentrated at a central location. Each branch in a centralized attendant service has its own listed directory number or other type of access from the public network. Incoming calls to the branch, as well as calls made by users directly to the attendants, are routed to the centralized attendants over release link trunks.

Display

The display feature shows call-related information that helps the attendant to operate the console. This feature also shows personal service and message information. Information is shown on the alphanumeric display on the attendant console. Attendants may select one of several available display message languages: English, French, Italian, or Spanish. In addition, your company may define one additional language for use by users and attendants on their display.

Making calls

Auto Start and Do Not Split

The Auto Start feature allows the attendant to make a telephone call without pushing the start button first. If the attendant is on an active call and presses digits on the keypad, the system automatically splits the call and begins dialing the second call.

The Do Not Split feature deactivates the auto start feature and allows the sending of touch tones over the line for the purposes of such things as picking up messages.

Auto Manual Splitting

Auto Manual Splitting allows an attendant to announce a call or consult privately with the called party without being heard by the calling party on the call. It splits the calling party away so the attendant can confidentially determine if the called party can accept the call.

Monitoring calls

Attendant control of trunk group access

Use the Attendant Control of Trunk Group Access feature to allow the attendant to control outgoing and two-way trunk groups. The attendant usually activates this feature during periods of high use. This is helpful when an attendant wants to prevent telephone users from calling out on a specific trunk group. Some reasons are to reserve a trunk group for incoming calls or for a very important outgoing call.

This feature also prevents telephone users from directly accessing an outgoing trunk group that the attendant has controlled.

Attendant direct extension selection

This feature allows the attendant to keep track of extension status - whether the extension is idle or busy - and to place or extend calls to extension numbers without having to dial the extension number. The attendant can use this feature in two ways:

- using standard direct extension selection access
- using enhanced direct extension selection access

Attendant direct trunk group selection

With this feature, the attendant directs access to an idle outgoing trunk by pressing the button assigned to the trunk group. This feature eliminates the need for the attendant to memorize, or look up, and dial the trunk access codes associated with frequently used trunk groups. Pressing a labelled button selects an idle trunk in the desired group.

Crisis alerts to an attendant console

Crisis alert uses both audible and visual alerting to notify attendant consoles when an emergency call is made. Audible alerting sounds like an ambulance siren. Visual alerting flashes the CRSS-ALRT button lamp and the display of the caller name and extension (or room). The display of the origin of the emergency call enables the attendant or other user to direct emergency service response to the caller. Though often used in the hospitality industry, it can be set up to work with any standard attendant console.

When crisis alerting is active, the console is placed in position-busy mode so that other incoming calls can not interfere with the emergency call notification. The console can still originate calls to allow notification of other personnel. Once a crisis alert call has arrived at a console, the console user must press the position-busy button to unbusy the console, and press the crisis-alert button to deactivate audible and visual alerting.

If an emergency call is made while another crisis alert is still active, the incoming call will be placed in the queue. If the system is administered so that all users must respond, then every user must respond to every call, in which case the calls are not necessarily queued in the order in which they were made. If the system is administered so that only one user must respond, the first crisis alert remains active at the telephone where it was acknowledged. Subsequent calls are queued to the next available station in the order in which they were made.

Trunk group busy/warning indicators to attendant

This feature provides the attendant with a visual indication that the number of busy trunks in a group has reached an administered level. A visual indication is also provided when all trunks in a group are busy. This feature is particularly helpful to show the attendant that the attendant control of trunk group access feature needs to be invoked.

Trunk identification by attendant

Trunk identification allows an attendant or display-equipped telephone user to identify a specific trunk being used on a call. This capability is provided by assigning a trunk ID button to the attendant console or telephone. This feature is particularly helpful for identifying a faulty trunk. That trunk can then be removed from service and the problem quickly corrected.

Visually Impaired Attendant Service

Visually Impaired Attendant Service (VIAS) provides voice feedback to a visually impaired attendant. Each voice phrase is a sequence of one or more single-voiced messages. This feature defines six attendant buttons to aid visually impaired attendants:

- Visually impaired service activation/deactivation button: activates or deactivates the feature. All ringers previously disabled (for example, recall and incoming calls) become reenabled.
- Console status button: voices whether the console is in position available or position busy state, whether the console is a night console, what the status of the attendant queue is, and what the status of system alarms is.

- Display status button: voices what is shown on the console display. VIAS support is not available for all display features (for example, class of restriction information, personal names, and some call purposes).
- Last operation button: voices the last operation performed.
- Last voiced message button: repeats the last voiced message.
- Direct trunk group selection status button: voices the status of an attendant-monitored trunk group.

The visually impaired attendant may use the Inspect mode to locate each button and determine the feature assigned to each without actually executing the feature.

Chapter 4: Call center

The Avaya call center provides a fully integrated telecommunications platform that supports a powerful assortment of features, capabilities, and applications designed to meet all of your customers' call center needs.

For a more complete description of Call Center features for various releases of Communication Manager, see the following documents:

- Avaya Call Center Change Description
- Avaya Call Center ACD Guide
- Avaya Call Center Call Vectoring and EAS Guide
- Avaya Business Advocate

Computer Telephony Integration

Computer Telephony Integration (CTI) enables Communication Manager features to be controlled by external applications, and allows integration of customer databases of information with call control features.

Avaya Computer Telephony (formally named CentreVu™ Computer Telephony) is server software that integrates the premium call control features of Communication Manager with customer information in customer's databases. It is a local area network (LAN)-based CTI solution consisting of server software that runs in a client/server configuration. Avaya Computer Telephony delivers the CTI architecture and platform that supports contact center application requirements, along with emerging applications programming interfaces (APIs).

Adjunct route support for network call redirection

This feature provides the capability to invoke Network Call Redirection (NCR) through the route request response to an adjunct route vector step. This allows a CTI application to directly utilize NCR for redirecting an incoming call in the PSTN through the ASAI adjunct routing application.

The redirection request, along with the PSTN redirected to a telephone number, is included in the route select message from the adjunct. The redirect request invokes whatever form of network redirection that is assigned to the trunk group for the incoming call in the same manner as a vector invoked NCR. Information forwarding to the redirected destination is supported in the same manner as a vector invoked NCR.

This capability functions with either the network transfer type where the switch sets up the 2nd leg of a call, or the network deflection type where the PSTN sets up the 2nd leg of a call of NCR protocols.

Co-resident DEFINITY LAN Gateway

In simplest terms, the DEFINITY Local Area Network (LAN) Gateway, or DLG, is an application that enables communications between TCP/IP clients and Communication Manager call processing. In more technical terms, the DLG application is software that both routes internetwork messages from one protocol to another (ISDN to TCP/IP) and bridges all ASAI message traffic by way of a TCP/IP tunnel protocol.

In previous configurations, a DEFINITY LAN gateway (DLG) was connected externally on a separate TN801 MAPD circuit pack. The DLG application is packaged internally where it co-resides with the Communication Manager. The internally packaged DLG is referred to as the co-resident DLG.

Co-resident DLG is only available with the S8300 Media Server.

Co-resident DLG provides the functionality of the Adjunct/Switch Application Interface (ASAI) using an ethernet transport instead of a Basic Rate Interface (BRI) transport. In the S8300 Media Server, connectivity is provided through the processor ethernet.

For more information on co-resident DLG and the G700 Media Gateway, see chapters “DEFINITY LAN Gateway and ASAI-Ethernet,” and “Installation and Test for CallVisor ASAI,” in the *Avaya MultiVantage Software CallVisor ASAI Technical Reference*, 555-230-220.

Also see the following documents:

- *DEFINITY Enterprise Communications Server CallVisor ASAI Applications Over MAPD,* 555-230-136
- *Installation for Adjuncts and Peripherals for Avaya Communication Manager*, 555-233-116.

Direct Agent Announcement

Direct Agent Announcement (DAA) enhances direct agent calling capabilities for Adjunct Switch Application Interface (ASAI) and Expert Agent Selection (EAS). It plays an announcement to direct agent callers waiting in a queue.

Flexible billing

The flexible billing feature allows Communication Manager or an adjunct to communicate with the public network using ISDN PRI messages to change the billing rate for an incoming 900-type call. Rate-change requests to specify a new billing rate can be made anytime after a call is answered and before it disconnects.

Flexible billing is available in the U.S. for use with AT&T MultiQuest 900 Vari-A-Bill service. Flexible billing requires an adjunct switch application interface and other application software.

Pending work mode change

This feature allows ASAI applications to change the current work mode of an agent while that agent is busy on a call. The change is a pending change that will take effect as soon as all the current calls are cleared.

Trunk group identification

Trunk group identification provides ASAI applications with the capability to obtain trunk group information even when the Calling Party Number (CPN) is known. ASAI will provide the trunk group information in the event reports for both inbound and outbound calls. If the Automatic Number Identification (ANI) is known, the event reports will contain the trunk group information and the CPN.

User-to-User Information propagation during manual transfer/conference operations

This feature enables UUI, specifically used by ASAI, to be propagated to the new call during a manual transfer or conference operation. Previously, ASAI UUI could not be sent in a setup message when the call was transferred to another system, so the ASAI UUI was never passed to an application monitoring calls on the system receiving the transfer.

This feature only applies to manual transfer and conference operations. If the transfer or conference operation is controlled by a software application (for example, controlling calls or agents over an ASAI link), the application can insert the desired ASAI UUI into the new call.

Block CMS Move Agent events

This feature lets you prevent the system from sending the ASAI logout-login event messages, that are related to an agent move. When this CTI link option is activated, the changes to the agent state, such as logout followed by login and return to previous state, will not be reported to the ASAI adjunct. This operation is required by Avaya IC since the initial logout causes IC to permanently logout the agent, disrupting normal operation. IC does not need to be informed of agent skill moves via this method. This option will be available to other applications for use where needed.

VDN override for ASAI messages

This feature provides a VDN option to override the called number in certain ASAI messages for ISDN calls. This applies to CTI applications that require the active VDN extension instead of the called number. This is a field on page 2 of the VDN Screen - “VDN Override for ISDN Trunk ASAI Messages. The default value is **no**.

For calls to VDNs with the option set to y(es), the called number provided will correspond to the active VDN for call instead of the original called number provided in the incoming ISDN SETUP message. This applies to the ASAI call-offered, alerting, queued and connect event messages and the adjunct route-request message.

Automatic Call Distribution

Automatic Call Distribution (ACD) is the basic building block for call center applications. ACD offers you a method for distributing incoming calls efficiently and equitably among available agents. With ACD, incoming calls can be directed to the first idle or most idle agent within a group of agents.

Agents in an ACD environment are assigned to a hunt group, a group of agents handling the same types of calls. A hunt group is also known as a split or skill with Expert Agent Selection (EAS).

A hunt group is especially useful when you expect a high number of calls to a particular telephone extension. A hunt group might consist of people trained to handle calls on specific topics. For example, the group might be:

- A benefits department within your company
- A service department for products you sell
- A travel reservations service
- A pool of attendants

In addition, a hunt group might consist of a group of shared telecommunications facilities. For example, the group might be:

- A modem pool
- A group of data-line circuit ports
- A group of data modules

In the following example (Figure 2: A basic example of automatic call distribution on page 55), hunt group “A” receives calls only when agents are available since it has no queue. Calls to hunt group “B” can be queued while agents are unavailable, and redirected to hunt group “C” if not answered within an administrable time. Calls to hunt group “C” are redirected to voice mail if not answered within an administrable time.

Figure 2: A basic example of automatic call distribution

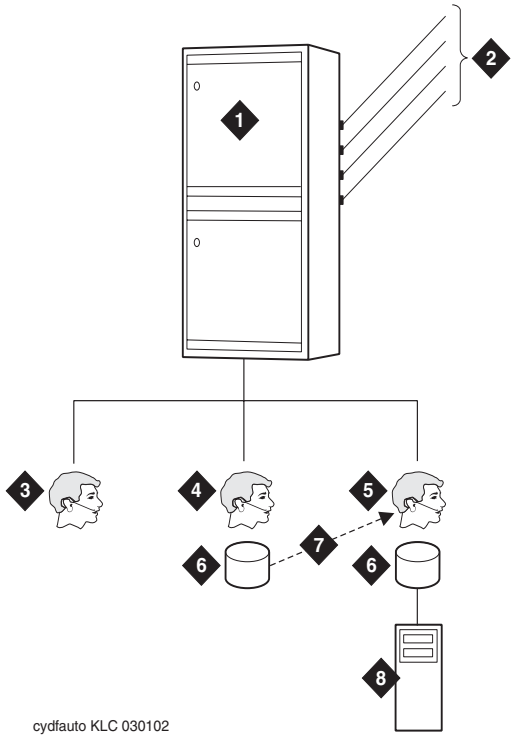

Figure notes:

1.	System running Avaya Communication Manager	**5.**	Hunt group C: general information
2.	Incoming lines	**6.**	Queues
3.	Hunt group A: business travel	**7.**	Call coverage to hunt group C
4.	Hunt group B: personal travel	**8.**	Voice mail

Abandoned Call Search

Abandoned Call Search allows a central office that does not provide timely disconnect supervision to identify abandoned calls. An abandoned call is one in which the calling party hangs up before the call is answered. Abandoned Call Search is suitable only for older central offices that do not provide timely disconnect supervision.

Adjunct Routing

Adjunct Routing is a vector step that, when executed, sends a route request over the specified link to the connected adjunct asking where to route the call being processed. The adjunct is then to respond with a route-select message specifying the destination either internal or outside number where the call is to be routed. Adjunct Routing is used in conjunction with ASAI.

Auto-Available Split

Auto-Available Split (AAS) allows members of an Automatic Call Distribution (ACD) split to be continuously in auto-in work mode. An agent in auto-in work mode becomes available for another ACD call immediately after disconnecting from an ACD call. You can use AAS to bring ACD-split members back into auto-in work mode after a system restart.

Although not restricted to such, this feature is intended to be used for splits containing only recorders or voice-response units.

Automatic Number Identification

Use the Automatic Number Identification (ANI) feature to display telephone number of the calling party on your display telephone. The system uses ANI to interpret calling party information that is signaled over multifrequency (MF) or Session Initiation Protocol (SIP) trunks.

Incoming Automatic Number Identification

Use inband signaling for information, such as the address digits for the called party, that is delivered over the same trunk circuit that is used for the voice or data connection. Use out-of-band or ISDN signaling when signaling information passes through a different signaling path than the path that is used for the voice or data connection.

For example, when a call is made from 555-3800 to your display telephone at extension 81120, and the **Incoming Tone (DTMF) ANI** field is set to ***ANI*DNIS*** on the **Trunk Group** screen, your trunk group receives *5553800*81120*. If the same field is set to **ANI*DNIS***, your trunk group receives 5553800*81120*. In both cases, *call from 555-3800* appears on your telephone display.

If you do not use inband ANI, the incoming trunk group name appears on your telephone display.

Outgoing Automatic Number Identification

Outgoing ANI applies to outgoing Russian MF ANI, R2-MFC ANI, China #1 MF ANI, and Spain Multi Frequency España (MFE) ANI trunks only.

Use Outgoing ANI to specify the type of ANI to send on outgoing calls. You can define MF ANI (the calling party number, sent through multifrequency signaling trunks) prefixes by COR. This allows a system to send different ANIs to different central offices (COs).

For a tandem call that uses different types of incoming and outgoing trunks, the server uses:

- The COR-assigned call type of the incoming trunk for Russian or R2-MFC outgoing trunks
- Automatic Route Selection (ARS) call types for MFE outgoing trunks

Local feedback for queued ACD calls

A cost saving trend used by many call centers is the movement of agent seats from locations in the US and EU to offshore locations. One detriment to achieving these savings is the increase in trunk costs by redirecting calls to these offshore locations.

When a call is rerouted to an alternate switch, it becomes the responsibility of the destination switch to provide audible feedback to the caller while that call remains in queue at the destination switch waiting for an available agent. Typically, such audible feedback takes the form of music interspersed with recorded announcements.

When the trunks between the sending and receiving switches are IP trunks, bandwidth is utilized when the music and recorded announcement packets are sent from the destination switch to the caller. Because of the continuous nature of music, the bandwidth required to provide this audible feedback to callers in queue is generally greater than that required to support a conversation between a caller and an agent.

Communication Manager allows vector processing to continue at the local sending switch, even after a call has been routed to a queue on an offshore destination switch. Vector processing at the sending switch can then continue to provide audible feedback to the caller while the call is in queue at the destination switch. No packets need be sent over the IP trunk during the queuing phase of the call.

Queue status indicators

Communication Manager allows you to assign queue status indicators for ACD calls based on the number of calls in queue and the time in queue. To help monitor queue activity, you can assign these indications to lamps on agent, supervisor, or attendant terminals, or on consoles.

In addition, you can define auxiliary queue warning lamps to track queue status. On display telephones, you can display the number of calls in queue, and the time in queue of the oldest call.

Avaya Basic Call Management System

The Avaya Basic Call Management System (BCMS) helps you fine tune your call center operation by providing reports with the data necessary to measure your call center agents performance.

The BCMS feature offers call management control and reporting at a low cost for call centers of up to 2000 agents. BCMS collects and processes ACD call data (up to seven days) within the system; an adjunct processor is not required to produce call management reports.

The following are the types of reports that can be generated:

- Real-time reports, such as:
  - Agent status
  - System status
  - Vector directory number status
- Historical reports, such as:
  - Agent
  - Agent summary
  - Split
  - Split summary
  - Trunk group
  - Vector directory number

Avaya Business Advocate

Avaya Business Advocate is the collection of features that provide flexibility in the way a call is selected for an agent in a call surplus situation, and in the way an agent is selected for a call. Instead of the traditional “first in, first out” approach, the needs of the caller, potential business value, and the desire to wait are calculated. The system then decides what agents should be matched to the callers.

Auto reserve agents

Auto reserve agents allows the system to use the percent allocation distribution feature for agent skills.

Call selection override per skill

Call selection override is determined by skill. Call center supervisors can override the normal call handling activity either on particular skills only, or for the entire call center.

Dynamic percentage adjustment

The dynamic percentage adjustment feature allows the system to compare actual levels of service with service targets. The system can then adjust the service target so that the overall use of the skill is more efficient.

Dynamic queue position

Dynamic queue position allows the system to put calls from multiple vector directory numbers (VDNs) into a skill queue. The calculation is based on the ratio of ASA for the VDNs being equal to the ratio of service objectives for the VDNs. This feature ensures balanced call handling across VDNs.

Dynamic threshold adjustment

Dynamic threshold adjustment allows the system to compare actual levels of service with service targets, and to adjust overload thresholds. This feature makes the use of overload agents more efficient.

Logged-in advocate agent counting

The logged-in advocate agent counting feature counts agents toward the advocate agent limit if a service objective, percent allocation, or a reserved skill is assigned to the agent login ID, or if one of the agent skills is assigned least occupied agent or service level supervisor.

Percent allocation distribution

Percent allocation distribution allows the system to distribute calls to auto reserve agents by comparing a reserve agent work time in a skill with the target allocation for that skill.

Reserve agent time in queue activation

This feature activates a reserve agent either if the expected wait time (EWT) exceeds a pre-determined threshold, or if the call time in the queue exceeds the administered service level supervisor threshold. Reserve agents are then dropped off a skill only when both of the following conditions are met:

- The EWT for the skill drops below both administered thresholds.
- The head call time in queue no longer exceeds the service level supervisor threshold.

Avaya call center features supported on the Avaya G700 Media Gateway

Avaya Call Center functionality is supported on the G700 Media Gateway with Communication Manager, with either an S8300 Media Server or an S8700 Media Server.

The Avaya S8300 Media Server or S8700 Media Server with the Avaya G700 Media Gateway provides Avaya Call Center “Basic” software (included with Communication Manager) capability and optional Computer Telephony Integration (CTI) as a lower-cost call center solution for small or branch offices. For the latest capacities of supported number of agents and media gateways, please see the capacities document available at http://www.avaya.com/support. See Capacities on page 32 for instructions how to locate the capacities document.

The Avaya G700 Media Gateway with the Avaya S8300 Media Server supports more robust call center capabilities including Avaya Call Center “Deluxe,” which supports Avaya Best Service Routing and optional Avaya Virtual Routing, and Avaya Call Center “Elite,” which features Avaya Expert Agent Selection and services as the foundational software for the optional Avaya Business Advocate and Avaya Dynamic Advocate software.

The call center capabilities found in either optional software package (Deluxe or Elite) allow Communication Manager Call Center customers to enhance their customer service, help desk, travel, and other operations by providing powerful, integrated call routing via “call vectoring” and resources selection.

Avaya Call Management System

The Avaya Call Management System (CMS) collects call traffic data, formats management reports, and provides an administration interface for Automatic Call Distribution (ACD). It helps you manage the people, traffic load, and equipment in an ACD environment by answering such questions as:

- How many calls are we handling?
- How many callers abandon their calls before talking with an agent?
- Are all agents handling a fair share of the calling load?
- Are our lines busy often enough to warrant adding additional ones?
- How has traffic changed in a given ACD hunt group over the past year?

Avaya virtual routing

Avaya virtual routing (formerly known as Look-Ahead Interflow or LAI) balances the load of ACD calls across multiple locations. Virtual routing helps customers balance call loads among their locations by analyzing demand and directing each call to the location best able to handle it - for example, based on call volume, waiting time in queue, or the time of day.

With Avaya virtual routing, you can optionally route a call to a backup location based on your system ability to handle the call within parameters defined in a vector. In turn, the backup system can accept or deny the call also based on defined parameters.

Avaya virtual routing allows interflowing of only the call(s) at or near the head of the queue to provide First In/First Out (FIFO) call distribution and significantly reduce call and trunk processing for Avaya virtual routing.

Enhanced information forwarding

Enhanced information forwarding allows call center related information to be passed transparently over some public networks and non-QSIG or QSIG private networks using codeset 0 shared user-to-user information (UUI) (for non-QSIG) or QSIG manufacturer-specific information (MSI). For more information on UUI, see User-to-user information over the public network on page 74.

Call center release control

Call center release control determines which features are “active” on your switch. The call center release control feature controls whether certain call center software features are available to you.

Call prompting

Call prompting allows the system to collect information from the calling party and direct the calls using call vectoring.

The caller is verbally prompted by the system and enters information in response to the prompts. This information is then used to redirect the call or handle the call in some other way (taking a message, for example). This feature is mostly used to enhance the efficient handling of calls in the automatic call distribution application.

Data collection

Data collection allows the calling party to enter data that can then be used by a host computer application to assist in call handling. For example, this data may be the calling party account number, which could then be used to support an inquiry/response application.

Data In/Voice Answer

Data In/Voice Answer (DIVA) allows the calling party to hear selected announcements based on the digits that he or she enters. This may be used for applications such as an audio bulletin board.

Call vectoring

Call vectoring is a versatile method of routing incoming calls that can be combined with automatic call distribution for maximum benefit and call center efficiency. A call vector is a series of call processing steps (such as providing ringing tones, busy tones, music, announcements, and queuing the call to an ACD hunt group) that define how calls are handled and routed. The steps, called vector commands, determine the type of processing that specific calls will receive.

Vector commands may direct calls to on-premises or off-premises destinations, to any skill or hunt group, or to a specific call treatment such as an announcement, forced disconnect, forced busy, or music.

With combinations of different vector commands, incoming callers can be treated differently depending on the time or day of the call, the expected wait time (EWT), the importance of the call, or other criteria. Each vector can have up to 32 commands. Communication Manager also allows vectors to be linked through the “goto vector” command.

Advanced vector routing

Advanced vector routing is a collection of features that enhance Communication Manager vector routing capabilities.

Average Speed of Answer routing

Average Speed of Answer (ASA) routing is an enhancement to call vectoring that provides a flexible method for routing calls or queuing calls based on their average speed of answer for a VDN or a split/skill.

Best service routing

Best service routing (BSR) distributes the call to the best local or remote split/skill among the resources to be considered, based on expected wait time (EWT) or available agent characteristics.

Best service routing polling over IP without B-channel

Best service routing (BSR) polling over IP without B-channel provides the ability to do BSR polling between multiple sites over H.323 IP trunks without requiring an ISDN PRI B-channel. This also eliminates the associated IP media processor hardware.

QSIG temporary signaling connections are used by the BSR polling software to eliminate the need for the IP media processor board, thereby making BSR an even more cost effective multi-site solution.

Expected Wait Time routing

The Expected Wait Time (EWT) feature makes call center routing decisions based on waiting time for calls in queue, using a patented algorithm that continuously estimates call waiting times. Announcements of the wait time customers can expect before their call is answered can make time in queue more tolerable.

Call center messaging

Call center messaging gives the calling party the option of leaving a message or waiting in queue for an agent. This may be used for an online order entry system or to further automate an incoming call center operation.

Holiday vectoring

With holiday vectoring, a flexible approach for managing incoming calls on special dates is available. Holiday vectoring allows for branching and routing of calls based on information about special schedules. The special schedules are recorded in tables, each of which can hold up to 15 special dates or ranges of dates. Holiday vectoring makes it possible for up to 10 tables to be treated differently in vector processing.

Vector Directory Number

Calls access Communication Manager vectors using Vector Directory Numbers (VDN). A VDN is a “soft” extension number that is not assigned to a physical equipment location. A VDN has several properties that are administered by the system manager.

A VDN can be accessed in almost any way that an extension can be accessed. When answering a call, the answering agent sees the information (such as the name) associated with the VDN on their display, and can respond to the call with knowledge of the dialed number. This operation provides dialed number identification service (DNIS), allowing the agent to identify the purpose of the incoming call.

Class of Restriction for VDN

Class of Restriction (COR) is checked for transfer to the VDN. It can also be used to block the AUX trunk announcement from some agents. Observing can also be set to allow or restrict to that VDN.

Display VDN for route-to DAC

Display VDN for route-to DAC provides a VDN option to have the display to the answering agent show the “caller to VDN” format. The option for the “caller to VDN” display is required for ACD applications where a call needs to be routed to a specific agent, and have the call go to coverage if the agent doesn't answer or is logged out.

VDN in a coverage path

VDN in a coverage path enhances call coverage and call vectoring to allow you to assign vector directory numbers as the last point in coverage paths. Calls that go to coverage can be processed by vectoring/prompting to extend call coverage treatments.

VDN of origin announcement

VDN of origin announcement provides agents with a short message about the city of origin or requested service or the caller, based on the VDN used to process the call. VOA messages help agents respond appropriately to callers.

For example, if you have two 800 numbers, one for placing orders and one for technical support, you can administer two VDNs to route calls to the same set of agents. When an incoming call is routed to a VDN with a VOA assigned (for example, “new order” or “tech help”), the VDN routes the call to a vector that can place the call in an agent queue. When an agent answers the call, he or she hears the VOA message and can respond appropriately to the caller request.

This feature is particularly useful for visually impaired agents or agents that do not have display telephones.

VDN return destination

VDN return destination is an optional feature that re-routes a call that has been processed through a vector, to the administered return destination. This step occurs once all parties, except the originator, have dropped. The return destination must be a VDN extension.

Call Work Codes

Call Work Codes (CWC) allows ACD agents to enter digits for an ACD call to record the occurrence of a customer-defined event, such as a social security numbers or telephone numbers. The agent enters the call work code by operating the CWC feature button and using the dial pad during an ACD (inbound) call without interrupting the conversation, or in the After Call Work (ACW) mode following the call. The digits are displayed on a display-equipped telephone while being entered.

Caller Information Forwarding

The Avaya call center also supports AT&T Caller Information Forwarding (CINFO) service, allowing customers to collect customer-provided data forwarded through the network. This information can be used to route calls or provide visual displays on agent voice terminals, or be passed along to Computer Telephony Integration (CTI) applications.

Circular station hunt group

This hunt group type is an alternative to the “ddc” or “hot-seat” algorithm in a hunt group. Communication Manager keeps track of the last extension in the hunt group that received a call. When another incoming call arrives, it is sent to the next idle extension, bypassing the extension that had received the previous call.

The first extension in the hunt group will no longer be the busiest telephone while the others in the group are sitting idle.

Clear the display of collected digits

You can define when the system clears the display of collected digits (**Callr-info**) from the agent telephone. The system allows the:

- existing default option to clear the display when the next call is received
- option to clear the display when the call is released
- option to keep the displayed digits while the agent is in After Call Work (ACW) mode.

CMS measurement of ATM

The Call Management System (CMS) measurement of ATM feature provides the capability to externally measure ATM trunks on CMS. The CMS messages and reports are modified to support the expanded equipment location.

Dialed Number Identification Service

This feature displays, for a called party or answering position, the service or product associated with an incoming call. You administer what the system displays.

Direct agent calling

Direct agent calling lets the customer's callers automatically go directly to the same agent whenever they call for prompt, personalized service. These direct-to-the-agent calls are also included in their call center measurement statistics.

Dual links to CMS

The dual links to CMS feature provides an additional TCP/IP link to a separate CMS for full, duplicated CMS data collection functionality and high availability CMS configuration. The same data is sent to both servers, and the administration can be done from either server.

The ACD data is delivered over different network routes to prevent any data loss from such conditions as:

- ACD link failures
- CMS hardware or software failures
- CMS maintenance
- CMS upgrades

Duplicate agent login ID administration

Duplicate agent login ID administration simplifies administration of similar agent login ID forms.

Agent-loginID skill pair increase

Since the LINUX platform supports 20,000 administered agent-loginIDs, the administered agent-loginID skill pairs has been increased from 65,000 to 180,000.

With this enhancement, customers could administer an average of 9 skills per agent for the 20,000 agent-loginIDs (180,000/20,000). Customers could also administer 9,000 agents with 20 skills each (180,000/20). The number of skill pairs is administered on the **Display Capacity SAT** screen using the `Administered Logical Agent-Skill Pairs` field.

Note:

This capacity increase applies only to the S8700 Media Server and other configurations that have the S8700 capacities.

Expert Agent Selection

Expert Agent Selection (EAS) enables certain skill types to be assigned to a call type or a Vector Directory Number (VDN). Routing calls through vectoring then allows the system administration to direct calls to agents who have the particular agent skills required to complete the customer inquiries.

Add/remove skills

Allows an agent using expert agent selection (EAS) to add or remove skills. A skill is a numeric identifier that refers to the specific ability of an agent. For example, an agent who speaks English and Spanish could be assigned a language-speaking skill with an identifier of 20. The agent then adds skill 20 to his or her set of working skills. If a customer needs a Spanish-speaking agent, the system routes the call to an agent with that skill. Each agent can have up to four active skills, and each skill is assigned a priority level.

Call distribution based on skill

Calls that require certain agent skills (such as “knowledgeable about product X” or “speaks Spanish”) can be matched to an agent who matches the required skill. You can assign one of up to 999 skill numbers to each need or group of needs. The skills are administered and associated for each of the following:

- Vector directory numbers (VDN)
- Agent login IDs
- Callers

This refined skill definition capability allows you to organize call handling based on customer, product, and language, for example.

Queue to best ISDN support

Queue to best information is passed transparently over several public networks and QSIG private networks using the envelopes that are part of the QSIG Manufacturer-Specific Information (MSI) and the ISDN platform enhancement.

Least Occupied Agent

The Least Occupied Agent (LOA) feature distributes calls evenly across all available agents, balancing the workload among agents with fewer skills and agents with several skills. LOA solves the problem of agents who are bombarded with calls after logging into a skill at the start of a shift, while the agents who are already logged in have maintained their current incoming call level.

Multiple call handling (forced)

This feature allows agents to receive an ACD call while other types of calls are alerting, active, or on hold.

Multiple music/audio sources

Multiple music/audio sources lets customers deliver music or customized announcements to callers while they are in queue, helping to make the waiting time more productive or entertaining. Customers can provide information about their products, services, other call center applications, offer public service information, or play music.

Locally sourced announcements and music

Use the Locally Sourced Announcements and Music feature to access announcement and music audio sources on a local port network or media gateway.

Locally sourced audio can:

- improve the quality of audio
- reduce resource usage, such as VoIP resources
- provide a backup mechanism for announcement and music sources

Multiple split queuing

Multiple split queuing lets customers direct a call to several splits at the same time, so that the first available agent can take the call. It can help customers handle the busiest periods with greater ease and provide faster service to their callers.

Network Call Redirection

Today, call center customers are looking for many ways to reduce their costs. One of these ways is to employ Public Switched Telephone Network (PSTN) virtual private networks (VPNs) to eliminate as much private network cost as possible. These cost reductions are particularly valuable in enterprises or multi-site call-center environments and especially to enterprise call centers where network costs are typically high.

Network call redirection (NCR) offers a call redirection method between sites on a public network or a PSTN VPN, to help reduce trunking costs. NCR may only be activated for incoming ISDN trunk calls where the associated trunk group has been enabled by the public network service provider to use network call transfer or network call deflection features.

ETSI Explicit Call Transfer signaling

The Network Call Redirection (NCR) support of the “ETSI Explicit Call Transfer” feature is desired by multi-site, non-U.S. Avaya call center customers who use various PSTN service providers for ISDN services. These non-U.S. call centers wish to accomplish call transfers between sites without holding the ISDN trunks of a transferred call at the call redirecting Communication Manager site.

The Network Call Redirection/Network Call Deflection (NCR/NRD) feature does not allow for announcement and call-prompting call-vectoring operations. Therefore, the ETSI ECT feature is for these call center customers who cannot use NCR/NRD since they wish to play an announcement to a caller and use Communication Manager call-prompting to allow the caller to determine the routing for the call.

Network call redirection 2B-channel transfer

This enhancement adds support for the 2B-Channel Transfer PSTN network transfer protocols to the Network Call Redirection (NCR) feature. The protocols that are supported are:

- Telcordia TBCT (offered by local and inter-exchange PSTNs with Lucent 5Ess or Nortel DMS100 switches in US or Canada)
- 1998 ANSI Explicit Call Transfer (ECT) for future use.

Another form of network transfer is where the PBX sets up the second leg call and asks the network to merge the incoming call with the outgoing call (the 2B-channels) and drops the trunks to the PBX.

PC Application Software Translation Exchange

PC Application Software Translation Exchange (PASTE) allows users to view call center data on display telephones, displaying what each terminal button is, and what the feature access codes for the switch are. PASTE is used in conjunction with Avaya IP agent.

Priority queuing

Priority queuing allows special callers to be assigned priority status and routed ahead of other callers. Clients can pamper their best customers with the fastest attention possible.

Reason codes

Allows agents to enter a numeric code that describes their reason for entering auxiliary (AUX) work mode or for logging out of the system. Reason codes give call center managers detailed information about how agents spend their time. You can use this data to develop more precise staffing forecasting models or use it with schedule-adherence packages to ensure that agents are performing scheduled activities at the scheduled time. You must have expert agent selection (EAS) enabled to use reason codes.

Redirection on no answer

This feature redirects a ringing ACD split or skill call or direct agent call after an administered number of rings. This prevents an unanswered call from ringing indefinitely. The call can redirect either to the split or skill to be answered by another agent or to a Vector Directory Number (VDN) for alternative call handling. Direct agent calls route to the agent coverage path, or to a VDN if no coverage path is administered. You must have ACD enabled to use this feature.

Remote logout of agent

The remote logout of agent feature allows a select set of users to log out an agent using a feature access code.

Service observing

Service observing allows a specified user, such as a supervisor, to observe or monitor calls of another user. A vector directory number call can also be observed. Observers can observe in listen-only or listen-and-talk mode. You set up service observing to observe a particular extension, not all calls to all extensions at a terminal.

Note:

Service observing may be subject to federal, state, or local laws, rules, or regulations or require the consent of one or both of the call parties. Familiarize yourself and comply with all applicable laws, rules, and regulations before using this feature.

Listen-only FAC for service observing

The system provides a no-talk, listen-only service observing feature access code (FAC). This FAC does not reserve a second timeslot for potential toggle to talk and listen mode. This feature is for call recording applications that use Service Observing of stations/ACD agents to provide increased call recording capacity by reducing the timeslot usage.

Note:

This capability only applies to Port Network Gateways such as the G650. The additional timeslot is still reserved in H.248 Media Gateways.

Service observing by COR

Service observing by class of restriction (COR) restricts certain users from using the service observing feature.

Service observing of VDNs

Service observing of VDNs (also known as VDN observing on agent answer) allows a supervisor to start observing a call to the VDN when the call is delivered to the agent station. The observer will not hear the call during vector processing (announcements, music, and so on).

Service observing remote

This option will allow observing from non-feature button equipped stations. An observer will be able to monitor a VDN or a physical extension remotely using an “observe FAC” procedure through the remote access feature and/or call vectoring/call prompting features (through VDNs).

Vector-initiated service observing

Vector-initiated service observing, also called VDN observing on agent answer, allows users to start observing of a call to the VDN when the call is delivered to the agent or station. This saves time for the observer after observing of the VDN has been activated since the observer does not have to wait listening for each subsequent call to go through vector processing and for the agent to answer.

Site statistics for remote port networks

The site statistics for remote port networks feature forwards location IDs to CMS to provide call center site-specific reports.

User-to-user information over the public network

This feature provides the mechanism to pass information across several key public networks, including information that is originated or destined for one of several applications on Communication Manager.

Voice Response Integration

Voice Response Integration (VRI) integrates call vectoring with the capabilities of voice response units such as the Avaya CONVERSANT voice information system. You can also integrate a voice response unit with ACD. All this provides a variety of advantages. For example, while a call is queued, a caller can listen to product information via an audiotext application or can complete an interactive voice-response transaction. It may be possible to resolve the caller questions while the call is queued, which helps reduce queuing time for other callers during peak times.

VuStats

VuStats presents BCMS statistics on telephone displays. Agents, supervisors, call center managers, and other users can press a button and view statistics for agents, splits or skills, VDNs, and trunk groups. These statistics can help agents monitor their own performance, or respond appropriately to the caller request. Features include:

- VuStats login IDs
- VuStats service level

Chapter 5: Collaboration

Avaya Communication Manager contains a variety of features aimed at providing easy ways to collaborate with groups of peers, customers, and partners such as executives, sales people, and professional specialists. These key work groups require a high level of effective interaction, and Communication Manager delivers.

This chapter is divided into three sections:

- Conferencing
- Multimedia calling
- Paging and intercom

Conferencing

Abort conference on hang-up

When you press the conference button and for any reason you hang up before you complete the conference, you will cancel the conference. The original call that was put on soft-hold is put on hard-hold.

Conference - three party

The conference button allows single-line telephone users to make up to three-party conference calls without attendant assistance.

Conference - six party

The conference button allows multi-appearance telephone users to make up to six-party conference calls without attendant assistance.

Conference/transfer display prompts

Conference/transfer display prompts are based on the user class of restriction (COR). The display prompts are based on the user COR, independent of the select line appearance conferencing and no dial tone conferencing feature. The display messages vary depending on the activation of the two features, but the choice of displaying the additional information or not is dependent on the station user COR.

Conference/transfer toggle/swap

The conference/transfer toggle/swap feature allows users to toggle between two parties in the middle of setting up a conference call prior to connecting all parties together, or to consult with both parties prior to transferring a call. The display also toggles between the two parties.

Group listen

The group listen feature simultaneously activates your speakerphone in listen-only mode, and your handset or headset in listen-and-speak mode. This allows you to serve as spokesperson for a group. You can participate in a conversation while everyone else in the room is listening to what is said.

Note:

This feature works only on certain types of telephones. It is not supported on IP telephones.

Hold/unhold conference

Allows user to use the **Hold** button to bring the held party back to the conversation. This is an alternative to using the line appearance button. Hold/unhold only applies if there is only one line on hold and no other line appearances are active. An error message is displayed if the unhold feature is attempted when not allowed.

Note:

This feature is not available for BRI stations or attendant consoles.

Meet-me Conferencing

The Meet-me Conferencing feature allows a person to set up a dial-in conference of up to six parties. The Meet-me Conferencing feature uses call vectoring to process the setup of the conference call.

Meet-me Conferencing can be optionally set up to require an access code. If an access code is assigned, and if the vector is programmed to expect an access code, each user dialing in to the conference call must enter the correct access code to be added to the call.

The Meet-me Conferencing extension can be dialed by any internal or remote access users, and by external parties if the extension number is part of the customer DID block.

Expanded Meet-me Conferencing

Use the Expanded Meet-me Conferencing application to set up multi-party conferences consisting of more than six parties. The Expanded Meet-me Conferencing application supports up to 300 parties. This application is available with Communication Manager release 3.0 or later.

The Expanded Meet-me Conferencing application requires an external Meeting Exchange (MX) server. For more information, see the Expanded Meet-me Conferencing section of the *Feature Description and Implementation for Avaya Communication Manager*, 555-245-205, Issue 3 or later. Also see the MX server documentation.

No dial tone conferencing

This feature can eliminate user confusion over receiving dial tone when trying to conference two existing calls. It skips the automatic line selection if there is already a party on hold or an alerting line appearance. Help messages help guide the user. This feature is assigned on a system wide basis.

No hold conference

This feature allows a user to automatically add another party to a conference call while continuing the conversation of the existing call. The new party is automatically entered into the conversation as soon as the call is answered. An optional tone can be provided prior to the party being added to the call.

Note:

The calling station cannot hold, conference, or transfer an Emergency Access to Attendant call. This applies to both the traditional means of using these features, and to the no-hold method of using these features.

After dialing is complete, if the No Hold Conference is not answered within the time specified in an administered “timeout” field, the No Hold Conference call is deactivated.

Select line appearance conferencing

If you are in a conversation on line “b”, and another line is on hold or an incoming call is alerting on line “a”, then pressing the **CONF** button bridges the calls together. Using the select line appearance feature on Communication Manager, the user has the option of pressing a line appearance button to complete a conference instead of pressing **CONF** a second time.

This feature only applies if the line is placed in soft hold by pressing the **CONF** button. This feature never applies if the soft hold was due to pressing a **TRANSFER** button.

Selective conference party display, drop, and mute

The selective conference party display, drop, and mute feature allows any user on a digital station with display or on an attendant console to use the display to identify all of the other parties on a two-party or conference call.

The user would press a feature button while on the call that puts the station or console into conference display mode. The user then can scroll through the display of each party currently on the call by repeatedly pressing the feature button. The display would show the number and name (when available) of the caller.

The user could then do either of the following:

- The user can selectively drop the party currently shown on the display with a single button push. This can be useful during conference calls when adding a party that does not answer and the call goes to voice mail.
- The user can selectively mute the party currently shown on the display with a single button push. This puts the selected party in “listen-only” mode. This can be useful during conference calls when a party puts the conference call on hold and everyone on the call is forced to listen to music-on-hold. The user can mute that party so the conference call can continue without interruption. The muted party can then rejoin the call by pressing the # key on their telephone.

CAUTION:

Station users must be careful when scrolling through the displays when using the selective conference party display feature. The station hyperactivity feature will take the station out of service if the user repeatedly scrolls through the displays at high enough rates. This causes the station to be reset and the user is dropped from the call.

The Expanded Meet-me Conferencing application impacts selective display for all types of conferences. For more information, see the Expanded Meet-me Conferencing section of the *Feature Description and Implementation for Avaya Communication Manager*, 555-245-205, Issue 3 or later.

Selective conference mute

Selective conference mute allows a conference call participant, who has a display station, to mute a noisy trunk line. Selective conference mute is also known as far end mute.

Examples of noisy trunk lines that might need to be muted during a conference call are:

- cell telephones
- telephones that utilize the Music-On-Hold feature
- telephones with no mute capabilities

Selective conference mute only applies to trunk lines on the conference call, and not to stations. Only one trunk line on the conference call can be selectively muted at a time. This enhanced conferencing feature can be activated from any display station with a “conf-dsp” button and an “fe-mute” button.

The selective conference mute feature works with any conference established through Communication Manager, either a traditional 3 or 6 party conference or a Meet-Me conference.

Note:

This feature requires that the enhanced conferencing feature be set to **Y** on the “system-parameters customer-options” screen.

Multimedia calling

Multimedia calls are initiated with voice and video only. Once a call is established, one of the parties may initiate an associated data conference to include all of the parties on the call who are capable of supporting data. The data conference is controlled by an adjunct device called an Expansion Services Module (ESM).

Multimedia Application Server Interface

The multimedia Application Server Interface (ASA) provides a link between Communication Manager and one or more multimedia communications eXchange nodes. A Multimedia Communications Exchange (MMCX) is a stand-alone multimedia call processor produced by Avaya. This link to Communication Manager enhances the capabilities of each multimedia communications eXchange system by enabling it to share some of the Communication Manager features.

In particular, the interface provides the following advantages:

- Call Detail Recording (CDR) - This allows you to capture call detail records so you can analyze the call patterns and usage of multimedia calls just as Communication Manager administrators analyze normal calls.
- Automatic Alternate Routing/Automatic Route Selection (AAR/ARS) - This allows for the intelligent selection of the most cost-effective routing for calls, based on available resources and your carrier preference. The system may select public trunks through a DEFINITY® MultiMedia Communication Exchange (MMCX).
- Voice mail integration - You can access your embedded AUDIX or INTUITY AUDIX voice messaging system from a MultiMedia Communication Exchange (MMCX).

Multimedia call early answer on vectors and stations

Early answer is a feature applied to multimedia calls in conjunction with conversion to voice. The early answer feature:

- Answers the data call
- Establishes the multimedia protocol prior to completion of a converted call
- Ensures that a voice path to/from the originator is available when the voice call is answered

For an incoming call, early answer answers the dynamic service-link calls when the destination endpoint answers, unless early answer is specified during routing or termination processing.

Note:

The “destination voice endpoint” might be an outgoing voice trunk if the destination voice station is forwarded or covered off-premises.

Multimedia Call Handling

See Multimedia Call Handling on page 176.

Multimedia call redirection to multimedia endpoint

A dual port multimedia station may be a destination of call redirection features such as call coverage, forwarding, and station hunting. The station can receive and accept full multimedia calls or data calls converted to multimedia.

Multimedia data conferencing (T.120) through an ESM

The data conference is controlled by an adjunct device called an Expansion Services Module (ESM). The ESM is used to terminate T.120 protocols [including Generalized Conference Call (GCC), a protocol standard for data conference control] and provide data conference control and data distribution. The MultiMedia Interface circuit pack, TN787, is used to rate adapt T.120 data to/from the ESM.

For more information on ESM, see *Installation for Adjuncts and Peripherals for Avaya Communication Manager*, 555-233-116.

Multimedia hold, conference, transfer, and drop

Station users have the ability to activate hold, conference, transfer, or drop on multimedia calls. Multimedia endpoints and voice-only stations may participate in the same conference.

Multimedia queuing with voice announcement

When multimedia callers queue for an available member of a hunt group, they are able to hear an audio announcement.

Paging and intercom

Code calling access

This feature allows attendants, users, and tie trunk users to page with coded chime signals. This feature is helpful for users who are often away from their telephones or at a location where a ringing telephone might be disturbing.

Group paging

Group paging allows a user to make an announcement to a group of people using speakerphones. The speakerphones are automatically turned on when the user begins the announcement. The recipients can listen to the message over the handset if they wish, but they cannot speak to the user in return.

A group page member will not receive the page if the member is active on a call appearance, has a call ringing, is off-hook, has “send-all calls” active, or has “do not disturb” active.

Intercom - automatic

With this feature, users who frequently call each other can do so by pressing one button instead of dialing an extension number. Calling users press the automatic intercom button and lift the handset. The called user receives a unique intercom ring and the intercom lamp, if provided, flashes.

Intercom - automatic answer

Automatic answer intercom (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 telephones with built-in speaker, headphones, or adjunct speakerphone.

Intercom - dial

This feature allows multi-appearance telephone users to easily call others within an administered group. The calling user lifts the handset, presses the dial intercom button, and dials the one-digit or two-digit code assigned to the desired party. The telephone of the called user rings, and the intercom lamp, if provided, flashes. With this feature, a group of users who frequently call each other can do so by pressing one button and dialing a one-digit or two-digit code instead of dialing an extension number.

Loudspeaker paging access

Loudspeaker paging access provides attendants and telephone users dial access to voice paging equipment. As many as nine paging zones can be provided by the system, and one zone can be provided that activates all zones at the same time.

Note:

A zone is the location of the loudspeakers - for example, conference rooms, warehouses, or storerooms.

A user can activate this feature by dialing the trunk access code of the desired paging zone, or the access codes can be entered into abbreviated dialing lists. Once you have activated this feature, you can simply speak into the handset to make the announcement.

Deluxe loudspeaker paging access (called deluxe paging) provides attendants and telephone users with integrated access to voice-paging equipment and call park capabilities. When you activate deluxe paging, the call is automatically parked. The parked call returns to the parking user with distinctive alerting when the time-out interval expires.

Manual signaling

Allows one user to signal another user. The receiving user hears a two-second ring. The signal is sent each time the button is pressed by the signaling user. The meaning of the signal is prearranged between the sender and the receiver. Manual signaling is denied if the receiving telephone is already ringing from an incoming call.

Whisper page

Whisper page allows an assistant or colleague to bridge onto your telephone conversation and give you a message without being heard by the other party or parties you are talking to. Whisper page works only on certain types of telephones.

Chapter 6: Communication device support

Avaya IP Agent

Avaya IP Agent is a PC-based IP application that allows agents to use their PCs as telephones. In addition to the traditional functionality of a standard telephone (transfer, hold, conference, and so forth), IP agent offers directory services, screen pops, call history, and agent mode history.

Avaya IP Softphone

Avaya IP Softphone extends the level of Communication Manager services. This feature turns a PC or a laptop into an advanced telephone. Users can place calls, take calls, and handle multiple calls on their PCs.

Note:

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 configuration and the Native H.323 configuration for the IP Softphone.

The R5 release of the IP Softphone supports a number of enhanced features, including the following:

- Improved endpoint connection recovery algorithm
- AES media encryption (see Encryption algorithm for bearer channels on page 193)
- Instant Messaging
- Unicode support (see Unicode support on page 88)
- Softphone and Telephone Shared Control (see IP Softphone and IP Agent - Shared Control mode on page 86)

The IP Softphone provides a graphical user interface with enhanced capabilities when used with certain models of DCP telephones. Communication Manager supports a mode of H.323 registration that allows an IP Softphone to register for the same extension as a DCP telephone without disabling the telephone. It also allows the IP Softphone to send button-push messages and receive display and call progress messages in parallel with the telephone. In this mode, the Softphone does not terminate any audio.

IP Softphone and IP Agent - RoadWarrior mode

IP Softphone and IP Agent, RoadWarrior mode, enables use of the full Avaya Communication Manager feature set from temporary remote locations anywhere in the world. The RoadWarrior application consists of two software applications running on a PC that is connected to Communication Manager over an IP network.

The single network connection between the PC and Communication Manager carries two channels, one for the signaling path and one for the voice path. On Communication Manager, the RoadWarrior application requires the CLAN circuit pack for signaling and the IP media processor for voice processing.

IP Softphone and IP Agent - Shared Control mode

IP Softphone and IP Agent, Shared Control mode, enables users to have a telephone endpoint and an IP Softphone in service simultaneously on the same extension number. IP Softphone and an IP telephone can be integrated so that the IP softphone can control a desk IP telephone. This allows the power of the PC desktop (LDAP directories, TAPI PIMs/Contact Managers, etc.) to be used in conjunction with a desktop IP telephone.

An IP softphone can register to an extension number that is already assigned to an in-service telephone endpoint. From that point on, user actions carried out by either endpoint apply to calls to or from the extension. Only the telephone endpoint carries audio for the extension, however.

IP Softphone and IP Agent - Telecommuter mode

IP Softphone and IP Agent, Telecommuter mode, enables telecommuters to use the full Communication Manager feature set from home. It consists of a PC and a telephone with separate connections to Communication Manager. The PC provides the signaling path and the user interface for call control. A standard telephone provides a high-quality voice path. The Telecommuter application requires the CLAN circuit pack for signaling. The Telecommuter application does not use the IP media processor.

Avaya IP Softphone for pocket PC

Avaya IP Softphone for pocket PC extends the level of Communication Manager services. This feature turns a hand-held personal digital assistant (PDA) into an advanced telephone. Users can place calls, take calls, and handle multiple calls on their PDAs.

Avaya Communication Manager PC console

The Communication Manager PC console allows your attendants to efficiently handle incoming calls by personal computer. Using the familiar Microsoft Windows graphical user interface (GUI), the attendants can easily keep track of how long callers have been on hold and who they are waiting for. Attendants can monitor up to six calls at once.

Attendants do not need to use pen and paper when handling calls because they can make notes on their computers about what each caller needs. All this contributes to make a favorable first impression with your customers. Having the call processing software on the same computer with spreadsheet, word processing, or other software allows the attendants to stay productive between calls.

The PC console is easily customized, so even if attendants from different shifts share the same computer, they can each preserve their preferences in the call processing environment. The PC console is available in English, Parisian French, Latin American Spanish, German, Dutch, Italian, and Portuguese. If a Spanish-speaking attendant takes over for a French-speaking attendant, for example, a single press of a button converts all labels, error messages, and online help to Spanish.

Avaya SoftConsole

The Avaya SoftConsole is a Windows-based GUI application that can replace the physical 302B “hard” console. It allows attendants to perform call answering and routing through a PC interface through an IP connection.

Increased text field length for feature buttons - DCP

The increased text field length for feature buttons - DCP feature is designed to allow the switch to better support the longer text label display capabilities that are available on the newer DCP (24xx) model telephones.

This feature allows the end user to program and store 13-character labels for all feature buttons and call appearances associated with the DCP telephones on the switch. The only exceptions are those labels that are administratively blocked or disabled.

Unicode support

Communication Manager supports the display of non-English static and dynamic display text on Unicode-enabled telephones. Non-English display information is entered into a Avaya Integrated Management application. Communication Manager processes, stores, and transmits the non-English text to telephones that support Unicode displays.

Unicode support provides the capability of supporting international and multi-national communications solutions. End-users are provided with a communications interface (delivered by an IP telephone or IP Softphone) in their own native language. This feature supports the Simplified Chinese, Japanese, and Korean (CJK) character sets.

QSIG support for Unicode

The QSIG support for Unicode feature extends the Unicode support on a single server to multi-node Communication Manager networks. This feature allows Unicode support across large campus configurations. Many configurations contain multiple Communication Manager servers due to scalability requirements. This feature also allows Unicode support across large corporate networks, frequently multinational corporations, where multiple Communication Manager servers are almost always provisioned.

Chapter 7: Hospitality

Alphanumeric dialing

Alphanumeric dialing allows you to place data calls by entering an alphanumeric name rather than a long string of numbers.

Attendant room status

See Attendant room status on page 42.

Automatic selection of Direct Inward Dialing numbers

This feature allows the system to automatically choose a number from a list of available Direct Inward Dialing (DID) numbers that will be assigned to a guest room extension when checking in.

With this feature, hotels can give a guest a second telephone number that is different from their room number, thereby protecting the privacy of the guest. When a particular DID number is called, the call routes to the guest room extension, and covers as if the room was called directly. Besides improving guest security, this eliminates the need for an attendant or front desk staff to extend a call to a guest room.

Automatic wakeup

The automatic wakeup feature allows attendants, front desk users, and guests to request that one or two wake-up calls be automatically placed to a certain extension number at a later time. When a wakeup call is placed and answered, the system can provide a recorded announcement (which can be a speech synthesis announcement), music, or simply silence. With the integrated announcement feature, multiple announcements enable international guests to use wakeup announcements in a variety of languages. See Daily wakeup on page 90, Dual wakeup on page 91, and VIP wakeup on page 93.

Check-in/check-out

This feature allows front desk personnel to check guests into a hotel and, when the guests leave, check them out. There are two ways this is done: through the PMS terminal or through the attendant console (or backup telephone). Check-in and check-out from the attendant console should be used only if there is no Property Management System (PMS), or if the link to the PMS is down. If the PMS is installed and working, check guests in and out using the PMS.

For guest check-in or check-out from the console, there are two buttons on the attendant console (or backup telephone): one labeled “Check in” and the other labeled “Check out.” The check-in procedure performs two functions: it deactivates the restriction on the telephone in the room allowing outward calls, and it changes the status of the room to occupied.

Custom selection of VIP DID numbers

This feature builds on the automatic selection of DID numbers feature. It allows hotel personnel to control what DID number is assigned to a hotel room at check-in. That is, the system asks the user to specify the desired DID number when a guest is checked in. The number comes from a pool of DID numbers that are separate from those used by the automatic selection feature. The system never automatically assigns numbers from this pool. Numbers from this pool are used only when explicitly specified by the user.

Daily wakeup

Daily wakeup allows a guest or front desk personnel to schedule a single wakeup request for a daily wakeup call. For example, if a guest needs to receive a wakeup call at 5:30 a.m. for the duration of his or her stay, one request can be placed on the system instead of placing a separate request for each day.

Dial-by-name

The dial-by-name feature allows callers to the system to access guest rooms simply by dialing the name of the guest they are trying to contact. This feature uses recorded announcements and the call vectoring feature to set up an automatic attendant procedure.

This automatic attendant procedure gives callers the ability to enter a guest name. When a single or unique match is found, the call is redirected to the telephone of the guest.

Do not disturb

The do not disturb feature allows guests, attendants, and authorized front desk users to request that no calls, other than priority calls, be connected to a particular extension until a specified time.

Dual wakeup

This feature allows guests to have two separate wakeup calls. The dual wakeup feature is an enhancement to the standard automatic wakeup feature used in hospitality environments.

With the standard wakeup feature, guests or front desk personnel can create one wakeup call for each extension. The dual wakeup feature allows guests and front desk personnel to create either one or two wakeup calls. The dual wakeup feature for guests is valid only when the system is not equipped with a speech synthesizer circuit pack.

Housekeeping status

The housekeeping status feature records the status for up to six housekeeping codes and reports them to the property management system (PMS). These status codes are usually entered by the housekeeping staff from the guest room or from a designated telephone. They can also be updated by the front office personnel using the attendant console or a backup telephone. Six status codes can be used from guest rooms, and four status codes can be used from telephones that do not have the client room class of service (COS).

Names registration

The names registration feature automatically sends a guest name and room extension from the property management system (PMS) to the switch at check-in, and automatically removes this information at check-out. The information may be displayed on any attendant console or display-equipped telephone at various hotel locations (for example, room service or security).

Property Management System digit to insert/delete

Many customer configurations base a room extension by adding an extra leading digit on the room number. The PMS digit to insert/delete feature allows users to delete the leading digit of the extension in messages. The feature is useful for a hotel that has multiple extensions sharing an extra leading digit in front of the room number. The leading digit is automatically inserted when the message goes to the switch.

The PMS interface supports 3-digit, 4-digit, 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.

Property Management System interface

The Property Management System (PMS) allows a customer to control features used in both a hospital-type and a hotel/motel-type environment. The communications link allows the property management system to interrogate the switch, and allows information to be passed between the switch and the PMS. The switch exchanges guest status information (room number, call coverage path, and other data) with the PMS.

There are two ways that the guest data can be encoded:

- Using a combination of Binary Coded Decimal (BCD) encoding and the ASCII character set
- Using only the ASCII character set

Single-digit dialing and mixed station numbering

This feature provides hotel staff and guests easy access to internal hotel/motel services, and provides the capability to associate room numbers with guest room telephones. The feature provides the following dial plan types: single-digit dialing, prefixed extensions, and mixed numbering.

Suite check-in

Suite check-in allows more than one station to be checked in at one time. This is useful for a guest room that may have multiple extensions. This feature allows all extensions to be checked in at the same time. Suite check-in using the hunt-to feature will also check out all the extensions in the entire suite at the same time.

VIP wakeup

The VIP wakeup feature allows front desk personnel to provide personalized wakeup calls to important guests. When a wakeup call has been scheduled for an important guest, a wakeup reminder call is placed to the front desk personnel, who in turn personally calls the guest to provide the wakeup call.

Wake-up activation using confirmation tones

If a speech synthesizer circuit pack is not installed, guests can still enter their own wakeup calls (two wakeup calls if the dual wakeup feature is active). The guests do not receive voice prompts as they would using the speech synthesizer circuit pack. Instead, guests receive call progress tones (recall dial tone and confirmation tone) to set up their wakeup calls.

Xiox call accounting

The Xiox call accounting works as an adjunct with any system with hospitality features. Xiox call accounting allows hotel management to use their telephone system as a major source of revenue by generating the information they need to make important decisions about their network and usage.

Chapter 8: Localization

Administrable language displays

This feature allows messages that appear on telephone display units to be shown in the language spoken by the user. These messages are available in English (the default), French, Italian, Spanish, or one other user-defined language. The language for display messages is selected by each user. The feature requires 40-character display telephones.

Administrable loss plan

The administrable loss plan provides the ability to administer signal loss and gain for telephone calls. This capability is necessary because the amount of loss allowed on voice calls can vary by country. With the administrable loss plan feature, switch endpoints are classified into 17 endpoint types, and the loss plan can be administered for trunks, stations, and personal CO lines. Loss values are in the range of 15 dB loss to 3 dB gain. Preset defaults are available and are based on country type.

Bellcore calling name ID

This feature allows the system to accept calling name information from a Local Exchange Carrier (LEC) network that supports the Bellcore calling name specification. The system can send calling name information in the format if Bellcore calling name ID is administered. The following caller ID protocols are supported:

- Bellcore (default) - US protocol (Bellcore transmission protocol with 212 modem protocol)
- V23-Bell - Bahrain protocol (Bellcore transmission protocol with V.23 modem protocol).

Block collect call

This feature blocks collect calls on class-of-restriction basis. This feature is available for any switch that uses the Brazil country code. If enabled for a station, all trunk calls that terminate to the station will send back a double answer to the central office (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.

Busy tone disconnect

In some regions of the world, the CO sends a busy tone for the disconnect message. With busy tone disconnect, the switch disconnects analog loop-start CO trunks when a busy tone is sent from the CO.

Country-specific localization

Italy

Distributed Communications Systems protocol

Enhanced DCS adds features to the existing DCS capabilities and requires the use of Italian TGU/TGE tie trunks.

Additional features include:

- Exchanging information to provide class of restriction (COR) checking between switches in the EDCS network
- Providing call-progress information for the attendant
- Allowing attendant intrusion between a main and a satellite PBX
- Allowing a main PBX to provide DID/CO intercept treatment rather than the satellite PBX

Japan

National private networking support

Provides support for Japanese private ISDN networks. The Japanese private network ISDN protocol is different from the standard ISDN protocol. The switch supports extensions to the ISDN protocol for switches using the Japanese country code.

Katakana character set

Communication Manager supports the katakana character set (Japan). This nine-point character font was designed to display katakana characters in the user interface as well as in switch-generated messages.

Russia

Central Office support on G700 Media Gateway

Communication Manager supports central office (CO) trunks in Russia using the G700 Media Gateway.

ISDN/DATS network support

This feature supports ISDN/DATS trunk networks when the tone generated field is set to 15 (Russia) on the system-parameters country-options screen. It modifies the overlap sending delay and ISDN T302 and T304 timers to support the Russian trunk network.

Multi-Frequency Packet signaling

Multi-Frequency Packet (MFP) address signaling is provided in Russia on outgoing CO trunks. Calling party number and dialed number information is sent on outgoing links between local and toll switches. Russian MFP is set on each trunk group on the **type** field on the trunk screen.

Note:

Russian MFP does not apply to PCOL trunks.

E&M signaling - continuous and pulsed

Continuous and pulsed E&M signaling is a modification to the E&M signaling used in the United States. Continuous E&M signaling is intended for use in Brazil, but can also be used in Hungary. Pulsed E&M signaling is intended for use in Brazil.

Multinational Locations

For customers who operate in more than one country, the Multinational Locations feature provides the ability to use a single Enterprise Communication Server (ECS) across multiple countries with:

- telephones
- port networks
- remote offices
- media gateways

The Multinational Locations feature allows the following Communication Manager features to work across international borders:

- A & Mu law companding
- Call Progress Tone Generation
- Loss Plan
- Analog line board parameters
- Call Detail Recording
- R2-MFC (multifrequency signaling) trunks

The Multinational Locations feature works across all Linux platforms supported by Communication Manager release 2.1 or higher.

The S8300, S8500, and S8700 Media Server each supports 25 location parameter sets. You can administer one parameter set for each country that you support, for a maximum of 25 countries.

Note:

Since the S8100 Media Server supports only 1 location, and since the Multinational Locations feature depends on multiple locations, the Multinational Locations feature is not supported on the S8100 platform.

Analog line board parameters per location

You can administer the following analog line board parameters for each location:

- Analog Ringing Cadence
- Analog Line Transmission
- Flashhook Interval Upper Bound
- Flashhook Interval Lower Bound
- Forward Disconnect Timer (msec)
- Analog line tests use the same parameters

Analog line circuit packs use these parameters, according to the location parameters of the circuit pack.

Companding for DCP telephones and circuit packs per location

You can administer the Companding Mode for each remote office, media gateway, and the rest of the system that is circuit switched.

- When a Digital Communications Protocol (DCP) telephone comes into service, Communication Manager downloads the correct companding mode for the location of the telephone.
- When a circuit pack comes into service, Communication Manager downloads the administered companding mode for the media server, remote office, or media gateway that is supporting that circuit pack.

Location ID in Call Detail Record records

You can administer the following CDR parameters in the custom CDR format for both the source and destination:

- location
- time zone
- country

Loss plans per location

For each location, you can administer Digital Loss & Tone Loss, DCP terminal loss parameters, and administrator-entered customizations.

When inserting loss for a multilocation intrasystem call, Communication Manager treats the call as if IP tie trunks are connecting the different parties. When an audio stream is converted from time-division multiplexing (TDM) to Internet Protocol (IP), the system adjusts the audio stream. The system adjusts the audio stream by the IP media processor board of the sending location, to an ISO standard level for voice over IP. The system then adjusts the audio stream by the media processor of the receiving location to match the TDM levels for that location.

This board level adjustment is not done for DS1 remoted expansion port networks (EPNs). Use DS1 remoted EPNs between countries only if the countries have similar voice transmit levels.

Multifrequency signaling per trunk group

Prior to Communication Manager release 2.1, you administer R2-Multifrequency Coded (MFC) signaling parameters per system. With Communication Manager release 2.1 and higher, you administer R2-MFC signaling parameters per trunk group.

R2-Multifrequency Coded signaling trunk groups use one of 8 sets of MFC signaling parameters according to the MFC signaling code administered for that trunk group.

Tone generation per location

You can administer tone generation characteristics and administrator-entered customizations per location. You can administer the server so that, when a telephone or trunk needs to play a Communication Manager ECS-generated tone, the software plays a tone into the call using the tone characteristics of the location of the listening endpoint, or of another endpoint on the call.

Public network call priority

Provides call retention, forced disconnect, intrusion, mode-of-release control, and re-ring to switches on public networks. Different countries frequently refer to these capabilities by different names.

World class tone detection

World class tone detection enables Avaya Communication Manager to identify and handle different types of call progress tones, depending on the system administration. You can use the tone detector and identification to display on data terminal dialing and to decide when to send digits on trunk calls through abbreviated dialing, ARS, AAR, and data terminal dialing.

Chapter 9: Message integration

Audible message waiting

Audible message waiting places a stutter at the beginning of the dial tone when a telephone user picks up the telephone. The stutter dial tone indicates that the user has a message waiting. This feature is particularly useful for visually impaired people who may not be able to see a message light. It is often used with telephones that have no message waiting lights.

Audible message waiting may not be available in countries that restrict the characteristics of dial tones provided to users.

Avaya Interactive Response

The Avaya Interactive Response (IR) - formerly known as INTUITY Conversant® - voice information system is an interactive voice-response system that automates telephone transactions from simple tasks, like routing to the right department, to complex tasks, such as registering college students or providing bank balances. It communicates with customers in natural-sounding, digitally recorded speech, and performs 24-hours a day without the services of an operator.

The system can handle single or multiple voice-response applications simultaneously, and can serve up to 48 callers at once. It can operate by itself to dispense information or collect data, or it can work with a host computer to access a large database such as bank account records. With its speech-recognition capability, even rotary telephone users can have access to sophisticated telephone-based services. Advanced telephone features provide intelligent call-transfer capabilities and allow you to use the system in your existing telephone environment.

Centralized voice mail through mode code integration

The centralized voice mail feature eliminates the need for a voice mail system at each of the sites in a network. It does so by allowing a network running Avaya Communication Manager to use a single INTUITY AUDIX voice messaging system as a centralized voice mail system that serves the whole network. The INTUITY AUDIX system can also serve as a centralized voice mail system within a hybrid network of Communication Manager, DEFINITY BCS, and Merlin Legend/Magix switches.

Dual DCP I-channels

This feature supports the use of dual DCP I-channels for **AUDIX** networking. In this case, networking refers to the ability to send data files between **AUDIX** systems, not to communications with the switch.

Embedded AUDIX

While many voice messaging systems require separate equipment and connections, the embedded AUDIX system easily installs directly into your cabinet to support advanced voice messaging capabilities without the need for an adjunct processor. Each embedded AUDIX system supports up to 2000 mailboxes and stores up to 100 hours of recorded messages.

Whenever you call the embedded AUDIX system, you interact with it by entering commands through your telephone touch-tone keypad. You simply specify the desired activity, and follow the voice prompts for the desired task.

Special voice-processing features include voice mail, call answering, outcalling, multi-level automated attendant, and bulletin board. The following is a summary of embedded AUDIX capabilities:

- *Shared extensions* provide personal mailboxes for each person sharing a telephone.
- *Multiple* p*ersonal greetings* allows you to prepare a pool of up to nine personal greetings to save time and provide more personal customer service. Separate messages can indicate that you are on the telephone, away from the desk, on vacation, etc. You can assign different messages to internal, external, or after-hours calls.
- *Priority messaging* places important messages ahead of others. Internal and external callers can mark the message as priority.
- *Outcalling* automatically dials a prearranged telephone number or pager when you have messages in your voice mailbox.
- *Priority outcalling* automatically dials a prearranged telephone number or pager when you have *priority* messages in your voice mailbox.
- *Broadcasting* allows you to send a single message to multiple recipients or to all users on the system.
- *System* b*roadcast* allows you to send broadcast messages as regular voice messages, or as messages that recipients hear as they log in.
- *AUDIX directory* allows you to look up the extension number of any other user by entering their name on the telephone keypad.
- *Personal directory* allows you to create a list of nicknames for quick access to telephone numbers.

- *Call answering for nonresident subscribers* provides voice mailboxes for users who do not have an extension number on the system.
- *Full mailbox answer mode* informs callers whenever messages cannot be left because there is no room in a subscriber mailbox.
- *Name record by subscriber* lets you record your own name on the system.
- *Automatic message scan* can play all new messages in part or in their entirety without requiring you to press additional buttons, which is particularly useful when you are getting messages from your mobile telephone.
- *Sending restrictions by community* enables you to limit the communities of callers who can communicate using AUDIX voice messaging.
- *Group lists* allows you to create mailing lists of up to 250 people to use for broadcasting messages.
- *Message forwarding* allows you to forward messages with or without attached comments.
- *Name addressing* allows you to address messages by name if you do not know the extension.
- *Private messaging* is a special coding feature that prevents recipients from forwarding messages.
- *Leave word calling* allows you to press a button on your telephone in order to leave a standard “call me” message on any extension.
- *Online help* provides you with instant access to voiced instructions at any time when you are using the system.
- *Multiple language support* allows you to install up to nine languages on the system, from a superset of 30 available languages.
- *Enhanced message handling* gives you the flexibility for handling messages. Two of these features are *optional advance/rewind* that lets you advance through and rewind individual messages, and *undelete messages* that lets you retrieve any messages that you may have accidentally deleted.

INTUITY AUDIX

INTUITY messaging solutions essentially offers the same user features as the embedded AUDIX system, plus the following features:

- *Fax messaging* allows you to handle faxes as easily as you handle voice mail. You can send, receive, store, scan, delete, skip, or forward faxes. This feature is fully integrated with voice messaging, so you can attach faxes to voice messages, for example. You can also create special mailboxes for each of your fax machines. These mailboxes accept fax telephone calls when the fax machine is busy and then deliver the fax to the fax machine when the fax machine is available.

- *Turn off AUDIX call answering* allows you to turn off call answering in order to conserve system resources. You can create a message that tells callers they cannot leave a message, giving them another number to call, for example.
- *Pre-addressing* allows you to address a message before recording it.
- *Integrated messaging* allows you access and manage incoming voice, fax, and e-mail messages and file attachments from your personal computer or your telephone. A voice message will thus appear in your e-mail mailbox, for example, and vice versa. You can also set options to have just the message headers appear in the alternate mailbox. You can also create a voice or fax message by telephone and send it to an e-mail recipient.
- *Text-to-speech* allows you listen to a voice rendering of text messages sent from a supported e-mail system and/or INTUITY message manager.
- *Print text* allows you to print messages sent from a supported e-mail system and/or INTUITY message manager.
- *Enhanced addressing* allows you to send a message to up to 1500 recipients.
- *Transfer restrictions* allow you to control toll fraud by restricting transfers going through the voice messaging system.
- *Internet messaging* allows you to exchange messages (voice and text) with any e-mail address via the World Wide Web.
- *Avaya voice director* allows you to address messages via spoken name, in addition to using touchtones to enter extensions or names. It also supports transferring to AUDIX subscribers, including those in other locations, by speaking a name.
- *International availability*.

Avaya IA770 INTUITY AUDIX messaging application

The IA770 application enhances communications and information exchange within enterprises, helping customers be more successful with call answering and messaging. The IA770 application enables customers to see messages on their PCs, add a voice mail component to an e-mail, and listen to e-mail using voice mail.

IA770 uses the Linux operating system, making it consistent with the operating system of the G700 and G350 Media Gateways. The distributed architecture is designed for reliability and survivability and is centrally managed for simplicity, efficiency and quick response to help ensure business recovery.

The IA770 application consists of license file-activated software residing on the S8300 Media Server, and a small card that can be installed and upgraded in the field.

The IA770 application includes INTUITY Message Manager. While the system provides text-to-speech capability in U.S. English only, there is no additional charge for initial implementation of any of the 35 available languages for prompts.

IA770 supports INTUITY digital (TCP/IP) networking protocol. More extensive networking can be provided with the Avaya Interchange.

Using the Web interface, the administrator can perform a system backup and restore of all administered data - announcements, recorded names, greetings - and approximately 50 hours of messages over the local area network (LAN). The screens are easier to understand and more intuitive, which should cut installation time and lessen the need for training and experience. The IA770 system uses smart defaults rather than requiring every field to be addressed.

For more information on the IA770 messaging application, see the *S8300 and S8700 Library CD*, 555-233-825.

Record on messaging

Users can record conversations by pressing a single button. This feature uses AUDIX as the recording device. This feature is not available with INTUITY AUDIX through Mode Codes or remote AUDIX.

Note:

It it important that anyone who wants to activate this feature should study and understand your local laws regarding the recording of calls *before* activating this feature.

A feature button named `audix-rec` is used for this feature. The button is available for all stations that have administrable feature buttons. When administered, the button also requires a hunt group extension number (for the AUDIX extension number) along with it.

Note:

Attendant consoles do not have this button.

To record a conversation when a call is in process, press the `audix-rec` button. When you push the button, the LED light for the feature button begins to flash. After about 4~6 seconds, internal users who are participating in the call will notice that the telephone display changes to `CONFERENCE`. The LED light on the telephone that initiated the recording is steadily illuminated. This indicates that the AUDIX recording facility is ready and begins to record the conversation.

The internal users on the same switch with the display equipment can notice that the number of parties in the call increases by 1. At this point, depending on the administration, a ready indication tone will play to all the parties in the call, the initiator only, or none of the parties.

After enough information has been recorded, the initiator can then stop the recording by pressing the `audix-rec` button a second time when the LED light is illuminated. The feature button LED light on the initiator telephone goes out. The internal users with the display equipment can again notice that the number of parties in the call decreases by 1. The call remains active.

The `Interval For Applying Periodic Alerting Tone` field is used to allow the switch administrator to choose an interval to play an alerting tone to all the parties on the call during recording. Values are 0 to 60, and the default is 15. This means, if the default value is used, that all parties on the call hear an alerting tone every 15 seconds that indicates the conversation is being recorded. If the value for the field is 0, then no periodic tone is played during recording.

S8100 Media Server embedded INTUITY AUDIX

This application provides voice, fax, and text messaging, along with text-to-speech and message manager functionality in a single processor mezzanine board on the S8100 Media Server.

Note:

Communication Manager, release 3.0 or later, is not available on the S8100 Media Server.

Included are Avaya Directory Enabled Management (DEM) and Fax Extended Dialing (FED).

- ADEM provides real time directory-based access to Communication Manager and INTUITY AUDIX.
- FED allows the customer to specify restrictions on the destination numbers, as well as eliminate the need to administer fax number ranges as remote AMIS networking machines. Additionally, FED addresses the entry of international destination numbers by allowing up to 23 digits for fax endpoints.

The INTUITY AUDIX mezzanine card also provides the necessary DSP resources for messaging. This hardware eliminates the need for the INTUITY Map 5P adjunct, usually required for this functionality.

Also see Embedded AUDIX on page 104.

INTUITY call accounting system

If you are using any of the INTUITY voice messaging products, the INTUITY call accounting system is probably the best call-accounting solution for you. The system works exclusively with INTUITY products, which reside on MAP/40 or MAP/100 computers. Offering many of same features as the call accounting system for Windows, the system also serves to help integrate your INTUITY applications.

INTUITY lodging

INTUITY lodging is a messaging system designed especially for lodging establishments such as hotels or other lodging providers such as hospitals or colleges. The system supplies guests with electronic mailboxes that store voice or fax messages. INTUITY lodging serves as a private answering machine for each extension.

Hotel guests can leave messages for each other without going through the attendant. For incoming calls, an attendant transfers the call to the appropriate room. If the guest does not answer the call or if the line is busy, the call is automatically transferred to the guest voice mailbox, where the caller can leave a voice message. A message-waiting indicator on the guest telephone notifies the guest that the voice mailbox contains messages. Guests are assigned a password for accessing messages remotely. They can retrieve and save messages from any telephone, on or off premises.

Guests can hear voice mail prompts and menus in one of several languages. The current set of available languages includes the following:

- American English
- Arabic (female voice)
- Brazilian Portuguese
- British English
- Canadian French
- German
- Greek
- Japanese
- Latin American Spanish
- Mandarin Chinese
- Parisian French
- Russian

Any or all of these languages may be installed, but only nine can be made available at any one time. The attendant enters the desired language of the guest at check-in time. The guests hear menus and prompts in their chosen languages after logging in to retrieve messages. For more information about language options, contact your Avaya account representative.

INTUITY lodging call accounting system

The INTUITY lodging call accounting package (an integrated offering from Homisco) takes call records supplied by the system, puts the records into a standard bill format, and sends the billing information to the property management system. When guests check out, their long distance calling charges are printed automatically on their bill. This gives you better control over telephone usage revenue.

Leave Word Calling

Leave Word Calling (LWC) allows internal system users to leave a short preprogammed message for other internal users. The preprogammed message usually is the word “call,” the caller name, extension, and the time of the call. When the message is stored, the message lamp on the called telephone automatically lights.

LWC messages can be retrieved using a telephone display, voice message retrieval, or AUDIX. Messages may be retrieved in English, French, Italian, Spanish, or a user-defined language.

Leave Word Calling - QSIG/DCS

The Leave Word Calling (LWC) feature is extended to enterprise networks with QSIG as the private network protocol, as well as those with DCS.

For enterprise networks that are mixed or in transition from DCS to QSIG, interworking of the LWC feature between the protocols can be provided. LWC also works within a single non-networked switch.

Note:

A DCS+ signaling group is needed, but can only be used in networks with 4-digit or 5-digit dial plans.

Manual message waiting

This feature allows multi-appearance telephone users to light the status lamp associated with the manual message waiting button at another multi-appearance telephone. They do this by simply pressing a button on their own telephone. This feature can be administered only to pairs of telephones such as a secretary and an executive. The secretary might press the button to signal to the executive that a call needs answering or someone has arrived for an appointment. The executive might use the button to indicate that he or she should not be disturbed.

Message demand print

Message demand print allows you to print your undelivered messages without calling the message center.

Message retrieval

With the message waiting lamp on their telephones, employees always know when they have messages. Messages can be retrieved in a variety of ways. These message retrieval options can be assigned to individual users.

Display retrieval

Users having digital telephones with displays or a personal computer integrated with a telephone can display messages.

Speak-to-me

Using any touch-tone telephone, employees can dial speak-to-me and hear a synthesized voice read their messages over the telephone.

Message Sequence Tracer enhancements

In the past, it had been difficult to trace messages through the Message Sequence Tracer (MST) tool pertaining to a particular socket because there was no tag in each message distinguishing it from other sockets.

New message formats for outgoing and incoming data now include the socket number/identifier. These new formats use new Type identifiers of 05 and 06. A pair of new formats 07 and 08 have also been created for outgoing and incoming socket control messages on the PROCR ip-interface.

By creating new format types for these new formats, the task of decoding these messages is easier.

The following enhancement was made to the Message Sequence Tracer (MST):

- Signaling messages between Communication Manager and the TN799 CLAN can now be traced for better diagnostics during network outages.
  - Add processor TN799 CLAN socket information to the MST trace in order to help developers debug socket problems.
  - Enhance MST to include the socket number in socket data.
  - Add TN799 CLAN board ID to CLAN MST IP socket trace messages.

Mode code interface

Communication Manager supports an analog mode code interface for communications with INTUITY AUDIX and other voice mail systems using the same interface. This interface employs DTMF tones, line signals, and feature access codes, and allows INTUITY AUDIX to exchange data with Communication Manager without using a data link. Other adjunct vendors can engineer their products to use this interface.

Octel integration

Communication Manager integrates with the entire line of Octel messaging systems including the Octel 200/300 message server, and the Octel 250/350 message server.

QSIG/DCS voice mail interworking

QSIG/DCS voice mail interworking is an enhancement to the QSIG feature. It integrates DCS and QSIG centralized voice mail using the DCS+/QSIG gateway. Switches labeled DCS+/QSIG integrate multi-vendor PBXs into a single voice messaging system. QSIG/DCS voice mail interworking works on G3r, G3si, and G3csi. It provides network flexibility, DCS functionality without a dedicated T1.

Multiple QSIG voice mail hunt groups

Communication Manager provides for ten message center hunt groups to support QSIG integrated messaging. This feature allows customers to spread users in a single Communication Manager system over multiple messaging systems. This allows users to move among Communication Manager systems while retaining their same voice mailbox. Users do not lose voice messages.

This feature also enhances customer usability of Avaya messaging systems in the enterprise by allowing not only for growth, but the ability to migrate end users on a single Communication Manager system.

Voice mail retrieval button

Avaya Communication Manager supports the voice mail retrieval feature as a fixed feature button on the 2420 DCP and the 4602 telephone.

A field, “voice-mail Number: _____” appears on the **Station** screen for stations of type 2420 and 4602. The allowed values for this field are identical to the values allowed for an autodial feature button number. The field is a fixed field allowing entry of up to 16 digits that are auto-dialed to access the user's voice mail system.

- If the number field is blank, the voice mail retrieval button is treated like the “Transfer to Voice Mail” button.

If the number field is not blank, the voice mail retrieval button is treated like an autodial button.

Voice message retrieval

Voice message retrieval allows telephone users, remote access users, and attendants to retrieve leave word calling and call coverage voice messages. You can use voice message retrieval to retrieve your own messages or messages for another user. However, you can only retrieve messages for another user:

- from a telephone or attendant console in the coverage path
- from an administered system-wide message retriever
- if you are a remote-access user and you know the extension and associated security code

The system restricts unauthorized users from retrieving messages.

Voice messaging and call coverage

Often an AUDIX system is set up as the last point on a call-coverage path, as shown in Figure 3: Typical call coverage options on page 114. A secretary or colleague who answers a redirected call intended for you can also transfer the caller to your AUDIX mailbox. The caller may prefer to leave voice-mail for you if the message is personal, lengthy, or technical.

Many other options are available. For example, a caller can redirect a call from the AUDIX system to an attendant. Or the caller can transfer to another extension instead of leaving a message. You can even have the AUDIX automated attendant answer all calls to the company and send calls to various extensions. In this case, callers are instructed to enter keypad commands to direct the call.

Figure 3: Typical call coverage options

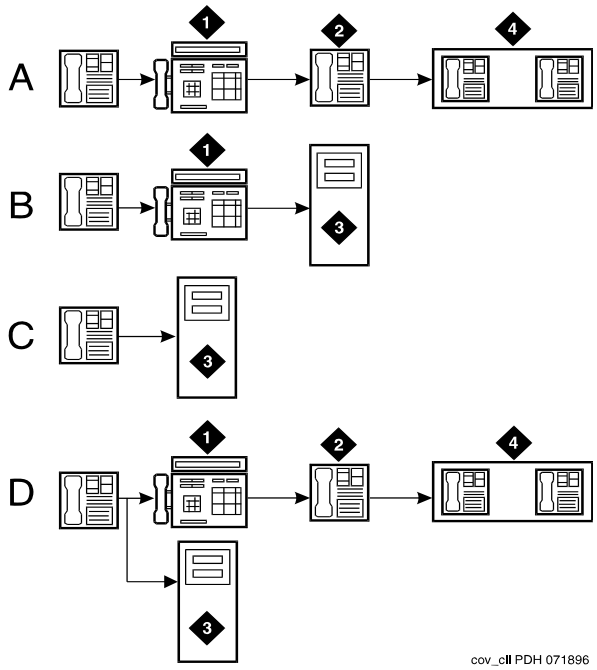

Figure notes:

A	External call: active, busy, do not answer	**1.**	Secretary
B	Internal call: cover all	**2.**	Clerk
C	Internal call: active, busy, do not answer	**3.**	AUDIX voice messaging
D	Internal call: send all calls	**4.**	Message center group

Chapter 10: Mobility

IP telephones or IP Softphones allow you to access the features of Avaya Communication Manager without having to be tied to one location. One of the major benefits of IP telephones is that you can move the telephones around on a LAN just by unplugging them and plugging them in somewhere else. One of the main benefits of IP softphones is that you can load them on a laptop PC, and then use the PC's modem to connect them to the switch from almost anywhere.

For more information, see Avaya IP Softphone on page 85.

Administration Without Hardware

This feature allows you to administer telephones that are not yet physically present on the system. This feature works the same as administration with hardware: when stations are moved, user-activated features such as call forwarding and send all calls are preserved and functional. This greatly facilitates the speed of setting up and making changes to the telephones on the system.

Automatic Customer Telephone Rearrangement

Automatic Customer Telephone Rearrangement (ACTR) allows a telephone to be unplugged from one location and moved to a different location without additional switch administration. The switch automatically associates the extension to the new port.

ACTR works with the 2420 DCP telephone and the 6400 serialized telephones. 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. ACTR uses the stored information and associates the telephone with new port when the telephone is moved.

ACTR makes it easy to identify and move telephones.

Avaya Wireless Telephone Solutions

Avaya Wireless Telephone Solutions (AWTS) is fully integrated with Communication Manager, and allows a user full access to Communication Manager features from a mobile telephone.

Note:

Avaya Wireless Telephone Solutions (AWTS) replaces the DEFINITY Wireless Business System (DWBS).

Avaya Extension to Cellular

The Avaya Extension to Cellular feature provides the expansion of mobile services, including one-number availability, increased user capacities, flexibility across facilities and hardware, more control over unauthorized usage, enhanced enable/disable capability, increased serviceability, and support of IP trunk facilities.

Avaya Extension to Cellular and off-PBX stations (OPS) provides users with the capability to have one administered telephone that supports Avaya Communication Manager features for both an office telephone and one outside telephone. Extension to cellular/OPS allows users to receive and place office calls anywhere, any time. People calling into an office telephone can reach users even if they are not in the office. Users could receive the call on their cell telephone, for example.

This added flexibility also allows them to use certain Communication Manager features from a telephone that is outside the telephone network.

Previous versions of Extension to Cellular allowed for office calls to be extended to the cell telephone of a user. Also, calls from the cell telephone would appear as if the call originated from the user office telephone when calling another telephone on the same call server. Certain features within Communication Manager are available from the cell telephone. These features are still available.

In previous versions of Extension to Cellular, cell telephones had to be administered as XMOBILE stations. This is no longer necessary with Communication Manager Release 2.0.

If you had administered Extension to Cellular in earlier releases of Communication Manager, you do not have to change the administration to continue using Extension to Cellular features. It still works. However, users would not have the full range of features that are now possible with Extension to Cellular/OPS.

Off-PBX station

With Avaya Communication Manager Release 2.0, the off-PBX station (OPS) application type is used to administer of a SIP telephone. OPS cannot be disabled using the Extension to Cellular enable/disable feature button.

Note:

A 4602 SIP telephone must register with the SIP proxy regardless of whether OPS is administered.

The Extension to Cellular/OPS application allows for many of the parameters used for the original Extension to Cellular application to be ported onto one of several DCP and IP station types. From a call processing perspective, Extension to Cellular/OPS is in fact dealing with a multi-function telephone, whereas the previous Extension to Cellular implementation utilized one or two XMOBILE stations that behaved like analog station types.

The Extension to Cellular/OPS application has its own documentation set. For a complete list of Extension to Cellular/OPS documentation, see your Avaya representative.

E911 ELIN for IP wired extensions

This feature automates the process of assigning an emergency location information number (ELIN) through an IP subnetwork (“subnet”) during a 911 call. The ELIN is then sent over either CAMA or ISDN PRI trunks to the emergency services network when 911 is dialed. This feature properly identifies locations of wired IP telephones that call an emergency number from anywhere on a campus or location.

Note:

This feature depends upon the customer having subnets that correspond to geographical areas.

This feature works for both types of IP endpoints:

- H.323
- SIP

A caller who needs emergency assistance dials a Universal Emergency Number - for example, 911 in the United States, 000 in Australia, and 112 in the European community. The call routes through a local Central Office, through an emergency tandem office, to the appropriate Public Safety Answer Point (PSAP). The PSAP answers the call.

A typical tandem office can route the call to a PSAP within at most four surrounding areas. (In the US, that translates to four surrounding area codes.) If the PSAP that receives the call is not the correct one to handle the emergency, the PSAP might be able to transfer the call to the correct PSAP. Such transfers can only occur between geographically adjacent or nearby PSAPs.

Each PSAP usually covers one city or one rural county. At the PSAP, emergency operators determine the nature of the emergency and contact the appropriate agency: police, fire, ambulance, etc. A single PSAP is usually responsible for an area covering several independent police and fire departments in the United States.

With Enhanced 911 (E911), the system might send to the emergency services network the Calling Party Number (CPN) with the call over Centralized Automatic Message Accounting (CAMA) trunks or through the Calling Number IE over ISDN trunks. A system at the PSAP uses the CPN to look up the documented street address location of the caller from the Automatic Location Information (ALI) database. The ALI database is usually owned and managed by Local Exchange Carriers. Many enterprise customers choose to contract with a third party to update the ALI database for them.

This depends on the assumption that a CPN always corresponds to the street address that the system owner arranged to have administered into the ALI database. This assumption is not always true.

- Users who have H.323 IP telephones can move them without notifying the system administrator.
- Users who have SIP IP telephones can use the same extension number simultaneously at several different telephones.

Without this feature, if these users dial 911, the emergency response personnel might go to the wrong physical location. With this feature, the emergency response personnel can now go to the correct physical location. In addition, emergency response personnel can now go to the correct physical location if a 911 emergency call comes from a bridged call appearance.

E911 device location for IP telephones

Communication Manager works with an E911 Manager device from RedSky Technologies. This third-party E911 Manager provides a flexible, complete, and automated E911 management system for customers who want to implement voice over IP (VoIP) telephony. The E911 Manager from RedSky Technologies works with Communication Manager release 2.1 and beyond to keep the Automatic Location Information (ALI) record for each extension correct. The E911 Manager also provides notification whenever someone moves an IP endpoint to a new subnet.

E911 SCCAN support for device location

This feature allows the user to call an emergency number from an SCCAN telephone on the customer's premises, and relaying the user's approximate physical location at the correct Public Safety Answering Point.

This feature automates the process of assigning an Emergency Location Extension number for each wireless Light Access Point port on the W310 Wireless LAN Gateway for use during an emergency call. The corresponding Emergency Location Information is then sent to the emergency services network when the emergency number is dialed. Public service personnel, who respond to the emergency call, can go to the approximate location of the caller, even when the caller uses an SCCAN telephone within the enterprise.

For more information about SCCAN, see Seamless Converged Communication Across Networks on page 120.

Personal Station Access

The Personal Station Access (PSA) feature allows you to transfer your telephone station preferences and permissions to any other compatible telephone. This includes the definition of telephone buttons, abbreviated dial lists, and class of service, and class of restrictions permissions. It can be used on-site or off-site (with DEFINITY® Extender).

PSA has several telecommuting applications. For example, several telecommuting employees can share the same office on different days of the week. The employees can easily and remotely make the shared telephone “theirs” for the day. Remote use requires DEFINITY Extender.

Do not answer reason code

The Personal Station Access (PSA) feature uses Administration Without Hardware (AWOH), a feature that allows the system administrator to assign a telephone without specifying a physical port. For example, use “X” as the port. If a telephone is disassociated, it means that it is not currently mapped to a particular physical telephone, such as a digital telephone. If a caller dials into an extension that is currently disassociated, they are provided a message that indicates “Don't answer” instead of “Busy”.

Name/number permanent display

When a person uses PSA to associate their extension with a station, a display appears on the station indicating their name and extension number. This information is displayed until the user disassociates their extension from the station using the PSA-associate feature access code.

Seamless Converged Communication Across Networks

The SCCAN (Seamless Converged Communication Across Networks) program consists of the joint collaboration of three companies, Motorola, Proxim, and Avaya. The purpose of SCCAN is to create and deploy a converged cellular Wireless Local Area Network (WLAN), and Internet Protocol (IP) Telephony services that delivers new levels of communications mobility and network connectivity.

The SCCAN system consists of new products including:

- A Wi-Fi/cellular dual-system telephone from Motorola (sometimes referred to as the Subscriber Unit)
- Session Initiation Protocol (SIP)-enabled IP Telephony software from Avaya (most often referred to as Avaya Communication Manager SIP Trunking)
- Voice enabled WLAN Infrastructure from Proxim (most often referred to as W310 Wireless LAN Gateway, formerly Converged Mobility Gateway (CMG)). The W310 is developed jointly by Avaya and Proxim.
- IPSec security client from Avaya (VPNRemote) that runs on the Motorola handset
- Light Access Points (LAPs) that are developed by Proxim

Terminal Translation Initialization

Communication Manager provides Terminal Translation Initialization (TTI), a feature that works with Administration Without Hardware (AWOH). TTI associates the terminal translation data with a specific port location through the entry of a special feature-access code, a TTI security code, and an extension number from a terminal that is connected to a wired (but untranslated) jack.

TransTalk 9000 digital wireless system

The TransTalk 9000 is a single-zone or dual-zone, in-building wireless system that provides a mobility solution on Communication Manager-based systems. It delivers the benefits and accessibility of a wireless telephone with all the power and functionality of a wired desk telephone.

X-station mobility

X-station mobility allows remote users to access switch features. That is, X-station mobility allows certain OEM wireless telephones remoted over a PRI trunk interface to be controlled by Communication Manager as if the telephones were directly connected to the switch.

The telephones are administered to be of the type XMOBILE and have additional administration information on the **Station** screen that assigns the capabilities of a remote station to the associated PRI trunk group. The wireless telephones thus have access to such features as call-associated display, bridging, message waiting, call redirection, and so forth.

X-station mobility is currently used for non-cellular wireless offers (DECT and PHS) in EMEA and APAC regions, and the Extension to Cellular offer globally.

Chapter 11: Port network and gateway connectivity

Asynchronous Transfer Mode

The Asynchronous Transfer Mode (ATM) switch is a replacement option for the CSS or the direct-connect switch. Several Avaya ATM switch types can provide Communication Manager port network connectivity. Non-Avaya ATM switches that comply with the ATM standards set by the European Union can also provide Communication Manager port network connectivity.

ATM WAN Spare Processor Manager

ATM WAN Spare Processor (WSP) Manager can be a key part of your emergency restoration and business continuity planning. This application enables users to download translations from a main server running Communication Manager, and simultaneously upload those translations to multiple (up to 15) ATM WAN Spare Processors (WSPs) over a LAN connectivity. This can be done according to a schedule specified by the administrator.You can schedule translations to run once now, or for a specified time and date in the future. You can also schedule regular daily or weekly updates.

The module also provides the ability to schedule regular daily or weekly updates of the Communication Manager translations. The ATM WAN Spare Processor Manager provides the current status of the main server running Communication Manager and any defined WSP devices in the network. A complete history log is created listing each of the switches, and the time and the resulting message from the scheduled action. On-line help is embedded into the module for ease of use.

Port Network Connectivity

ATM Port Network Connectivity (ATM-PNC) provides an alternative to the Center Stage Switch (CSS) configurations for connecting the Processor Port Network (PPN) to one or more Expansion Port Networks (EPN). ATM-PNC replaces the CSS in a DEFINITY R8r and later network with an ATM switch or network. ATM-PNC is available with all three Communication Manager reliability options - standard, high, and critical. In addition, it offers ATM-PNC duplication.

ATM-PNC integrates delivery of voice, video, and data via ATM over a converged large bandwidth network, providing reduced infrastructure cost and improved network manageability. ATM-PNC uses standards-based open interfaces that can be provisioned with either new or existing systems running Communication Manager.

Port Network Connectivity over WAN

ATM-PNC over a public Wide Area Network (WAN) represents an environment where the customer uses a service provider's ATM network between privately-owned ATM switches. The customer does not control the ATM switches in the network, including traffic policing policies and product quality.

Using a public WAN, Permanent Virtual Paths (PVP) may be set up between customer-owned ATM switches similar to the dedicated circuits in a private WAN. However, ATM cell processing occurs in a public WAN so the customer is dependent on ATM switches owned and managed by the service provider.

Switched Virtual Circuits (SVC) use the ATM protocol to transmit voice-like applications over ATM networks. The advantage of the SVC solution is that Communication Manager can dynamically signal the ATM network to provide more bandwidth as needed to handle peaks in the call traffic. If the ATM network cannot handle the additional traffic, calls will be denied.

WAN Spare Processor

An ATM WAN Spare Processor (WSP) provides a disaster recovery option for a media gateway G3r expansion port networks deployed over an ATM WAN.

An ATM WSP acts as a PPN in the event of a catastrophic failure in the network. The ATM WSP continually monitors a path to the PPN to determine if it is on-line and reachable. The WSP functions as a PPN if the main PPN is not functional or is not communicating to one or more of the other EPNs. From one to 15 ATM WSPs can be placed in a Communication Manager ATM port network configuration to provide a backup arrangement of PPNs, thus maintaining the availability of the Communication Manager features and functions.

Note:

ATM WSPs cannot be used with a conventional CSS.

Circuit switched

Center Stage Switch

Communication Manager supports CSS as a method to interface between the PPN and EPNs using circuit switched technology to carry the voice traffic.

Center Stage Switch separation

S8700 Media Servers in an Avaya MCC1 or SCC1 Media Gateway configuration, with four or more Port Networks (PN), use a Center Stage Switch (CSS) to interconnect the PNs.

The Center Stage Switch (CSS) separation feature allows for the physical separation of redundant S8700 Media Servers, and their corresponding CSS, to improve their survivability. Media Servers and the CSS can be separated up to 6.2 miles (10 km), providing backup and survivability for a communications network in one or more remote locations.

Internet Protocol

H.248 media gateway control

Communication Manager uses standards based H.248 to perform call control to Avaya media gateways such as the G700. H.248 defines a framework of call control signaling between the intelligent media servers and multiple “unintelligent” media gateways.

Inter-Gateway Alternate Routing

For single-server systems that use the IP-WAN to connect bearer between port networks or media gateways, Inter-Gateway Alternate Routing (IGAR) provides a means of alternately using the public switched telephone network (PSTN) when the IP-WAN is incapable of carrying the bearer connection. IGAR may request that bearer connections be provided by the PSTN under the following conditions:

- The number of calls allocated or bandwidth allocated via Call Admission Control-Bandwidth Limits (CAC-BL) has been reached
- VoIP RTP resource exhaustion in a MG/PN is encountered
- A codec set is not specified between a network region pair
- Forced redirection between a pair of network regions is configured

IGAR takes advantage of existing public and private network facilities provisioned in a network region. Most trunks in use today can be used for IGAR. Examples of the better trunk facilities for use by IGAR are:

- Public or Private ISDN PRI/BRI
- R2MFC

IGAR provides enhanced Quality of Service (QoS) to large distributed single-server configurations.

Network Region Wizard

For large distributed single-server systems that have multiple network regions, the Network Region Wizard (NRW) simplifies and expedites the provisioning of multiple IP network regions, including Call Admission Control using Bandwidth Limits (CAC-BL) and Inter-Gateway Alternate Routing (IGAR).

IP Port Network Connectivity

Communication Manager allows Control Channel Message Set (CCMS) messages to be packetized over IP LAN and WAN connections to control multiple port networks.

Link Recovery

IP calls must have an H.248 link between the Avaya G700 Media Gateway and the call controller. The H.248 link between an Avaya server running Communication Manager and the Avaya Media Gateway provides the signaling protocol for:

- Call setup
- Call control (user actions such as Hold, Conference, or Transfer)
- Call tear-down

If the link fails for any reason, the Link Recovery feature preserves any existing calls and attempts to re-establish the original link. If the gateway cannot reconnect to the original server, then Link Recovery automatically attempts to connect with alternate TN799DP (CLAN) circuit packs within the original server configuration or to a Local Spare Processor (LSP).

Link Recovery does not attempt to recover or overcome any network failure that created the link outage. Link Recovery also does not diagnose or repair the network failure that caused the link outage.

Since there is no communication possible between the Media Gateway and call controller during a link outage, button depressions are not recognized, feature access does not work, and neither does any other type of call handling. In essence, the system is unresponsive to any stimuli until the H.248 link is restored. This might be the only indication that a Link Recovery is in process.

CAUTION:

If an administrator attempts to add a telephone to a gateway while that gateway is in Link Recovery, that station is not put into service when the gateway comes back. If this happens, perform a **busyout/release** command on that telephone when the gateway comes back into service.

Feature highlights:

- Call signaling channel failures are detected at a fast rate (in the order of 30 seconds, by default).
- The endpoint has an awareness about the primary and the alternate gatekeepers for the purpose of faster and less disruptive recovery from signaling channel failures.
- The endpoint attempts to re-establish the signaling channels with the primary gatekeeper while preserving an existing call.
- An IP endpoint's registration (while it is recovering from a signaling failure) can be accepted while preserving that endpoint's existing call(s).
- The customer can administer the endpoint recovery parameters (such as timers and gatekeepers).

In order for IP endpoints to take advantage of this feature, the firmware or application software must be updated with the new algorithm that supports the resiliency feature. IP endpoints include IP telephones and IP softphones. However, since the feature provides backward compatibility, it is not mandatory that existing IP endpoints be upgraded.

Separation of Bearer and Signaling

The Separation of Bearer and Signaling (SBS) feature provides a low cost virtual private network with high voice quality for customers who cannot afford private leased lines. SBS provides a DCS+ VPN replacement for those customers needing Dial Plan Expansion (DPE) functionality.

Note:

DCS does not work with six-digit or seven-digit dial plans. Although QSIG does work with six-digit and seven-digit dial plans, QSIG does not work over VPNs.

The SBS feature supports:

- QSIG private networking signaling over a low cost IP network
- Voice (bearer) calls over public switched network
- Association between QSIG feature signaling information and each voice call

You must always use AAR/ARS/UDP to originate an SBS call. You cannot use a Trunk Access Code / Dial Access Code to originate an SBS call.

Proper administration and configuration is required for SBS calls to work correctly. This includes:

- Fields in the **System-Parameters Features** screen, a field on the **Trunk Group** screen, and a **Station** type called an **SBS Extension** (an extension number without hardware assigned to it that is used to associate the separate bearer and signaling calls).
- Customers must allocate a sufficient number of SBS extensions based on expected SBS traffic volume. The same applies to SBS trunk group members.
- Each administered SBS extension must correspond to a DID/DDI number obtained from a local service provider.

  Note:

  Obtaining a DID/DDI number for each SBS extension is not necessary if the Feature Plus Pseudo DID feature is available.

- In remote office configurations or other remote gateway configurations with limited direct network access, these DID/DDI numbers should be obtained from a service provider that is local to the controlling gateway server, not local to the remote office/gateway. This eliminates excessive traffic through the remote office/gateway to its controlling gateway server.
- The **ISDN Public-Unknown Numbering** screen must be correctly administered to map every SBS extension to the corresponding national public network complete number (that is, the DID/DDI number). This screen is used to develop the complete number even if the incoming SBS trunk group numbering format is administered for private numbering.

Chapter 12: Trunk connectivity

Asynchronous Transfer Mode

See Asynchronous Transfer Mode on page 123.

Circuit Emulation Service

ATM-circuit emulation service (ATM-CES) lets Communication Manager emulate ISDN-PRI trunks on an ATM facility. These virtual trunks can serve as integrated access, tandem, or tie trunks.

ATM-CES trunk emulation maximizes port network capacities by consolidating trunking. For example, the CES interface can define up to eight virtual circuits for tie-line connectivity, consolidating onto one circuit card network connectivity that usually requires multiple circuit packs.

CMS measurement of ATM

See CMS measurement of ATM on page 66.

Circuit switched

DS1 trunk service

Bit-oriented signaling that multiplexes 24 channels into a single 1.544-Mbps stream. DS1 can be used for voice or voice-grade data and for data-transmission protocols. E1 trunk service is bit-oriented signaling that multiplexes 32 channels into a single 2.048-Mbps stream. Both DS1 and E1 provide a digital interface for trunk groups. Digital Service 1 (DS1) trunks can be used to provide T1 or ISDN Primary Rate Interface (PRI) service.

Echo cancellation - with UDS1 circuit pack

The universal DS-1 (UDS1) circuit pack (TN464GP/TN2464BP) available for all Communication Manager platforms has echo cancellation circuitry. The echo cancellation capability of the circuit pack is intended only for channels supporting voice communication. It is not desirable to provide echo cancellation over channels supporting data communication.

The TN464GP/TN2464BP is intended for Communication Manager customers who are likely to encounter echo over circuits connected to the public network. The occurrence of echo is likely if Communication Manager is configured for complex services such as ATM or IP. In addition, echo is likely to occur if Communication Manager interfaces to local service providers who do not routinely install echo cancellation equipment in all their circuits.

E1

Communication Manager also supports E1 connections. T1/E1 access and conversion allows simultaneous connection to both T1 (1.544 Mbps) and E1 (2.048 Mbps) facilities (using separate circuit packs).

T1

When planning your networking requirements, one of the options you should consider is multiplexing over digital services 1 (DS1) facilities.

Separate licensing for TDM stations and TDM trunks

Prior to release 2.0, Communication Manager was sold by licensed ports that included stations and trunks. The system displayed the total of licensed ports in the **Maximum Ports** field on the **Optional Features** screen.

As of release 2.0 of Communication Manager, Avaya sells licenses for stations, but not trunks. Currently, the **Maximum Ports** field on the **Optional Features** screen is used for licensing ports, which include both trunks and stations.

With Communication Manager release 3.0, a separate field, **Maximum Stations**, is created on the **Optional Features** screen to track station licenses only. This helps customers easily identify the number of station licenses on the system.

Internet Protocol

H.323 trunk

A TN802B in MedPro mode or a TN2302AP IP interface enables H.323 trunk service using IP connectivity between two systems running Communication Manager. The H.323 trunk groups can be configured as system-specific tie trunks, generic tie trunks, or direct-inward-dial (DID) public trunks. In addition, the H.323 trunks support ISDN features such as QSIG and BSR.

IP loss groups

A primary reason to accomplish a loss plan for voice communication systems is the desire to have the received speech and tone loudness at a comfortable listening level. This should be accomplished so that users can listen to each other without being concerned who or where the remote party is, or what kind of telephone equipment each may be using.

A connection with an end-to-end loss (called an Overall Loudness Rating) of 10 dB - which approximates a normal conversation between a talker and listener spaced one meter apart - provides a high degree of satisfaction for the majority of users. Therefore, voice communication standards for end-to-end loss are based on this number.

Communication Manager has now defined two additional loss groups for IP telephony. The purpose of these two loss groups is to set speech and tone loudness separately for IP connections. These loss groups use country-specific gateway loss plans.

The two IP loss groups are:

- Loss Group 18: IPtrunk - loss group for IP trunks (IP Carrier Medium)
- Loss group 19: IPphone - loss group for IP terminals (IP ports)

On an upgrade, if the default for an IP station loss plan is 2, and the IP trunk loss plan is 13, Communication Manager changes the defaults to 19 and 18 respectively.

IP trunks

IP trunk groups may be defined as virtual private network tie lines between systems or ITS-E servers running Communication Manager. Each IP trunk circuit pack provides a basic 12-port package that can be expanded up to a total of 30 ports. The number of ports that are defined will correspond to the total number of simultaneous calls transmitted over the IP trunk interface.

The benefits of IP trunk include a reduction in long distance voice and fax expenses, facilitating global communications, providing a full function network with data and voice convergence and optimizing networks by using the available network resources.

IP trunking is a good choice for basic, corporate voice and fax communications, where cost is a major concern. IP trunk calls travel over a company intranet rather than the public telephone network. So, for the most common types of internal corporate communications, IP trunks offer considerable savings.

IP trunking is usually not a good choice for applications where calls have to be routed to multiple destinations (as in most conferencing applications) or to a voice messaging system. IP trunk calls are compressed to save network bandwidth. Repeated compression and decompression results in a loss of data at each stage and degrades the final quality of the signal.

The maximum number of compression cycles acceptable on a call is three, and three compression cycles can compromise voice quality. Normal corporate voice or fax calls typically go through fewer than three compression cycles. However, multipoint conference calls and most voice messaging systems add too many compression cycles for acceptable quality.

Session Initiation Protocol

Session Initiation Protocol (SIP) is a signalling protocol used for establishing sessions in an IP network.

SIP has a separate set of documentation. For more information on SIP, click the **documentation** link from the http://www.avaya.com Web site.

SIP trunks

SIP trunking functionality allows a Linux server to function as a POTS gateway between traditional legacy endpoints (stations and trunks) and SIP endpoints. It also provides SIP to SIP routing. In the routing scenario, the server supports call routing similar to what a SIP proxy would provide.

SIP links can be secured using TLS to encrypt signaling, and use Digest Authentication to perform validation. When using TLS, the Media Encryption feature is also available to encrypt audio channels.

SIP trunking functionality:

- Provides access to less expensive local and long distance telephone services, plus other hosted services from SIP service providers
- Provides presence and availability information to members of the enterprise and authorized consumers outside the enterprise, including other enterprises and service providers
- Facilitates SIP-enabled converged communications applications within the enterprise, such as the Seamless Service Experience.

Allowing encryption of signaling and audio channel provides the customer with the option to provide a secure communications infrastructure. See Signaling encryption for SIP trunks on page 200.

Auxiliary trunks

Auxiliary trunks connect devices in auxiliary cabinets with Communication Manager. Some of the features that are supported with this type of trunk are recorded announcements, telephone dictation service, malicious call trace, and loudspeaker paging.

Advanced Private Line Termination

Provides access to and termination from CO (Central Office)-based private networks; namely, Common Control Switching Arrangements (**CCSA**) and Enhanced Private Switched Communications Service (**EPSCS**).

APLT trunks are physically the same as those used for analog tie trunks, where the trunk signaling is compatible with **EPSCS** and **CCSA** network switches. The outgoing **APLT** trunk repeats any number of digits to the private network as dialed. **APLT** trunks can tandem through the **PBX** from **EPSCS** network only; **CCSA** networks require an Attendant to complete the call.

Central Office

Central Office (CO) trunks connect Communication Manager to the local central office for incoming and outgoing calls.

Central Office support on G700 Media Gateway - Russia

See Central Office support on G700 Media Gateway on page 97.

Digital multiplexed interface

The digital multiplexed interface feature supports two signaling techniques: bit-oriented signaling and message-oriented signaling for direct connection to host computers.

Digital multiplexed interface offers two major advantages:

- digital multiplexed interface delivers a standard, single-port interface for linking host computers internally and externally through a T1 carrier.
- Since it is compatible with ISDN standards and is licensed to numerous equipment manufacturers, digital multiplexed interface promotes multi-vendor connectivity.

Communication Manager supports two versions of digital multiplexed interface, each differing in the way information is carried over the 24th channel:

- Bit-oriented signaling
- Message-oriented signaling

Bit-oriented signalling

Digital multiplexed interface bit-oriented signalling carries framing and alarm data and signalling information for connections to host computers and other vendor equipment.

Message-oriented signalling

Digital multiplexed interface message-oriented signalling, fully compatible with ISDN-PRI, uses the same message-oriented signalling format - link access procedure on the D-channel - as ISDN-PRI for control and signalling. These signalling capabilities extend the advantages of digital multiplexed interface message-oriented signalling multiplexed communications to the public ISDN network.

Direct Inward Dialing

Direct Inward Dialing (DID) trunks connect Communication Manager to the local central office for incoming calls dialed directly to stations without attendant assistance.

Direct Inward/Outward Dialing

Traditionally, Central Office (CO) trunks and Direct Inward Dialing (DID) trunks interface an attendant console with a central office. A CO trunk services outgoing calls and accepts incoming calls that are terminated at the attendant. A Direct Inward/Outward Dialing (DIOD) trunk is used for calls that need to be terminated without an attendant interaction.

E&M signaling - continuous and pulsed

See E&M signaling - continuous and pulsed on page 98.

E911 CAMA trunk group

This screen administers the Centralized Automatic Message Accounting (CAMA) trunks and provides Caller Emergency Service Identification (CESID) information to the local enhanced 911 system through the local central office.

Foreign Exchange

Foreign Exchange (FX) trunks connect Communication Manager to a Central Office other than to the local office.

ISDN trunks

Gives you access to a variety of public and private network services and facilities. The ISDN standard consists of layers 1, 2, and 3 of the Open System Interconnect (**OSI**) model. Systems running Communication Manager can be connected to an ISDN using standard frame formats: Basic Rate Interface (BRI) and the Primary Rate Interface (PRI).

An **ISDN** provides end-to-end digital connectivity and uses a high-speed interface that provides service-independent access to switched services. Through internationally accepted standard interfaces, an **ISDN** provides circuit or packet-switched connectivity within a network and can link to other **ISDN** supported interfaces to provide national and international digital connectivity.

Automatic Termination Endpoint Identifier

The user side will support automatic TEI assignment by the network. Both fixed and automatic TEI assignment will be supported on the network side.

Call-by-call service selection

Enables a single ISDN-PRI trunk group to carry calls to a variety of services, rather than requiring each trunk group to be dedicated to a specific service. It allows you to set up various voice and data services and features for a particular call.

ETSI functionality

The full set of ETSI public-network and private-network ISDN features is officially supported. This includes Look-Ahead Interflow (LAI), look-ahead routing, and usage allocation.

Also included is all QSIG supplementary services, such as:

- Name identification
- Call diversion (including rerouting)
- Call transfer
- Path replacement

ETSI functionality does not include:

- DCS
- Non-facility associated signaling
- D-channel backup
- Wideband signaling

Facility and non-facility associated signaling

Facility and non-facility associated signaling allows an **ISDN**-**PRI DS1**/E1 interface D-channel to carry signaling information for B-channels (voice or data). D-Channel Backup can also be administered to increase system reliability.

Feature plus

Feature plus enables those users without DID service to direct dial users on a remote PBX through the public network. ISDN feature plus eliminates the need for attendant intervention for those without DID capabilities.

ISDN-Basic Rate Interface

Enables connection of the system to equipment or endpoints that support an Integrated Services Digital Network (ISDN) by using a standard format called the Basic Rate Interface (BRI). This feature is a 192-Kbps interface that carries two 64-Kbps B-channels and one 16-Kbps D-channel.

ISDN is a global access standard that uses a layered protocol. It eliminates the need for multiple, separate access arrangements for voice, data, facsimile, and video services and networks. Using the same pair of wires that carry simple telephone calls, ISDN can deliver voice, data, and video services in a digital format.

The ISDN-BRI Trunk circuit pack allows Communication Manager to support the T interface and the S/T interface as defined by ISDN standards (ITU-T recommendation I.411). The circuit pack provides eight ports to the network and supports two B channels and one D channel.

The ISDN-BRI Trunk provides the following advantages:

- Provides an inexpensive way to connect to ISDN services provided by the network provider
- Meets almost all ETSI Country protocol requirements
- Supports essential (not supplementary) ISDN services

BRI trunks support public-network access outside the U.S. on point-to-midpoint connections, with the restriction that Communication Manager must not be configured in a passive bus arrangement with other BRI endpoints. ISDN-BRI trunks can be used as inter-PBX tie lines using the QSIG peer protocol. See Figure 4: Communication Manager and ISDN on page 138.

Figure 4: Communication Manager and ISDN

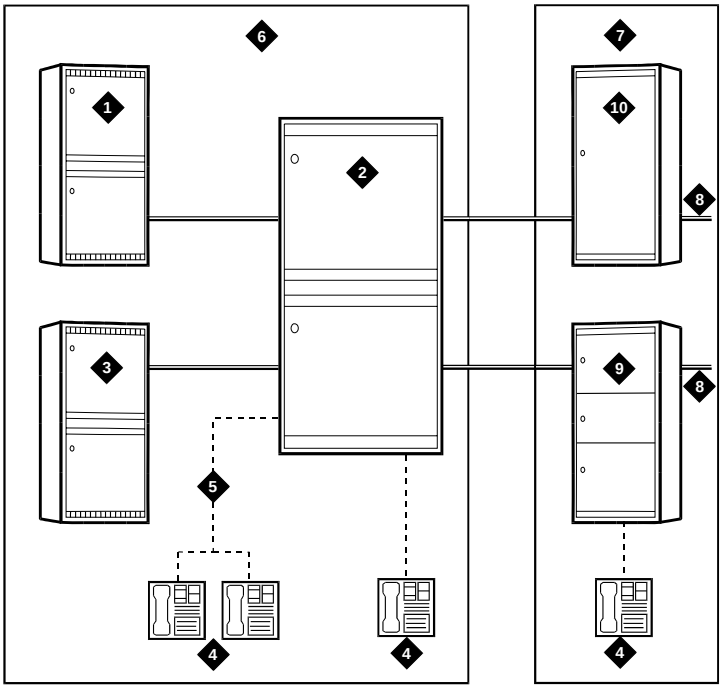

Figure notes:

1.	System running Communication Manager	**6**	Private ISDN (can be carried over ATM-CES)
2.	System running Communication Manager	**7.**	Public ISDN (can be carried over ATM-CES)
3.	System running Communication Manager	**8**	Public and private networks
4.	Basic rate interface telephone	**9**	Central office switch
5.	Passive bus	**10.**	Tandem switch

Multiple subscriber number - limited

The ISDN standard MSN feature lets customers assign multiple extension to a single BRI endpoint. The MSN feature works with BRI endpoints that allow the channel ID IE to be encoded as “preferred.”

NT interface on TN556C

Communication Manager supports the NT (network) side of the T interface using the TN556C circuit pack. This gives the switch full tie trunk capability using BRI trunks. Communication Manager supports leased BRI connections through the public network, with a TN2185 on each end of the leased connection. Communication Manager will not, however, allow customers to administer both endpoints and trunks on the same TN556C circuit pack.

Presentation restriction

Restricts the display of calling/connected numbers over ISDN trunks. ISDN trunk groups can be administered to control the display of calling/connected numbers. Each trunk group can be administered to display “presentation restricted,” “number no available due to networking,” or an administered text string instead of the calling/connected number.

Wideband switching

Provides the ability to dedicate two or more **ISDN** B-channels or **DSO** endpoints for applications that require large bandwidth. Certain applications, such as video conferencing and high-speed data transmission, require extra bandwidth and it becomes necessary to put several ISDN-PRI narrowband channels into one wideband channel to accommodate the needs of these applications.

This feature supports both European and North American standards.

Multi-Frequency Packet signaling - Russia

See Multi-Frequency Packet signaling on page 97.

National private networking support - Japan

See National private networking support.

Personal Central Office Line

Provides a dedicated trunk circuit between multi-appearance telephones and a CO or other switch via the network.

Release Link Trunks

Release Link Trunks (RLT) are used between switch locations to provide centralized attendant service or automatic call distribution group availability.

Remote access trunks

Tie trunks

Tie trunks carry communications between Communication Manager and other switches in a private network. Several types of trunks can be used, depending on the type of private network you establish.

Timed automatic disconnect for outgoing trunk calls

This feature provides the capability to automatically disconnect an outgoing trunk call after an administrable amount of time. The amount of time that can elapse before the trunk is dropped can be specified, and can vary between 2 and 999 minutes. If the timer field is blank (the default value), the feature is disabled and the trunk will not be automatically disconnected.

Timed call disconnection applies to all outgoing trunk calls initiated by a party belonging to a specified Class of Restriction (COR).

The outgoing trunk disconnect timer only affects outgoing public network trunks (CO, DIOD, FX, WATS, and ISDN public-network).

Note:

The outgoing trunk disconnect timer should be administered to a value large enough to provide users with adequate response time.

The outgoing trunk disconnect timer does not apply to outgoing trunk calls that are emergency or service calls. Specifically, the outgoing trunk disconnect timer does not apply to calls with ARS call types `alrt`, `emer`, `nsvc`, `op`, `svcl`, `svfl`, `svct`, or `svft`.

The outgoing trunk disconnect timer starts after the outgoing trunk call is answered. The outgoing trunk call is considered answered if:

- the network provides an answer supervision line signal
- an ISDN CONNect message is received
- the Answer Supervision Timeout timer expires
- the call classifier classifies the call as answered
- the Outgoing End of Dial Timer expires

Prior to disconnecting the trunk, warning tones are applied to all parties on the call. The first warning tone occurs when one minute remains on the call. The second warning tone occurs when 30 seconds remain on the call.

Wide Area Telecommunications Service

Wide Area Telecommunications Service (WATS) trunks allow you to place long-distance outgoing voice-grade calls to telephones in defined service areas. The calls are priced according to distance in the service area, length of the call, time of day, and the day of the week.

Chapter 13: Public Networking and connectivity

Caller ID on analog trunks

Caller ID on analog trunks allows the system to accept calling name information from a Local Exchange Carrier (LEC) network that supports the Bellcore calling name specification. The system can send calling name information in the format if Bellcore calling name ID is administered.

Caller ID on digital trunks

In the United States, the telephone of a user displays calling party information (if the telephone is a display telephone). Name and calling number are available from the US central offices.

This feature may be used in countries that comply with either US. The display of name and number will work with all Communication Manager digital telephones (DCP and BRI) equipped with a 40-character or a 32-character alphanumeric display.

DS1 trunk service

See DS1 trunk service on page 129.

Echo cancellation - with UDS1 circuit pack

See Echo cancellation - with UDS1 circuit pack on page 130.

E1

See E1 on page 130.

T1

See T1 on page 130.

Flexible billing

See Flexible billing on page 53.

Local exchange trunks

Local exchange trunks connect Communication Manager to a central office. The following local exchange trunks are some of the types available.

800-service trunks

800-service trunks let your business pay the charges for inbound long-distance calls so that callers can reach you toll-free.

Central Office trunks

See Central Office on page 133.

Digital Service 1 trunks

See DS1 trunk service on page 129.

Direct Inward Dialing trunks

See Direct Inward Dialing on page 134.

Direct Inward/Outward Dialing trunks

See Direct Inward/Outward Dialing on page 135.

Foreign Exchange trunks

See Foreign Exchange on page 135.

Wide Area Telecommunications Service

See Wide Area Telecommunications Service on page 141.

Chapter 14: Intelligent networking

Avaya VoIP Monitoring Manager

See Avaya VoIP Monitoring Manager on page 221.

Distributed Communications System protocol

The Distributed Communications System (DCS) protocol allows you to configure two or more switches as if they were a single, large system. DCS provides attendant and voice-terminal features between these switch locations. DCS simplifies dialing procedures and allows transparent use of some of the Communication Manager features. (Feature transparency means that features are available to all users on DCS regardless of the switch location.) For more information, also see Centralized Attendant Service on page 46 and Inter-PBX attendant calls on page 43.

Attendant with DCS

Direct trunk group selection

See Direct trunk group selection on page 43.

Display

See Display on page 47.

DCS automatic circuit assurance

See DCS automatic circuit assurance on page 210.

DCS over ISDN-PRI D-channel

Enhances DCS by allowing access to the public network for DCS connections between DCS switch nodes. With this feature (also known as DCS Plus or DCS+), DCS features are no longer restricted to private facilities. The ISDN-PRI B-channel is used for voice communications, and the ISDN-PRI D-channel is used to transport DCS control information.

DCS protocol - Italy

See Distributed Communications Systems protocol on page 96.

DCS with reroute

Allows a DCS call to be rerouted over a different path if the switch finds a better quality and lower cost route. This feature allows for rerouting the call after a transfer or rerouting during a call. DCS with reroute is similar to the rerouting capabilities used with QSIG.

QSIG/DCS voice mail interworking

See QSIG/DCS voice mail interworking on page 112.

Electronic Tandem Network

In an Electronic Tandem Network (ETN) - also known as Private Network Access (PNA) - Communication Manager provides a variety of features on a network-wide basis. It allows calls to other systems in a private network. These calls do not use the public network. Instead, they are routed over your dedicated facilities.

Automatic alternate conditional routing

You can control the routing of particular calls using conditional routing. For example, you can limit the number of communications satellite hops (communications satellite links used as trunks) in any end-to-end private network routing pattern. Limiting the number of satellite hops may be desirable for controlling transmission quality or call delay in both voice and data calls.

Trunk signaling and error recovery

The reliability of electronic tandem network calls is improved by allowing a trunk call to be retried on another circuit when signaling failures occur.

- **tandem switch**: A switch within an ETN that provides the logic to determine the best route for a network call, possibly modifies the digits outpulsed, and allows or denies certain calls to certain users.
- **tandem through**: The switched connection of an incoming trunk to an outgoing trunk without human intervention.
- **Tandem Tie-Trunk Network (TTTN)**: A private network that interconnects several customer switching systems.

See also Port network and gateway connectivity on page 123.

Extension number portability

When employees move within the network, they can retain their extension numbers. The ability to keep extension numbers, and even electronic tandem network and direct inward dialed numbers, when moving to other locations within the company eliminates missed calls and saves valuable time.

Internet Protocol

The capabilities and applications of Communication Manager are extended using IP. Communication Manager IP supports audio/voice over a LAN or WAN, and it ensures that remote workers have access to communication system features from their PCs. Communication Manager also provides standards based control between media server and media gateways allowing communications infrastructure to be distributed to the edge of the network.

The Communication Manager IP engine offers features that enables users to increase the quality of voice communications. The Quality of Service (QoS) feature enables users to administer and download the differentiated services type-of-service value to optimize voice quality. The QoS feature reduces latency by implementing buffers in the audio-processing board, and assists some routers in prioritizing audio traffic.

Communication Manager IP also includes hairpin and IP-IP direct connections, two features that make voice communications more efficient. These features increase the efficiency of voice communications by reducing both per port costs and IP bandwidth usage.

IP solutions supports trunks, IP communications devices, IP port networks, and IP control for media gateways. IP solutions is implemented using various IP-media processor circuit packs inside the servers or the Avaya media gateways. The IP media processors provides H.323 trunk connections and H.323 voice processing for IP telephones. The features that use the IP media processor also require the CLAN circuit pack or native processor ethernet connectivity.

The IP LAN can also connect through VPN and WAN facilities to extend the customer IP network across geographically disparate locations. Distributed communication services (DCS+), or QSIG services, can extend feature transparency, centralized voice mail, centralized attendant service, call center applications, and enhanced call routing across IP trunks.

Note:

To maximize voice quality using IP, you must consider both your hardware and network configurations. For example, with IP softphones, you can send the audio over traditional circuit switch lines, providing high quality voice, or over IP using LAN connections. The IP network must be a switched ethernet infrastructure and have the appropriate engineering to accommodate bandwidth, latency and packet loss requirements to effectively provide for real-time voice over IP traffic.

Alternate gatekeeper and registration addresses

When an IP endpoint (including softphones, IP telephones, and Avaya R300) registers with the switch, the switch sends back an IP registration address. The switch sends a different IP address for each registration according to a cyclic algorithm.

If registration with the original CLAN circuit pack IP address is successful, the switch sends back the IP addresses of all the CLAN circuit packs in one network region, not including interconnected regions. These CLAN addresses are called gatekeeper addresses. These addresses can also be used if the call signaling on the original CLAN circuit pack fails.

Note:

On switches using the LAN region based on IP Address feature, it is likely that the network region number assigned to an IP telephone would be different from the network region number of the TN799 that the telephone is registering through. That difference would mean the list of TN799 addresses in the same network region as the IP telephone would be empty. The alternate gatekeeper feature would send a blank list to the IP telephone.

To prevent that from happening, an IP terminal registers with Communication Manager. Communication Manager then sends to the endpoint the IP addresses of the CLANs in the same region as the terminal, followed by network regions interconnected with the network region of the terminal.

If the network connection to one CLAN circuit pack fails, the IP endpoint re-registers with a different CLAN. Alternate gatekeeper and registration addresses, and CLAN circuit pack load sharing, spread IP endpoint registration across more than one CLAN circuit pack, increasing performance and reliability.

Classless Interdomain Routing

Classless Interdomain Routing (CIDR) is a redefinition of the subnet mask, allowing for the aggregation of contiguous classful networks under a single network definition. This allows for more efficient routing table management when administering IP address on Communication Manager.

Multiple network regions per CLAN

Multiple network regions per CLAN enables a single CLAN to provide registration and call control to IP endpoints in multiple network regions. Communication Manager implements this approach by allowing IP addresses to be mapped to network regions in a mapping screen, instead of just to a CLAN. When an IP telephone registers, the switch determines the telephone's network region number based on the telephone's IP address.

Multiple location support for network regions

Multiple location support for network regions allows remote Avaya media gateways connected to a central Avaya media server to retain:

- Local user time
- Local ARS public analysis tables for local trunking
- Automatic daylight savings time
- Local touch tone receivers for IP communications devices, such as Avaya IP telephones.

Communication Manager allows administrators to map locations to IP network regions.

Network regions

Network regions provide the administrative foundation on which Communication Manager features are allocated to IP endpoints. A network region is a collection of IP endpoints and switch IP interfaces interconnected by an IP network.

Endpoints that share network regions typically represent users with common interests. For example, a customer might have two separate small campuses in a large metropolitan area, interconnected by a WAN, and both served by the same server running Communication Manager. Communication Manager allows the customer to define a network region for each campus, and associate each of their CLAN and IP media processor circuit packs with these regions.

Quality of Service

By employing a variety of Quality of Service (QoS) features, Communication Manager provides the best possible end-to-end audio experience when all or part of the audio path is carried over packet facilities. “Best” in this context is defined by the customer as represented by the system administrator, and represents a trade-off between audio reproduction quality, audio path delay (latency), audio loss, and network resource consumption.

802.1p/Q

IEEE standard 802.1Q and 802.1p provide the means to specify both a Virtual LAN (VLAN) and a frame priority at layer 2 for use by LAN hubs, or bridges, that can do routing based on MAC addresses. 802.1p/Q provides for 8 levels of priority (3 bits) and a large number (12 bits) of VLAN identifiers. The VLAN identifier at layer 2 permits segregation of traffic to reduce traffic on individual links. Because 802.1p operates at the MAC layer, its presence may vary from LAN segment to LAN segment within a single network region. Flexibility requires that 802.1p/Q options be administered individually for each network interface.

Camp-on/Busy-out

A camp-on/busy-out command is commonly used by system technicians to busy-out system resources that need maintenance or repair. Without it, all active calls using those resources are indiscriminately dropped if the resource is physically removed from the system. This disruptive action causes problems for customers, especially when a large number of calls are torn down.

The Camp-on/Busy-out feature for Prowler, MedPRO, and Cruiser adds the ability to remove idle VoIP resources from the system pool of available VoIP resources.

Note:

This feature is not supported by the G700 or G350 Media Gateway platforms.

The Camp-on/Busy-out feature enables the user to select the media processor to be busied-out while the media processor is still in service. After a call ends that was using resources within the specified media processor, the idled resource is removed from the system pool of available resources. Once all of the media processor resources are in a “busy-out” state, the associated board can be removed from the system without disrupting active calls.

Call Admission Control bandwidth management

In order to ensure Quality of Service for Voice over IP calls, there is a need to limit overall VoIP traffic on WAN links. The Call Admission Control (CAC) Bandwidth Management feature of Communication Manager allows the customer to specify a VoIP bandwidth limit between any pair of IP network regions. The feature then denies calls that need to be carried over the WAN link that exceed that bandwidth limit.

CLAN load balancing

CLAN load balancing is the process of registering IP endpoints to CLAN circuit packs (TN799x). Load balancing occurs among CLANs within a network region.

IP endpoint registration among CLAN circuit packs is done through an algorithm. This algorithm tracks the number of sockets being used per TN799x circuit pack, and registers IP endpoints to the TN799x with the most available (unused) sockets. This algorithm applies to H.248, H.323 signaling groups, H.323 stations, and SIP endpoints. Sockets used by adjuncts are not included in the socket count.

Codecs

A codec (coder/decoder) provides the means by which audio is compressed. A codec is typically used in VoIP. Codecs supported by Communication Manager include G.711, G.723, and G.729.

Differentiated services

With the Differentiated Services (DiffServ) option, the system administrator can administer (by region) and download, to the TN2302AP, the DiffServ Type-Of-Service (TOS) value. This allows data networking equipment to prioritize the audio stream at the IP level to promote voice quality. DiffServ makes use of the TOS octet in the existing IP version 4 header. As such, it may be set by information senders and used by IP (layer 3) routers within the network.

Dynamic jitter buffers

Propagation delay and jitter is caused when a human's voice is sampled, encoded, and packetized for transport over the IP network, but is received and decoded at different rates. Jitter buffers are used to buffer the audio output to smooth the conversions. Communication Manager provides dynamic jitter buffers to balance both delay of conversation and rapid bursts that may occur.

Integration with Cajun rules

Cajun rules provide a central repository for QoS parameters and allows comprehensive QoS management across routers, switches, and endpoints. QoS parameters and policies are assigned according to network regions on a network region and are distributed through enterprise directory gateway to Communication Manager and to routers and switching devices.

IP overload control

This enhancement more effectively manages processor occupancy overload situations. The enhancement applies selected overload mechanisms at a lower occupancy threshold in an effort to avoid more serious symptoms experienced at higher occupancy levels.

The IP overload control enhancement:

- fortifies the system against bursts of registration traffic
- provides a mechanism to alert the far-end to abstain from issuing registrations for some specified period of time
- records the event to maintain a history of potential performance problems
- optimizes the maximum number of simultaneous registrations the server can handle based on the available memory and CPU cycles
- reduces the frequency that a server might go into overload due to network problems

QoS for call control

Communication Manager allows QoS for the signaling packets coming from gatekeepers such as the CLAN by employing the same standards based DiffServ and 802.1p/Q schemes as with audio channels. This QoS services further improve the users VoIP audio experience.

QoS for VoIP

Communication Manager implements QoS through the selection of audio codec such as G.711, G.723 and G.729, and by requesting network prioritization through the layer 3 differentiated services (DiffServ) scheme, as well as the layer 2 IEEE 802.1p/Q prioritization. Diffserv and 802.1p/Q are supported on voice packets coming to/from the gateway, all the way down to the endpoints such as IP telephones. Dynamic jitter buffers are also used.

QoS to endpoints

Users can set operating parameters to optimize the audio performance, or quality of service (QoS), on calls made over your IP network. These parameters include the audio codec, network priority through DiffServ capability, and the IEEE 802.1p/Q MAC-layer prioritization and segregation.

Default QoS parameters are downloaded to the IP telephone R1.5 and the IP softphone R3 when the values are not provided by the endpoint installer or the user. Certain options can be set locally by the endpoints or through the gatekeeper. The endpoints receive the parameters when the endpoints register, and once they are registered, whenever the administered values of the QoS parameters are modified.

Resource Reservation Protocol

Resource Reservation Protocol (RSVP) is a QoS signaling protocol. RSVP provides a means of specifying the requirements of IP packet flow, and determining if the intervening network can provide the resources to protect that flow through a process called “admission control.”

RSVP protection of VoIP audio streams on WANs and other links that are susceptible to congestion can safeguard the quality of VoIP calls already in progress.

- IP telephones or gateways request the network routers to reserve bandwidth.
- The routers act upon the request to allocate bandwidth according to the QoS request.
- When the bandwidth is reserved, the call is protected against other network traffic in a loaded or congested network, thereby ensuring good voice quality.

Administrators can now configure RSVP settings in Communication Manager. When the `RSVP enable` field in the **IP Network Region** screen is set to **y**, the `RSVP Reservation Parameters` field appears.

Sending and receiving faxes over IP

Starting with Communication Manager release 2.1, users can send and receive faxes over the voice over IP (VoIP) and modem over IP (MoIP) networks. The firmware that is resident on the TN2302AP circuit packs (Hardware Vintage 10 or later), the MM760 Media Module, the G700 Media Gateway, and the G350 Media Gateway, actually performs the processing necessary to allow proper handling of faxes over an IP network.

Modem over IP

The modem over IP (MoIP) feature allows for transport of data over a 64kbps unrestricted clear channel. Starting with Communication Manager release 3.0, when a clear channel data call is originated, the system communicates to the media processor or VoIP engine to allow a 64kpbs clear channel to be opened for transport.

Relay mode

In relay mode, the firmware detects fax tones and uses the appropriate modulation protocols (V.xx) to terminate or originate the fax so that the fax can be carried over the IP network. To reduce bandwidth over the IP network, the system encodes the modulated analog signal from the fax, and uses a relay coder/decoder. This process improves the reliability of transmission. Also, because the data packets for faxes in relay mode are sent almost exclusively in one direction, from the sending endpoint to the receiving endpoint, bandwidth use is reduced.

Relay mode works only if the receiving fax endpoint and the sending fax endpoint both communicate through Avaya Communication Manager media servers. This transport of fax signals occurs at a 9600 bps rate (though this rate may vary with the version of firmware). This mode may be used for fax calls to and from Communication Manager R2.0 systems.

Pass through mode

Alternatively, you can choose to have fax signals sent in “pass through” mode. Pass through mode means the fax signals are transported using G.711-like encoding and are delivered to the receiving fax endpoint as IP signals. This capability provides higher quality transmission in the circumstance where endpoints in the network are all synchronized to the same clock source. Pass through mode works only if the receiving fax endpoint and the sending fax endpoint both communicate through Avaya Communication Manager media servers.

The transport speed is up to the equivalent of circuit-switched calls and supports G3 and Super G3 fax rates, up to and including 33.6 kbps.

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.

You can assign packet redundancy in both pass through and relay mode, which means the media gateways use RFC 2198 packet redundancy to improve packet delivery and robustness of fax transport over the network.

Pass through mode uses more network bandwidth than relay mode. Redundancy increases bandwidth usage even more.

Encryption

You can encrypt fax pass through calls using either Avaya Encryption Algorithm (AEA) or Advanced Encryption Standard (AES). You can encrypt fax relay calls with AEA only. For more information about encryption, see Security, privacy, and safety on page 193.

T.38 faxes over the Internet

With Communication Manager, Release 2.1, users can send and receive faxes over the VoIP network using the T.38 standard for relay. The firmware resident on the TN2302AP circuit packs (Hardware Vintage 10 or later), the MM760 Media Module, the G700 Media Gateway, and the G350 Media Gateway actually performs the processing necessary to allow proper handling of faxes over an IP network. This transport of fax signals occurs at a 9600 bps rate.

The T.38 fax capability allows users to send faxes to and receive faxes from endpoints that are connected to non-Avaya systems. This capability is standards-based and uses IP trunks and H.323 signaling to allow communication with non-Avaya systems. Additionally, the T.38 fax capability uses the UDP protocol.

Note:

Fax endpoints served by two different Avaya media servers can also send T.38 faxes to each other if both systems are enabled for T.38 fax. In this case, the media servers also use IP trunks.

However, if the T.38 fax sending and receiving endpoints are on port networks or media gateways that are registered to the same media server, the gateways or port networks revert to Avaya fax relay mode. Avaya fax relay mode is more efficient that T.38 from a bandwidth perspective.

Both the sending and receiving systems must announce support of T.38 fax data applications during the H.245 capabilities exchange. Avaya systems announce support of T.38 fax if the capability is administered on the Codec Set screen for the region and a T.38-capable media processor was chosen for the voice channel. In addition, for a successful fax transmission, both systems should support the H.245 null capability exchange (shuffling) in order to avoid multiple IP hops in the connection.

Note:

The T.38 fax capability does not support TCP.

You can assign packet redundancy to T.38 standard faxes to improve packet delivery and robustness of fax transport over the network.

Pass through mode

You cannot send faxes in pass through mode with the T.38 standard.

Shuffling and hairpinning

Shuffling and hairpinning can improve traffic handling performance and improve voice quality by more efficiently using both Communication Manager switching fabric by allocating, when possible, available IP network resources.

“Shuffling” means rerouting the audio channel connecting two IP endpoints. After shuffling, the audio which previously was carried in a mixed connection of IP signaling and TDM bus signaling, goes directly through the LAN or WAN between the two IP endpoints. Shuffling also can mean reversing this process if an endpoint requests a resource to support a feature, such as conferencing that requires the TDM bus.

“Hairpinning” means rerouting the audio channel connecting two IP endpoints so that the bearer (audio) packets are routed through the TN2302AP IP Media Processor board in IP format, without having to go through the IP to TDM conversion or traverse the TDM bus.

NAT with shuffling

Communication Manager allows IP endpoints to shuffle if they are behind a Network Address Translation (NAT) device in an IP network.

Note:

Network Address Translation (NAT) is a method to address the shortage of IP V4 addresses by allowing globally register IP addresses to be reused by native networks. A NAT device translates between translated and native IP addresses.

Communication Manager supports IP direct calls (a call that has been shuffled) between two IP endpoints that are translated through a NAT device.

This enhancement works with static one-to-one NAT. It does not facilitate Port Address Translation (PAT), also known as Network Address Port Translation (NAPT). This enhancement does not work with many-to-one NAT.

TTY

People with hearing or speech disabilities often rely on a device known as a TTY in order to communicate on telephone systems. The term “TTY” is an abbreviation for Teletypewriter. The term “TDD” (Telecommunication Device for the Deaf) is also frequently used. The term TTY is generally preferred, however, because many people who use these devices are not deaf.

TTY devices typically resemble small laptop computers, except that there is a one- or two-line alphanumeric display in place of the computer screen.

Connection to the telephone network is generally through an acoustic coupler into which the user places the telephone's handset, or through an analog RJ-11 tip/ring connections.

Reliable transmission of TTY signals is supported by Communication Manager. This complies with the requirements and guidelines outlined in United States accessibility-related laws. Those laws include:

- Titles II, III, and IV of the Americans with Disabilities Act (ADA) of 1990.
- Sections 251 and 255 of the Telecommunications Act of 1996.
- Section 508 of the Workforce Investment Act of 1998.

Communication Manager TTY support is currently restricted to TTY devices that use the:

- US English standard TTY protocol, specified by ANSI/TIA/EIA 825 as: “A 45.45 Baud FSK modem.”
- UK English standard TTY protocol, Baudot 50.

Important characteristics of this standard are:

- TTYs are silent when not transmitting. Unlike fax machines and computer modems, TTYs have no “handshake” procedure at the start of a call, nor do they have a carrier tone during the call. This approach has the advantage of permitting TTY tones, DTMF, and voice to be intermixed on the same call.

  Note:

  A large percentage of people who use TTY devices intermix voice and typed TTY data on the same call. The most common usage is by people who are hard of hearing, but nevertheless able to speak clearly. These people often prefer to receive text on their TTY device and then speak in response. This process is referred to as Voice Carry Over (VCO).

- Operation is “half duplex.” TTY users must take turns transmitting and typically cannot interrupt each other. If two people try to type at the same time, their TTY devices might show no text at all or show text that is unrecognizable. Also, there is no automatic mechanism that lets TTY users know when a character they have correctly typed has been received incorrectly.
- Each TTY character consists of a sequence of seven individual tones. The first tone is always a “start tone” at 1800 Hz. This is followed by a series of five tones, at either 1400 or 1800 Hz, which specify the character. The final tone in the sequence is always a “stop tone” at 1400 Hz. The stop tone is a border that separates this character from the next.

The following types of systems support TTY communication:

- Analog telephones and trunks
- Digital telephones and trunks
- VoIP gateways
- Messaging systems
- Automated attendant systems
- IVR systems
- Wireless systems in which a TTY-compatible coder is used

As long as the user's TTY device supports the following, Communication Manager allows:

- Voice and TTY tones to be intermixed on the same call.
- DTMF and TTY (with or without voice) to be intermixed on the same call. This allows TTY users to access DTMF-based voice mail, auto-attendant, and IVR systems.
- The use of acoustically coupled and “direct connect” (RJ-11) TTY devices.

TTY over analog and digital trunks

Communication Manager supports TTY calls within a gateway or port network between two analog telephones. TTY calls from a gateway or port network over analog trunks or digital trunks is also supported.

TTY over Avaya IP trunks

Communication Manager supports calls over IP trunks, as well as Inter-Gateway Calls (IGC).

Note:

For this feature to work, both the sender (near end) and the receiver (far end) of a TTY call must each be connected to Avaya IP trunks. This feature does not work if either telephone is an IP telephone.

TTY relay mode

In relay mode, the system:

- detects TTY characters
- transports a representation of the characters over the IP network
- regenerates TTY characters/tones for delivery to the TTY device

This transport of TTY supports US English TTY (Baudot 45.45) and UK English TTY (Baudot 50). TTY uses RFC 2833 or RFC 2198 style packets to transport TTY characters.

Depending on the presence of TTY characters on a call, the transmission toggles between voice mode and TTY mode. The system uses up to 16 kbps of bandwidth when sending TTY characters, and normal bandwidth of the audio codec for voice mode. This mode may be used for TTY calls to and from Communication Manager R2.0 systems.

In relay mode, you can also assign packet redundancy. Packet redundancy means the media gateways send duplicated TTY packets to ensure and improve quality over the network.

TTY pass through mode

Alternatively, you can choose to have TTY signals sent in pass through mode. With pass through mode enabled, when the system detects TTY characters, the system uses G.711 encoding to transport the TTY signals end-to-end over the IP network. G.711 encoding pass through mode means the TTY signals are changed to digital format, and are delivered to the receiving endpoint after unencoding the digital signals.

Pass through mode provides higher quality transmission when endpoints in the network are all synchronized to the same clock source.

In pass through mode, you can also assign packet redundancy. Packet redundancy means the media gateways send duplicated TTY packets to ensure and improve quality over the network.

Pass through mode uses more network bandwidth than relay mode. Pass through TTY uses 87-110 kbps, depending on the packet size, whereas TTY relay uses, at most, the bandwidth of the configured audio codec. Redundancy increases bandwidth usage even more.

Variable length ping

Provides an enhancement to the ping command included in R7.1. This enhancement specifies a longer packet to be sent by ping and shows if a router or host has a problem fragmenting or integrating transferred packets.

Variable Length Subnet Mask

Variable Length Subnet Mask (VSLM) is a redefinition of the subnet mask, allowing for a more efficient allocation of IP addresses within a traditional classful block when administering IP address on Communication Manager.

QSIG

Basic

QSIG provides compliance to the International Standardization Organization (ISO) ISDN-PRI private-networking specifications. QSIG is defined by ISO as the worldwide standard for private networks. QSIG features are supported on BRI trunks.

QSIG is the generic name for a family of signaling protocols. The Q-reference point or interface is the logical point where signaling is passed between 2 peer entities in a private network. QSIG signaling can provide feature transparency in a single-vendor or multi-vendor environment.

QSIG provides call-related supplementary services. These are services **that go** beyond voice or data connectivity and number transport and display. Examples of supplementary services include name identification, call forwarding (diversion), and call transfer.

Call completion

Call completion utilizes the QSIG platform enhancement call independent signaling connections and is functionally equivalent to the Distributed Communications System (DCS) feature: auto-callback. The call completion feature includes a connection release method. The connection release method clears the Temporary Signaling Connection (TSC) after each phase of call-independent signaling and establishes a new TSC for each subsequent phase.

Call forwarding (diversion)

QSIG call forwarding (diversion) is based on the Communication Manager call forwarding feature. It extends the feature transparency aspects of call forwarding over a QSIG trunk:

- If QSIG call forwarding is activated, all calls are diverted immediately.
- If QSIG call forwarding with busy/do not answer is activated and a station is busy, a call is diverted immediately.
- If QSIG call forwarding with busy/do not answer is activated and a station is idle but the call is not answered, a call is diverted after a specified number of rings.

These features are activated either by dialing a Feature Access Code (FAC) or by pressing a button.

Call Independent Signaling Connections

Call Independent Signaling Connections (CISC) are used to pass QSIG supplementary service information that is independent of an active call between two QSIG compliant nodes. Implementation is based on the ISO standard for CISC. It is possible to determine the status of a QSIG TSC by using the “status signaling group” command on the SAT.

Call offer

This feature, on request from the calling-user (or on behalf of that user), enables a call to:

- Be offered to a busy called-user
- Wait for a busy called-user to accept the call when the necessary resources have become available

Call transfer

QSIG call transfer differs from the standard Communication Manager transfer feature in that additional call information is available for the connected parties after the transfer completes. However, the information is only sent for QSIG trunks. If one call is local to the transferring switch, that user receives the name of the party at the far end.

Called name ID

The QSIG called name feature presents the name of the called party on the display of the calling party while the call is ringing. It then reverts to “connected name” when answered.

Centralized Attendant Service

Provides you with the capability to have all your attendants in one location, serving users in multiple locations. QSIG CAS does not utilize separate Release Link Trunks (RLT). This feature will not restrict calls from going out over non-QSIG trunks; however, the full functionality of the QSIG CAS will not be available.

Attendant display of Class of Restriction

While on a call, the attendant can press a “COR display” button to see the class of restriction of the user. The attendant will not block the transfer of the restricted line to the user. This feature is used for informational purposes only.

Attendant return call

If a call that is transferred by the attendant goes unanswered for a specified period of time, the call is returned to the attendant. Preferably the call will transfer back to the attendant who transferred the call.

Priority queue

QSIG MSI will pass more information to the main PBX. This information enables calls coming in from a QSIG CAS branch to be placed in the appropriate place in the queue, as if the call originated on the main PBX.

RLT emulation through a PRI

ISDN QSIG trunks will route calls from the branch PBX to the main PBX. You no longer have to specify a dedicated RLT network. The QSIG path replacement takes care of the trunk optimization. You have the flexibility to route calls to the main PBX.

Communication Manager/Octel QSIG integration

Communication Manager enables integration of Octel messaging servers through QSIG. See Octel integration on page 112.

Leave Word Calling

See Leave Word Calling on page 110.

Manufacturer-Specific Information

QSIG handles non-standardized information that is specific to a particular PBX or network. This information is known as Manufacturer Specific Information (MSI). A manufacturer can define manufacturer-specific supplementary services operations after it has:

- Applied to a sponsoring and issuing organization (ECMA or European Computer Manufacturers Association in this case)
- Been assigned an organization identifier. This organization identifier is used as the root of the manufacturer-specific service-operation value.

All MSI operation values should be unique to that manufacturer.

Manufacturer-specific supplementary services can be created using specific operations encoded with the identifier of the manufacturer. Communication Manager supports non-QSIG applications that transport information across QSIG networks in MSI. Applications have the same functionality over QSIG networks that they have over non-QSIG networks. Applications that use MSI include Centralized Attendant Service, Transfer to AUDIX, Best Service Routing, and QSIG VALU.

Message Waiting Indication

The system indicates that a guest telephone has received one or more messages in their voice mailbox. An automatic message waiting lamp light at the telephone of the called party.

Name and number identification

Allows a **switch** to send and receive the calling number, calling name, connected number, and connected name. Additional parameters that control the display of the connected name and number are administered on the **Feature-Related System-Parameters** screen. QSIG Name and Number Identification displays up to 15 characters for the calling and connected name and up to 15 digits for the calling and connected number across ISDN-PRI interfaces.

Path replacement with path retention

With this feature, a call between switches in a private network can be replaced with new connections while the call is active. This feature is invoked when a call is transferred and improvements may be made in costs.

For example, after a call is transferred, the two parties on the transferred call can be connected directly and the unnecessary trunks are dropped off the call. The routing administered at the endpoints may allow for a more cost-effective connection.

Earlier versions of DEFINITY could not route a call over the original trunk when path replacement was turned on. Path Replacement features Path Retention, which allows Communication Manager to use the original trunk group path when the routing analysis performed by the switch shows the original trunk group to be the best route.

QSIG/DCS voice mail interworking

QSIG/DCS Voice Mail Interworking is an enhancement to the current QSIG feature. It integrates DCS and QSIG Centralized Voicemail through the DCS+/QSIG gateway. Switches labeled DCS+/QSIG integrate multi-vendor PBXs into a single voice messaging system. QSIG/DCS Voice Mail Interworking works on G3r, G3si, and G3csi. It provides network flexibility, DCS functionality without a dedicated T1.

Reroute after diversion to voice mail

Supports path optimization for calls that are diverted to a QSIG voice mail hunt group. That is, the switch moves the call to the shortest route between the caller and the voice mail system. For example, if user A on switch A calls user B on switch B and the call goes to a voice mail system attached to switch C, then the call is using up two trunks: A-B and B-C. If there is a trunk that directly connects switches A and C, this feature will drop the A-B and B-C connection and set up a new call from switch A to switch C, thus saving one trunk. The reroute happens automatically; the user never knows that the extra trunk was dropped.

Stand-alone path replacement

Path Replacement is the process of routing an established call over a more efficient path, after which the old call is torn down leaving those resources free. Path Replacement offers potential savings by routing calls more efficiently, saving resources and trunk usage.

Path replacement can exist as a stand-alone feature, or occur in the following additional cases:

- Call Forwarding by Forward Switching supplementary service, including the case where Call Diversion by Rerouting fails, and Call Forwarding is accomplished via forward switching
- Gateway scenarios where Communication Manager, serving as an incoming or outgoing gateway, invokes PR to optimize the path between the gateways
- Calls in queue/vector processing even though no true user is on the call yet
- QSIG Lookahead Interflow call, Best Service Route call, or adjunct route

Supplementary services and rerouting

The QSIG standard defines Supplementary Services as those service beyond voice or data connectivity and number transport and display. Examples include call forwarding, transfer and call hold.

VALU

Call coverage

This feature provides similar call coverage as DCS call coverage and Call Coverage Remote Off Net (C-CRON). The call will come back if covered over QSIG. The functionality will only be complete when all the switches are running under Communication Manager and using QSIG VALU. The covered-to party can still receive distinct alerting.

Call coverage and CAS

When a trunk has both CAS and VALU Call Coverage activated, the coverage display information is provided on calls that cover from a branch PBX to the main PBX. Path replacement will be attempted after coverage.

Distinctive alerting

Provides distinctive ringing, internal and external, to the remote called party when the call is routed over the QSIG network.

Uniform Dial Plan

A unique four- or five-digit number assigned to each station on the network. Uniform numbering gives each station a unique number (location code plus extension) that can be used at any location in the electronic tandem network to access that station, Communication Manager enhances the standard UDP with the unrestricted 5-digit Uniform Dial Plan, which allows up to five digits to be parsed for call routing.

Dial Plan Expansion

Communication Manager allows you to expand your dial plan to 6 or 7 digits (from 4-digit or 5-digit dial plans). This affects all extensions, including stations, data modules, agent login IDs, vectors, and so on.

This change increases the total number of extensions that can exist in any dial plan. It also allows Avaya servers to participate in networks that already use 6-digit or 7-digit dial plans - for example, a network of switches made by other vendors.

Administrators have the flexibility to administer dial plans between 3 and 7 digits in length, and Communication Manager supports mixed digit lengths in the same dial plan.

Customers upgrading to Communication Manager can choose to migrate to the 6-digit or 7-digit dial plan or not. Customers who choose not to migrate may convert their dial plans at a later date.

Distributed Communications System (DCS) protocol is limited to a dial plan of 3-5 digits. If your dial plan requires 6 or 7 digits, QSIG, which is the generic name for a family of signaling protocols, is required.

Multi-location dial plans

When a customer migrates from a multiple voice server QSIG/DCS network to a single voice server whose gateways are distributed across a data network, it may initially seem as if some dial plan functions are no longer available.

This feature preserves dial plan uniqueness for extensions and attendants that were provided in a multiple QSIG/DCS network, but were lost when customers migrated to a single distributed network. This feature provides dial plan capabilities similar to those provided before the migration, including:

- extension uniqueness
- announcement per location
- local attendant access
- local ARS code administration

A major reason to migrate customers from a multiple QSIG/DCS environment to a single S8700 network is to provide a greater set of features and help reduce costs. Migrating to a single network reduces the number of systems a customer has to maintain. That in turn lowers administration costs - one switch to administer instead of multiple switches, one dial plan to maintain, and so on. With a single distributed network solution, some features no longer work transparently across multiple locations.

For example, in a department store with many locations, each location might have had its own switch with a QSIG/DCS network. That way, the same extension could be used to represent a unique department in all stores. For example, extension 123 might be the luggage department in all stores. If the customer migrates to a single distributed network, this functionality is not available without this feature.

In addition, an S8700 solution does not assure that a call that is routed to an attendant would terminate at the local attendant. Let us use an example of a public school district that previously was networked with a switch at each school. If the school district migrates to an S8700 network, dialing the attendant access code at your school may not route your call to the local attendant.

Instead of having to dial a complete extension, the multi-location dial plan feature allows a user to dial a shorted version of the extension. For example, a customer can continue to dial 4567 instead of having to dial 123-4567. Communication Manager takes the location prefix and adds those digits 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.

Punctuation on station displays

On digital telephone displays, Communication Manager can display punctuation to make reading a 6-digit or 7-digit extension easier. The number of digits plus punctuation that can be displayed cannot exceed eight characters.

Punctuation marks that are allowed include:

- hyphen (for example, `xxx-xxxx`)
- period (for example, `xxx.xxxx`)
- space (for example, `xx xx xx`)

Formats for displaying numbers with punctuation are on the **Dial Plan Parameters** screen.

- the default 6-digit extension display format is `xx.xx.xx`
- the default 7-digit extension display format is `xxx-xxxx`

For more information on the **Dial Plan Parameters** screen, see the *Administrator Guide for Avaya Communication Manager*, 03-300509.

Extended trunk access

Used with Uniform Dial Plan, allows the system to send any unrecognized number (such as an extension not administered locally) to another system for analysis and routing. Such unrecognized numbers can be Facility Access Codes, Trunk Access Codes, or extensions that are not in the Uniform Dial Plan table. Non-Uniform Dial Plan numbers are administered on either the First Digit Table (on the **Dial Plan Record** screen) or the Second Digit Table. They are not administered on the Extended Trunk Access Call Screening Table. Extended Trunk Access helps you make full use of automatic routing and Uniform Dial Plan.

Extension Number Portability - When employees move within the network, they can retain their extension numbers. The ability to keep extension numbers, and even Electronic Tandem Network and Direct Inward Dialed numbers, when moving to other locations within the company eliminates missed calls and saves valuable time.

Chapter 15: Data interfaces

Administered connections

Automatically establishes an end-to-end connection between two access or data endpoints based on administered attributes. This feature provides capabilities such as alarm notification, including an administrable alarm type and threshold; automatic restoration of connections established over a Software-Defined Data Network; ISDN-PRI trunk group [service may be referred to as ISDN-PRI (AC/AE) Service]; scheduled as well as continuous connections; and administrable-retry interval for failed connection attempts.

Data call setup

Enables the setting up of data calls using a variety of methods, such as: keyboard dialing, telephone dialing, Hayes command dialing, permanent switched connections, administered connections, automatic calling unit interface, and Hot Line dialing. Data Call Setup is provided for both DCP and ISDN-BRI telephones.

Data hot line

Provides for automatic placement of a data call when the originator hangs up. Data Hot Line may be used for security purposes. This feature offers fast and accurate call placement to commonly called data endpoints. Data terminal users who constantly call the same number can use Data Hot Line to automatically place the call when they hang up the telephone.

Data modules

Data modules connect systems running Communication Manager with other communications equipment, changing protocol, connections, and timing as necessary.

Communication Manager supports the following types of data module:

- High Speed Links
- Data stands
- Modular-processor data module
- 7000-series data modules
- Modular-trunk data module
- Asynchronous Data Unit
- Asynchronous Data Module (for ISDN-Basic Rate Interface telephones)
- Terminal adapters

All of these data modules support industry standards and include options for setting the operating profile to match that of the data equipment.

Data privacy

Data Privacy protects analog data calls from being disturbed by any overriding or ringing features of the system. Data Privacy is activated when you dial an activation code at the beginning of the call.

Data restriction

Protects analog data calls from being disturbed by any overriding or ringing features of the system. It is administered at the system level to selected analog and multi-appearance telephones and trunk groups.

Default dialing

Provides data terminal users who dial a specific number the majority of the time a very simple method of dialing that number. This feature enhances Data Terminal (Keyboard) Dialing by allowing a data terminal user to place a data call to a pre-administered destination in several different ways, depending on the type of data module. Data Terminal Dialing and Alphanumeric Dialing are unaffected.

IP asynchronous links

IP asynchronous links enable Communication Manager to transfer existing asynchronous adjunct connectivity to an Ethernet (TCP/IP) environment. IP asynchronous links support switch server applications, as well as client applications. Systems running Communication Manager can connect to System Management applications such as the Avaya Visibility Suite over the LAN. Call Detail Recording (CDR) devices, Property Management System (PMS) and printers can be connected using asynchronous TCP/IP links.

IP asynchronous links:

- Reduce the cost of connecting to systems running Communication Manager for various adjuncts
- Allow for an open architecture to transport information and increases the speed at which data is transferred
- Allow customers to manage applications from on-site or remote locations
- Allow several system management applications to run on a single PC, thereby reducing hardware requirements
- Guarantee data delivery through a reliable session-layer protocol
- Support the existing serial hardware investment of a customer through use of Network Terminal Servers

Modem pooling

Enables switched connections between digital data endpoints (data modules) and analog data endpoints and acoustic coupled modems. Data transmission between a digital data endpoint and an analog endpoint requires a conversion since 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. The Modem Pooling feature provides a set of modems for such conversions.

Communication Manager modem pools are assigned into modem pool groups. A group can have up to 32 modems, called “members.” Communication Manager can have as many as 63 modem pool groups. See Figure 5: Modem pooling on page 174.

Figure 5: Modem pooling

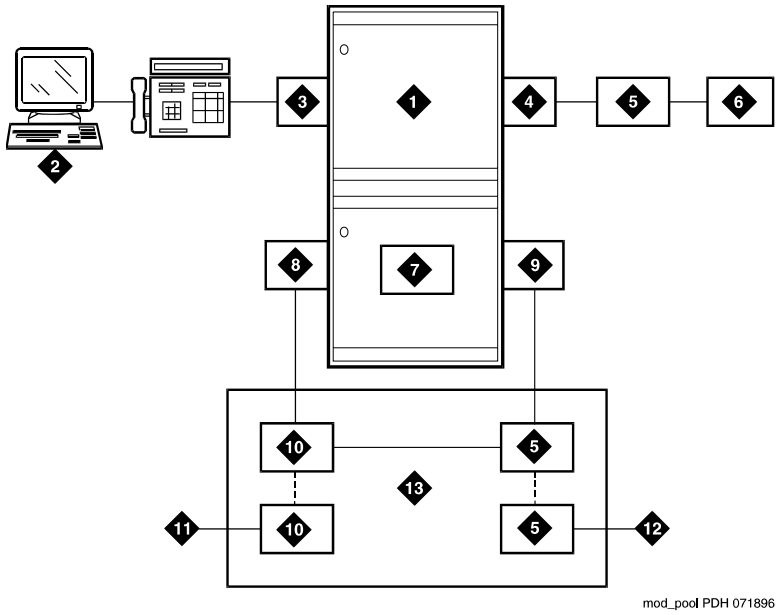

Figure notes:

1. System running Communication Manager
2. Asynchronous terminal
3. Digital port
4. Analog trunk
5. Modem
6. Remote application
7. Integrated pooled modem
8. Data line port
9. Analog port
10. 7400A
11. Digital communications protocol
12. Analog
13. EIA standard

Multimedia application server interface

The Multimedia Application Server Interface provides a link between Communication Manager and one or more Multimedia Communications eXchange nodes. A Multimedia Communications eXchange is a stand-alone multimedia call processor produced by Avaya. This link to Communication Manager enhances the capabilities of each Multimedia Communications eXchange system by enabling it to share some of the Communication Manager features. In particular, the interface provides the following advantages:

- Call Detail Recording (CDR). The capture of call detail records so you can analyze the call patterns and usage of multimedia calls just as Communication Manager administrators analyze normal calls.
- Automatic Alternate Routing/Automatic Route Selection (AAR/ARS). The intelligent selection of the most cost-effective routing for calls, based on available resources and your carrier preference. The system may select public trunks via DEFINITY Multimedia eXchange (MMCX).
- Voice Mail Integration. You can access your EMBEDDED AUDIX or INTUITY AUDIX voice messaging system from a Multimedia Communication eXchange (MMCX).

Multimedia calling

Multimedia calls are initiated with voice and video only. Once a call is established, one of the parties may initiate an associated data conference to include all of the parties on the call who are capable of supporting data. The data conference is controlled by an adjunct device called an Expansion Services Module (ESM).

Multimedia call early answer on vectors and stations

Early Answer is a feature applied to multimedia calls in conjunction with conversion to voice. Early Answer:

- Answers the data call
- Establishes the multimedia protocol prior to completion of a converted call
- Ensures that a voice path to/from the originator is available when the (voice) call is answered

For an incoming call, Early Answer answers the dynamic service-link calls when the destination endpoint answers, unless Early Answer is specified during routing or termination processing.

Note:

The “destination voice endpoint” might be an outgoing voice trunk if the destination voice station is forwarded or covered off-premises.

Multimedia Call Handling

Multimedia Call Handling (MMCH) enables you to control voice, video, and data transmissions using your telephone set. The feature buttons on a multi-function telephone enable you to conduct video conferences, and forward, cover, hold, or park multimedia calls much as you would a standard voice call. You can also share PC applications so that you and colleagues can collaborate while working from remote sites. See Figure 6: Multimedia Call Handling (MMCH) on page 176.

Figure 6: Multimedia Call Handling (MMCH)

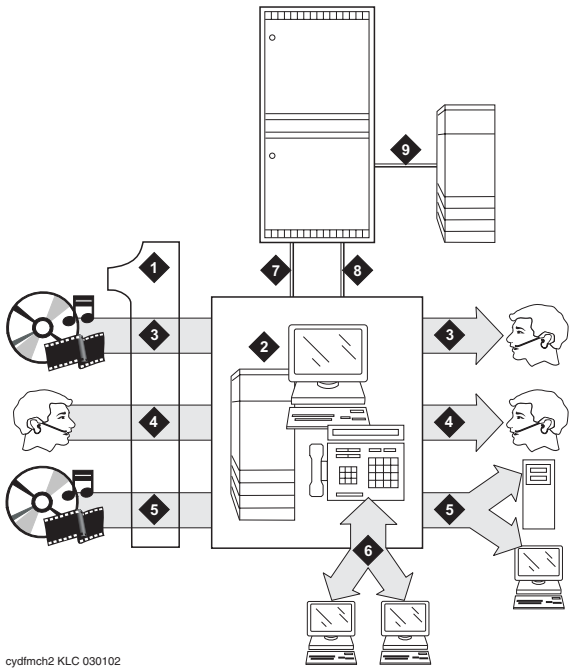

Figure notes:

1. One number access
2. Multimedia call complex
3. Multimedia to voice conversion
4. Standard voice call handling
5. Call redirection
6. Multimedia conferencing
7. BRI data connection
8. DCP voice connection
9. ESM data collaboration

Multimedia call redirection to multimedia endpoint

A dual port multimedia station may be a destination of call redirection features such as call coverage, forwarding, and station hunting. The station can receive and accept full multimedia calls or data calls converted to multimedia.

Multimedia data conferencing (T.120) through ESM

The data conference is controlled by an adjunct device called an Expansion Services Module (ESM). The Expansion Services Module is used to terminate T.120 protocols [including Generalized Conference Call (GCC), a protocol standard for data conference control] and provide data conference control and data distribution. The MultiMedia Interface circuit pack, TN787, is used to rate adapt T.120 data to/from the ESM.

Multimedia hold, conference, transfer, and drop

Station users have the ability to activate hold, conference, transfer, or drop on multimedia calls. Multimedia endpoints and voice-only stations may participate in the same conference.

Multimedia multiple-port networks

Communication Manager supports the equivalent of 580 Basic mode complexes operating at 6CCS traffic level. All enhanced mode complexes operate with soft-mode service links since the use of hard-mode service links reduces capacities. G3si limits are 1/3 to 1/2 of the G3r limits, depending on memory limitations and port network limitations.

Pass advice of charge information to world class BRI endpoints

Provides Advice of Charge (AOC) information to World Class BRI (WCBRI) endpoints. On a call using a WCBRI endpoint, AOC information will be displayed on the endpoint after the call has completed and the far end has hung up.

Chapter 16: Call routing

Alternate facility restriction levels

Allows Communication Manager to adjust facility restriction levels or authorization codes for lines or trunks. Each line or trunk is normally assigned a facility restriction level. With this feature, Alternate Facility Restriction Levels are also assigned. Attendants can change to the alternates, thus changing access to lines and trunks. You might want to use this feature to disable most long-distance calling at night, for example, to prevent unauthorized staff from making long-distance calls.

CAUTION:

This feature may change the AAR and ARS routing preferences. Using it on tandem and tie-trunk applications affects entire networks. Calls that are part of a cross-country private network may be blocked.

Automatic routing features

Communication Manager provides a variety of automatic routing features for public and private networks. Automatic Alternate Routing (AAR) and Automatic Route Selection (ARS) are the foundation for these automatic-routing features. They route calls based on the preferred (normally the least expensive) route available at the time the call is placed. Generally, AAR routes calls over a private network and ARS routes calls using the public network numbering plan. However, both AAR and ARS support public and private networks. You can use the other features listed in this section when you use AAR and ARS.

Automatic Alternate Routing

Automatic Alternate Routing (AAR) allows private network calls to originate and terminate at one or many locations without accessing the public network. When you dial an access code and telephone number, **AAR** selects the most desirable route for the call and performs digit conversion as necessary. If the first choice route is unavailable, another route is chosen automatically.

The numbers you call using AAR are normally private-network numbers. However, you can call a public-network number, a service code, an **international** number, operator access code, or an operator-assisted dialing number. With **AAR** and Subnet Trunking, you have a convenient way to place **international** calls to frequently-called foreign cities. Such calls route as far as possible over the private network, and then access the public network. This saves toll charges and allows you to use your private network as much as possible.

Automatic Route Selection

Automatic Route Selection (ARS) selects carriers automatically and routes calls inexpensively over the public network. When there are one or more long-distance carriers and Wide Area Telecommunications Service (**WATS)** provided, Communication Manager selects the most preferred route for the call. Long-distance carrier-code dialing is not required on routes selected by the system. You assign long-distance carrier-codes and Communication Manager translates them. The system inserts codes as needed to guarantee automatic-carrier selection. **ARS** can route calls to a variety of types-of-numbers and access a variety of types of trunk groups.

ARS/AAR dialing without FAC

The Automatic Route Selection (ARS) version of this feature allows users to place calls by dialing the full public-network numbers without first having to dial a Feature Access Code (FAC), such as the number “9” to access an outside line. The system recognizes the call as an ARS call and uses the ARS digit analysis and digit conversion tables to manipulate the digits to route the call.

The Automatic Alternate Routing (AAR) version of this feature is similar except that the call is routed as an AAR call and therefore uses the AAR digit analysis and digit conversion tables.

AAR/ARS overlap sending

Communication Manager supports overlap sending for AAR and ARS calls that are routed over ISDN-PRI trunk groups. ISDN-PRI call-address information is sent one digit at a time instead of in one block. In countries with complex public-network numbering plans, this allows for a significant decrease in call setup time. When overlap receiving is enabled, this is especially significant for tandem calls.

AAR/ARS partitioning

Allows A**AR** and **ARS** to be partitioned into 8 user groups within a single system and provides individual routing treatment for each of these user groups.

User groups share the same Partition Group Number, which indicates the choice of routing tables that are used on a particular call. Each Class of Restriction (COR) is assigned a specific **Partition Group Number** or Time of Day specification. Different **classes of restriction** may be assigned the same **Partition Group Number**.

Generalized route selection

Provides voice and data call-routing capabilities. You use it to select not only the least-cost routing, but also optimal routing over the appropriate facilities. **It enhances AAR and ARS by** providing additional parameters in the routing decision and maximizing the chance of using the right facility to route the call. Also, if an endpoint incompatibility exists, it provides a conversion resource (such as a modem from a modem pool) to attempt to match the right facility with the right endpoint.

Look-ahead routing

Provides an efficient way to use trunking facilities. It allows you to continue to try to reroute an outgoing ISDN-PRI call that is not completing. When Communication Manager receives a cause value that indicates congestion, Look-Ahead Routing tells the system what to do next. For each routing preference, you can indicate if the next routing-preference should be attempted or if the current routing-preference should be attempted again.

Node number routing

Allows you to specify the route pattern associated with each node in a private network. It is a required capability for Extension Number Portability and is used in conjunction with Automatic Route Selection, AAR and ARS Partitioning, Private Networking, and Uniform Dial Plan. **Uniform Dial Plan** extensions can be routed to a specified node using its associated pattern. Node Number Routing allows a **Uniform Dial Plan** route pattern based on node numbers or on location codes. On the **AAR** and **ARS** Digit Analysis Tables, you also can specify a Node Number instead of a Route Pattern.

Time of day routing

Provides the most economical routing of **ARS** and **AAR** calls. This routing is based on the time of day and day of the week that each call is made. Up to 8 TOD routing plans may be administered, each scheduled to change up to 6 times a day for each day in the week. This allows you to take advantage of lower calling rates during specific times of the day and week. In addition, companies with locations in different time zones can use different locations that have lower rates at different times of the day or week. This feature is also used to change patterns during the times an office is closed in order to reduce or eliminate unauthorized calls.

Multiple location support

Multiple Location Support enables local user time, local ARS Public Analysis Tables for local trunking, automatic Daylight Savings Time, and enhances shared resource algorithms (touch tone receivers) when Remote Expansion Port Networks (EPNs), ATM Port Networks, and Avaya Media Gateways are remoted off of a central server at a different location.

Multiple location support for network regions

See Multiple location support for network regions on page 151.

Traveling class marks

Traveling Class Marks are a mechanism for passing the facility restriction level of a caller from one Electronic Tandem Network switch to another. Traveling Class Marks allow privilege checking to be passed across switches through the Electronic Tandem Network.

Answer detection

For purposes of Call-Detail Recording (CDR), it is important to know when the called party answers a call. Communication Manager provides three ways to determine whether the called party has answered an outgoing call.

Answer supervision by time-out

You set a timer for each trunk group. If the caller is off-hook when the timer expires, Communication Manager assumes that the call has been answered. This is the least accurate method. Calls that are shorter than the timer duration do not generate call records, and calls that ring for a long time produce call records whether they are answered or not.

Call-classifier board

A call-classifier board detects tones and voice-frequency signals on the line and determines whether a call has been answered.

Network answer supervision

The Central Office (CO) sends back a signal to indicate that the far end has answered. If a call has traveled over a private network before reaching the CO, the signal is transmitted back over the private network to the originating system. This method is extremely accurate, but is not available in the United States over CO, FX, or WATS trunks.

Chapter 17: Reliability and survivability

Alternate gatekeeper

The alternate gatekeeper enhancement can provide survivability between Avaya Communication Manager and IP communications devices such as IP Telephones and IP Softphone. This is accomplished by providing alternate gatekeepers (CLAN) in the event of network or gatekeeper failure and by load balancing endpoint traffic among multiple gatekeepers. It is important to recognize that calls will drop during that interval while the communication is re-established to the switch.

Auto fallback to primary for H.248 gateways

This feature automatically returns a fragmented network, where a number of H.248 media gateways are being serviced by one or more Local Survivable Processors (LSP), to the primary media server. This feature is targeted to H.248 media gateways only.

This feature allows the administrator to define any of the following rules for migration:

- Whether or not the media gateways, serviced by LSPs, should automatically migrate to the primary media gateway.
- Whether or not the media gateway should migrate immediately when possible, regardless of active call count.
- Whether or not the media gateway should only migrate if the active call count is 0.
- Whether or not the media gateway should only be allowed to migrate within a window of opportunity, by providing day of the week and time intervals per day. This option does not take call count into consideration.
- Whether or not the media gateway should be migrated within a window of opportunity by providing day of the week and time of day, or immediately if the call count reaches 0. Both rules are active at the same time.

Internally, the primary call controller gives priority to registration requests from those media gateways that are currently not being serviced by an LSP. This priority is not administrable.

Connection preserving failover/failback for H.248 media gateways

The Connection Preserving Migration (CPM) feature preserves existing bearer (voice) connections while an H.248 media gateway migrates from one Communication Manager server to another. Migration might be caused by a network or server failure.

Only stable calls are preserved. Call that are not preserved are:

- Unstable calls. An unstable is any call where the call talk path between parties has not been established, or is not currently established. Some examples are:
  - Calls with dial tone
  - Calls in dialing stage
  - Calls in ringing stage
  - Calls listening to announcements
  - Calls listening to music
  - Calls on hold (soft, hard)
  - Calls in ACD queues
  - Calls in vector processing
- IP trunks, both SIP and H.323
- ISDN-BRI telephones
- ISDN-BRI trunks

Users on connection-preserved calls cannot use such features as Hold, Conference, or Transfer.

Connection preserving upgrades for duplex servers

The connection preserving upgrades for duplex servers feature provides connection preservation on upgrades of duplex servers for:

- connections involving IP telephones
- connections involving TDM connections on port networks
- connections on H.248 gateways
- IP connections between port networks and media gateways

Stable calls are preserved. Unstable calls are dropped.

Enterprise Survivable Servers

The Enterprise Survivable Server (ESS) provide survivability by allowing backup servers to be placed in various locations in the customer network. The backup servers supply service to port networks in the case where the S8500-series media server, or the S8700-series media server pair fails, or connectivity to the main server or server pair is lost.

In an ESS environment, there can only be one main server, either one S8500-series media server, or one pair of S8700-series media servers. If the main server is an S8500-series media server, all ESSs in the configuration must also be S8500-series media servers. During normal operation, the main server communicates with and controls all the port networks. The main server contains a license file that identifies the server as the main server and activates the ESS functionality.

IP bearer duplication using the TN2602AP circuit pack

The TN2602AP IP Media Resource 320 circuit pack provides high-capacity voice over Internet protocol (VoIP) audio access to the switch for local stations and outside trunks. The IP Media Resource 320 provides audio processing for the following types of calls:

- TDM-to-IP
- IP-to-TDM
- IP-to-IP

The TN2602AP IP Media Resource 320 circuit pack has two capacity options, both of which are determined by the license file installed on Communication Manager:

- 320 voice channels, considered the standard IP Media Resource 320
- 80 voice channels, considered the low-density IP Media Resource 320

Only two TN2602AP circuit packs are allowed per port network.

Note:

The TN2602AP IP Media Resource 320 is not supported in CMC1 and G600 Media Gateways. For more information about the TN2602AP circuit pack, see the *Hardware Description and Reference for Avaya Communication Manager*, 555-245-207.

Load balancing

Up to two TN2602AP circuit packs may be installed in a single port network for load balancing. The TN2602AP circuit pack is also compatible with and can share load balancing with the TN2302 and TN802B IP Media Processor circuit packs. Actual capacity may be affected by a variety of factors, including the codec used for a call and fax support.

Bearer signal duplication

Two TN2602AP circuit packs may be installed in a single port network for bearer signal duplication. In this configuration, one TN2602AP is an active IP media processor and one is a standby IP media processor. If the active media processor, or connections to it, fail, active connections failover to the standby media processor and remain active. This duplication prevents active calls in progress from being dropped in case of failure.

For bearer duplication, both TN2602AP circuit packs must be Hardware Version 2, and must have firmware version 211 or higher.

Note:

The 4606, 4612, and 4624 telephones do not support the bearer duplication feature of the TN2602AP circuit pack. If these telephones are used while an interchange from active to standby media processor is in process, calls might be dropped.

Important:

If you change from load balanced to duplicated TN2602s, existing calls retain the real IP address on the TN2602AP circuit pack. New calls are associated with the virtual IP address of the TN2602AP circuit pack. If an interchange occurs during this time, existing calls that are associated with the real IP address will drop.

Reduced channels with duplicated TN2602AP circuit packs

If a pair of TN2602AP circuit packs, previously used for load balancing, are re-administered to be used for bearer duplication, only the voice channels of the active circuit pack can be used. For example,

- If you have two TN2602 AP circuit packs in a load balancing configuration, each with 80 voice channels, and you re-administer the circuit packs to be in bearer duplication mode, you have 80, not 160, channels available.
- If you have two TN2602 AP circuit packs in a load balancing configuration, each with 320 voice channels, and you re-administer the circuit packs to be in bearer duplication mode, you will have 320, not the maximum 484, channels available.

- When two TN2602AP circuit packs, each with 320 voice channels, are used for load balancing within a port network, the total number of voice channels available is 484, not 640. The reason is that 484 is the maximum number of time slots available for connections within a port network.

Local Survivable Processor

A Local Survivable Processor (LSP) is an Internal Call Controller (ICC) with an integral G700 Media Gateway, in which the ICC is administered to behave as a spare processor rather than as the main processor. The standby Avaya S8700 Media Server runs in duplex mode with the main server ready to take control in the event of a outage with no loss of communication.

An LSP is a configuration used to provide redundancy of the Avaya call processing application. Usually, a media module serves as the ICC for the system, but it can also serve as a redundant processor for call processing. In the LSP configuration, the processor serves as an alternate controller/gatekeeper for IP entities, such as IP telephones and media gateways. These IP entities use the LSP when they lose connectivity to their primary controller.

In the event that the communication link is broken between the remote Avaya G700 Media Gateway and the primary call controller (either an Avaya S8300 Server or an Avaya S8700 Server), the LSP provides service for the Avaya IP telephones and Avaya G700 Media Gateways that were controlled by the primary call controller.

How the Avaya G700 Gateways and IP endpoints change control from the primary to the LSP is driven by the endpoints themselves, using a list of call controllers. During initialization, each IP endpoint and Avaya G700 Gateway receives a list of call controllers. The IP endpoints ask each call controller in the list for service until one responds with a positive reply. If the link to that call controller fails at some later time, the endpoint will try to receive service from the other call controllers in the list, including the LSP.

The LSP provides service to all Avaya G700 Gateways and IP endpoints that register with it. When the primary call controller is prepared to provide service, the LSP is reset. This informs the IP endpoints to try their call controller list again, and returns to the primary call controller for service.

The LSP provides redundancy in a variety of configurations, and can be located anywhere in a network of Avaya G700 Gateways.

For LSP capacities, refer to the capacities table. See Capacities on page 32 for instructions how to view the capacities table.

Automatic upgrade tool of server/LSP software and license

This feature adds the following functionality to the Web page upgrade tool:

- Distribution of license files from the server on which the upgrade tool is running to the LSPs that need them
- Display of SID/MID on the query results
- Support for the G350 gateways as an upgrade target
- Upgrade of the standby server
- Upgrade of the sever on which the upgrade tool is running
- Upgrade of ESS servers
- Support for FTP as well as TFTP as a gateway upgrade protocol
- Support for administration of the number of simultaneous FTP/TFTP sessions

This feature is implemented only on Linux servers.

Multiple network regions per CLAN

See Multiple network regions per CLAN on page 151.

Power failure transfer

Provides service to and from the local telephone company central office (CO), including wide area telecommunications system, during a power failure. This allows you to make or answer important or emergency calls during a power failure. This feature is also called emergency transfer.

Survivable Remote EPN

The Survivable Remote Expansion Port Network (SREPN) allows a DEFINITY ECS (R6r or later) EPN to provide service to the customer when the link to the main processor fails or is severed or when the processor or CSS fails. When the links to the system are restored and stable, the logic switch is manually reset and the EPN is reconnected to the links from the switch. There are both command and manual resets. The resets can be done remotely at the SAT or manually at the equipment.

The SREPN must be administered separately (not as a duplicated PPN) to function in a disaster recovery scenario. It does not function as a survivable remote EPN without the administration (stations, trunks, features) to support its operation.

Note:

SREPN is not compatible with ATM port network connectivity (ATM-PNC). If that is the case, see WAN Spare Processor on page 124.

WAN spare processor

See WAN Spare Processor on page 124.

Chapter 18: Security, privacy, and safety

System administrator

Access security gateway

Access security gateway is an authentication interface used to secure the system administration and maintenance ports and/or logins on the system. Access security gateway employs a challenge/response protocol to confirm the validity of a user and reduce the opportunity for unauthorized access.

Successful authentication is accomplished when the feature communicates with a compatible key. The challenge/response negotiation is initiated once an RS-232 session is established and a valid system login ID has been supplied by a user. The authentication transaction consists of a challenge, issued by the system and based on the login ID supplied by the user, followed by receipt of the expected response, which is supplied by the user.

Encryption algorithm for bearer channels

Communication Manager supports the Advanced Encryption Standard (AES) format of signal encryption for IP telephony. This encryption algorithm is in addition to the Avaya proprietary encryption protocol, the Avaya Encryption Algorithm (AEA).

AES encryption is a cryptographic algorithm developed by the U.S. Government to protect unclassified information. Communication Manager uses AES with 128 bit keys in counter mode (AES-128-CTR).

Administration is supported to select a combination of no encryption, AEA encryption, and/or AES encryption on a per codec set basis.

Alternate facility restriction levels

This feature allows Communication Manager to adjust facility restriction levels or authorization codes for lines or trunks. Each line or trunk is normally assigned a facility restriction level. With this feature, alternate facility restriction levels are also assigned. Attendants can change to the alternates, thus changing access to lines and trunks.

You might want to use this feature to disable most long-distance calling at night, for example, to prevent unauthorized staff from making long-distance calls.

CAUTION:

This feature may change the AAR and ARS routing preferences. Using it on tandem and tie-trunk applications affects entire networks. Calls that are part of a cross-country private network may be blocked.

Alternate operations support system alarm number

This feature allows you to establish a second number for Communication Manager to call when an alarm event occurs. This feature is useful for alerting a second support organization, such as INADS or OneVision.

Privacy - attendant lockout

See Attendant lockout - privacy on page 43.

Authorization codes - 13 digits

Authorization codes extend calling-privilege control and enhance security for remote-access callers. Authorization codes can be up to 13 digits in length.

Avaya site administration authorization codes may be used to:

- Override facility restriction levels assigned to originating stations or trunks
- Restrict individual incoming tie trunks and remote-access trunks from accessing outgoing trunks
- Track CDR calls for cost-allocation purposes
- Provide additional security control

Call restrictions

By dialing an access code, administrators and attendants have the ability to restrict users from making or receiving certain types of calls. There are five restrictions:

- Outward. User cannot place external calls.
- Station-to-station. User cannot place or receive internal calls.
- Termination. User cannot receive any calls (except priority calls).
- Toll. User cannot place toll calls but can place local calls.
- Total. User can neither place nor receive any calls.

Class of Restriction

Defines many different classes of call origination and termination privileges. Communication Manager may have no restrictions, only a single COR, or may have as many classes of restrictions as necessary to effect the desired restrictions. Many different types of classes of restriction can be assigned to many types of facilities on the switch. For example, you can use a calling-party COR to prevent callers from accessing the public network.

Block collect call

See Block collect call on page 96.

Customer-provided equipment alarm

Provides you with an indication that a system alarm has occurred and that the system has attempted to contact a service organization. A device that you provide, such a lamp or a bell, is used to indicate the alarm situation. You can administer the level of alarm about which you want to be notified.

Data privacy

See Data privacy on page 172.

Data restriction

See Data restriction on page 172.

Facility restriction levels and traveling class marks

Allows certain calls to specific users, while denying the same calls to other users. For example, certain users may be allowed to use Central Office (CO) trunks to other corporate locations while other users may be restricted to less expensive private-network lines. **You can administer up to eight** levels of restriction for users of AAR and ARS.

H.248 link encryption

To provide privacy for media streams carried over IP networks, the H.248 signaling channel between the media server (media gateway controller) and the media gateways is encrypted. This signaling channel is used to distribute the media session keys to the media gateways, and may carry user-dialed authorization codes and passwords.

This feature protects our customer investments by encrypting the signaling channel between the H.248 gateway and server. This feature also protects the media encryption key, PINs, and account codes between the media gateway and the media gateway controller.

Encryption of the H.248 link to any given media gateway may be enabled or disabled through the **Media Gateway** screen. However, the encryption protocol cannot be disabled.

Malicious call trace

Allows you to trace malicious calls. You define a group of terminal users who can notify others in the group when they receive a malicious call. These users can then retrieve information related to the call. Using this information, you can identify the malicious call source or provide information to personnel at an adjacent system to complete the trace. It also allows you to record the malicious call, as well as trace a malicious call over ETSI PRI.

Malicious call trace logging

Malicious call trace logging allows a PC to receive information from Communication Manager to log malicious calls.

Media encryption

Media Encryption is the encryption of the audio (voice) portion of a Voice Over IP (VoIP) call. Media Encryption can be used to provide enhanced privacy for VoIP communications that involve exchange of sensitive information. Media Encryption is provided between Avaya media gateways and media servers.

Digitally encrypting the audio (voice) portion of a VoIP call can reduce the risk of electronic eavesdropping. IP packet monitors, sometimes called sniffers, are to VoIP calls what wiretaps are to circuit-switched (TDM) calls. One exception is that an IP packet monitor can watch for and capture unencrypted IP packets, and can play back the conversation in real-time or store it for later playback.

Communication Manager encrypts IP packets before they traverse the IP network. An encrypted conversation sounds like white noise or static when played through an IP monitor. End users do not know that a call is encrypted because there are:

- No visual or audible indicators to indicate that the call is encrypted.
- No appreciable voice quality differences between encrypted calls and non-encrypted calls.

SECURITY ALERT:

Be sure that you understand these important media encryption limitations:

- Any call that involves a circuit-switched (TDM) endpoint, such as a DCP or analog telephone, is vulnerable to conventional wire-tapping techniques.
- Any call that involves an IP endpoint or gateway that does not support encryption can be a potential target for IP monitoring. A common example of this is are IP trunks to 3rd-party vendor switches.
- Any party that is not encrypting an IP conference call exposes all parties on the IP call between the unencrypted party and its supporting media processor to monitoring, even though the other IP links are encrypting.

For a list of the supported hardware, software, and firmware requirements for Media Encryption, and a list of the equipment that is not supported, see the *Hardware Description and Reference for Avaya Communication Manager*, 555-245-207.

License file requirements

Media Encryption does not work unless the server has a valid License File with Media Encryption enabled. To determine whether Media Encryption is enabled in the current License File:

1. Type **`display system-parameters customer-options`** a. Press **Enter**.

   The system displays the **Optional Features** screen.

2. Press **Next** until you see the **Media Encryption Over IP** field (Figure 7: Optional Features screen on page 198).

3. Ensure that the `Media Encryption Over IP` field is set to **y**.

Figure 7: Optional Features screen

```
display system-parameters customer-options                      Page 3 of 10
                                OPTIONAL FEATURES

   Emergency Access to Attendant? y                      ISDN Feature Plus? y
            Enable 'dadmin' Login? y            ISDN Network Call Redirection? y
            Enhanced Conferencing? n                        ISDN-BRI Trunks? y
                   Enhanced EC500? y                              ISDN-PRI? y
          Extended Cvg/Fwd Admin? y                    Local Spare Processor? n
     External Device Alarm Admin? y                     Malicious Call Trace? y
  Five Port Networks Max Per MCC? y              Media Encryption Over IP? y
                Flexible Billing? y     Mode Code for Centralized Voice Mail? y
   Forced Entry of Account Codes? y
      Global Call Classification? y                  Multifrequency Signaling? y
              Hospitality (Basic)? y Multimedia Appl. Server Interface (MASI)? n
 Hospitality (G3V3 Enhancements)? y         Multimedia Call Handling (Basic)? n
                       IP Trunks? y      Multimedia Call Handling (Enhanced)? n
                                                       Multiple Locations? y
            IP Attendant Consoles? y            Personal Station Access (PSA)? y
                     IP Stations? y
```

Restriction - controlled

Allows an attendant or telephone user, with console permission, to activate and deactivate for an individual telephone or a group of telephones, the following restrictions:

- outward
- total
- station-to-station
- termination restrictions

Secure shell and secure FTP

The Telnet protocol allows remote access to a network device console that is based on login and password authentication. Beginning with Communication Manager release 3.0, Secure Shell (SSH) provides this capability over an encrypted channel. Similarly, Secure FTP (SFTP) is an encrypted version of the FTP protocol that allows remote file transfers. SSH/SFTP provides a secure alternative for file transfer of firmware download files and voice announcements, as well as secure remote server access.

The **`enable filexfer`** command enables SSH and SFTP for both the TN799DP Control LAN (CLAN) circuit pack, and the TN2501AP Voice Announcement over LAN (VAL) circuit pack.

The Telnet, FTP, SSH, and SFTP enabling capabilities on the TN2312A/BP IP Server Interface (IPSI) circuit pack continue to be handled through the Communication Manager web interface and the Communication Manager Linux bash shell.

Security of IP telephone config files

This feature supports the inclusion of a digital certificate and the use of TLS to allow an IP telephone to authenticate the server for the download of configuration files. This enables IP telephones to ensure that configuration parameters come only from an authenticated source. Configuration files that are delivered through this mechanism can deliver message digest values for the authentication of software code files delivered through a non-secure connection.

Security of IP telephone registration/H.323 signaling channel

Note:

Please check with your Avaya Sales Representative or your Avaya Authorized Business Partner for availability of this feature.

This feature provides a secure mechanism for an H.323 endpoint and a Communication Manager gatekeeper to mutually authenticate themselves and the contents of the messages that they exchange during IP registration, admission, and status (RAS). This authentication is based on the simple 3-to-8 digit PIN administered for each extension.

Execution of Encrypted Key Exchange (EKE) procedures during RAS results in the negotiation of a strong secret that is shared between the endpoint and the gatekeeper. This strong secret is used to derive a set of secrets which are used to digitally sign all RAS and call signaling messages, and to encrypt selected elements of call signaling messages, media session keys and CCMS messages. If one or the other parties does not possess the correct PIN, the computed shared secrets will, in fact, be different. Message authentication fails, and the parties refuse to communicate.

In summary, these procedures permit:

- The endpoint and the gatekeeper to authenticate each other;
- The endpoint and the gatekeeper to sign/authenticate each message sent;
- Privacy for selected elements of call signaling, including media session encryption keys and dialed digits.
- Security of the endpoint/gatekeeper communication even if an observer obtains the user's PIN.
- Security of past or future communications even if one session is penetrated by an attacker with knowledge of the PIN. (This is known as “perfect forward secrecy”.)
- Reuse of the negotiated strong secret (identified by a unique session ID) to secure new signaling links between parties for re-registration or trunking.

Security Violation Notification

Security Violation Notification (SVN) allows you to set security-related parameters and to receive notification when the limits that you have established are violated. You can run reports related to both valid and invalid access attempts. You can also disable a login ID or remote access authorization that is associated with a security violation.

Signaling encryption for SIP trunks

Signaling encryption for SIP trunks protects customer investments by encrypting the voice channel over SIP trunks.

Station security codes

To provide additional security around the customer options the “init” login has been provided with additional security for the purpose of establishing an authentication procedure for attempts to remotely log into the system.

Tripwire security

Tripwire is a security program provided on all Linux-based media servers. The list of files that Tripwire monitors needs to be determined during design when all administration and configuration files have been identified.

If there are any detected security violations, Tripwire reports its findings through the security log. These events generate an alarm.

Note:

Tripwire normally reports security violations through e-mail. However, by reporting events to the security log, security violations can be immediately acted upon.

End user

Backup alerting

Notifies backup attendants that the primary attendant cannot pick up a call. It provides both audible and visual alerting to backup stations when the attendant queue reaches its queue warning level. When the queue drops below the queue warning level, alerting stops. Audible alerting also occurs when the attendant console is in night mode, regardless of the attendant queue size.

Barrier codes

A barrier code is a security code that is used with remote access to prevent unauthorized access to your system. To increase your system security, use a 7-digit barrier code with remote access barrier code aging. A barrier code automatically expires if an expiration date or number of accesses has exceeded the limits you set. If both a time interval and access limits are administered for a barrier code, the barrier code expires when one of the conditions is satisfied.

Note:

Barrier codes are not tracked by call detail recording (CDR). Barrier codes are incoming access codes, whereas, authorization codes are primarily outgoing access codes.

Calling/Connected Party Number restriction

Per call CPN restriction

Users may indicate calling number privacy information. For ISDN calls, the CPN presentation indicator is encoded accordingly. For non-ISDN calls going to a public network that supports the CPN restriction feature, the network specific feature activation code gets passed to the network for interpretation and activation of the desired feature.

If per call CPN restriction is activated for an outgoing call, it will override any per line CPN restriction administration for the calling station, and will override any ISDN trunk group administration for sending calling number.

Per line CPN restriction

Users may block the calling party number when originating calls. For ISDN calls, the CPN presentation indicator is encoded accordingly. For non-ISDN calls, going to a public network that supports the CPN restriction feature, the network specific feature activation code gets passed to the network for interpretation and activation.

If per line CPN restriction is administered for a station, it will override any ISDN trunk group administration for sending calling party number.

Crisis alerts to a digital numeric pager

Crisis alert can also send notification of an emergency call to a digital pager. In this case, it sends a message of 7-digits to 22-digits to the pager and displays a crisis alert code, an extension and room number, and a main number (if one is entered). The person paged thus knows the origin of the emergency call and can direct emergency-service response to the appropriate location.

To use crisis alert with a digital pager, the system is administered so that at least one digital set has a CRSS-ALRT button and the `Alert Pager` field is set to **y**. Any station with a CRSS-ALRT button and a pager receives the correct alert.

Crisis alerts to a digital station

Crisis alert uses both audible and visual alerting to notify administered digital display stations when an emergency call is made. Audible alerting sounds like an ambulance siren. Visual alerting flashes the CRSS-ALRT button lamp and displays the name and extension, or room, of the caller. The crisis alert display of the origin of the emergency call enables the attendant or other user to direct emergency-service response to the caller.

When crisis alerting is active, the station is placed in position-busy mode so that other incoming calls can not interfere with the emergency call notification. The station can still originate calls to allow notification of other personnel.

If an emergency call is made while another crisis alert is still active, the incoming call will be placed in the queue. If the system is administered so that all users must respond, then every user must respond to every call, in which case the calls are not necessarily queued in the order in which they were made. If the system is administered so that only one user must respond, the first crisis alert remains active at the telephone where it was acknowledged. Subsequent calls are queued to the next available station in the order in which they were made.

Crisis alerts to an attendant console

Crisis alert uses both audible and visual alerting to notify attendant consoles when an emergency call is made. Audible alerting sounds like an ambulance siren. Visual alerting flashes the CRSS-ALRT button lamp and displays the name and extension, or room, of the caller. The crisis alert display of the origin of the emergency call enables the attendant or other user to direct emergency-service response to the caller. Though often used in the hospitality industry, it can be set up to work with any standard attendant console.

When crisis alerting is active, the console is placed in position-busy mode so that other incoming calls can not interfere with the emergency call notification. The console can still originate calls to allow notification of other personnel. Once a crisis alert call has arrived at a console, the console user must press the position-busy button to unbusy the console, and press the crisis-alert button to deactivate audible and visual alerting.

If an emergency call is made while another crisis alert is still active, the incoming call will be placed in the queue. If the system is administered so that all users must respond, then every user must respond to every call, in which case the calls are not necessarily queued in the order in which they were made. If the system is administered so that only one user must respond, the first crisis alert remains active at the telephone where it was acknowledged. Subsequent calls are queued to the next available station in the order in which they were made.

Emergency access to the attendant

Provides for emergency calls to be placed to an attendant. These calls can be placed automatically by the system or can be dialed by system users. Emergency access calls can receive priority handling by the attendant.

E911 CAMA trunk group

See E911 CAMA trunk group on page 135.

Privacy - auto exclusion

When the class of service (COS) is set for the automatic exclusion option, the feature is activated when you take your telephone off-hook. The feature can be deactivated when you push the exclusion button before dialing a call or during a call. An excluded call that is on hold can be taken off hold by any telephone that has a bridged appearance of the telephone that put the call on hold.

Privacy - manual exclusion

Allows multi-appearance telephone users to keep other users with appearances of the same extension number from bridging onto an existing call. Exclusion is activated by pressing the exclusion button on a per-call basis.

Restriction - controlled

See Restriction - controlled on page 198.

Station lock

Station lock allows users to lock their telephones to prevent unauthorized outgoing calls. Users can block outgoing calls and still receive incoming calls. This feature is activated by pressing a telephone button or dialing a feature access code (FAC).

Station lock allows users to block all outgoing calls, except for emergency calls, on all telephones, unless the telephone is pre-administered. An example of a pre-administered telephone is a telephone that is administered to block all outgoing calls except for emergency calls. Telephones can be remotely locked and unlocked.

Chapter 19: System management

Avaya Communication Manager system management provides the administrator powerful tools to maintain their communication solutions and to drive down the total cost of ownership.

Administration Without Hardware

See Administration Without Hardware on page 115.

Alternate facility restriction levels

This feature allows Communication Manager to adjust facility restriction levels or authorization codes for lines or trunks. Each line or trunk is normally assigned a facility restriction level. With this feature, alternate facility restriction levels are also assigned. Attendants can change to the alternates, thus changing access to lines and trunks. You might want to use this feature to disable most long-distance calling at night, for example, to prevent unauthorized staff from making long-distance calls.

CAUTION:

This feature may change the AAR and ARS routing preferences. Using it on tandem and tie-trunk applications affects entire networks. Calls that are part of a cross-country private network may be blocked.

Announcements

Use the Announcements feature to administer announcements that play for callers to your business. For example, you can inform callers that the call cannot be completed as dialed, the call is in a queue, or that all lines are busy. 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 on an adjunct, and are played back from the adjunct equipment.

Authorization codes - 13 digits

See Authorization codes - 13 digits on page 194.

Automatic circuit assurance

Assists in identifying possible trunk problems. Communication Manager maintains a record of the performance of individual trunks and automatically calls a designated user when a possible failure is detected. This feature provides better service through early detection of faulty trunks and consequently reduces out-of-service time.

Automatic transmission measurement system

Measures voice and data trunk facilities for satisfactory transmission performance. The measurement report contains data on trunk signal loss, noise, signaling return loss, and echo return loss. Acceptable performance, the scheduling of tests, and report contents are administrable.

Barrier codes

See Barrier codes on page 201.

Bulletin board

Provides a place on the switch where you can post information and receive messages from other switch users, including Avaya personnel. Anyone with appropriate permissions can use the bulletin board for everyday messages. In addition, Avaya personnel can leave high-priority messages that are displayed on the first ten lines of the bulletin board.

Busy verification of telephones and trunks

Allows attendants and users of multi-appearance telephones to make test calls to trunks, telephones, and hunt groups to check the status of an apparently busy resource. With this feature, an attendant or multifunction telephone user can distinguish between a telephone that is truly busy and one that only appears busy because of some problem. You can also use the feature to quickly identify faulty trunks.

Call charge information

Provides two ways to know the approximate charge for calls made on outgoing trunks:

- Advice of Charge, for ISDN trunks

  Advice of Charge (AOC) collects charge information from the public network for each outgoing call. Charge advice is a number representing the cost of a call; it is recorded as either a charging or currency unit.

- Periodic pulse metering, for non-ISDN trunks

  Periodic Pulse Metering (PPM) 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.

Call-charge information helps you to account for the cost of outgoing calls without waiting for the next bill from your network provider. This is especially important in countries where telephone bills are not itemized. You can also use this information to let employees know the cost of their telephone calls, and so encourage them to help manage your company telecommunications expenses.

Note:

This feature is not offered by the public network in some countries, including the United States.

In addition, the pass advice of charge to BRI endpoints feature will transparently pass AOC information that has been received from PRI networks to WCBRI endpoints.

Call Detail Recording

Records detailed call information on incoming and outgoing calls for the purpose of call accounting, and sends this call information to a Call Detail Recording (CDR) output device. You can specify the trunk groups and extensions for which you want records to be kept as well as the type of information to be recorded. You can keep track of both internal and external calls. This application contains a wide variety of administrable options and capabilities.

Call Detail Recording display of physical extension

For Expert Agent Selection (EAS) agent-originated calls, if the `Record Agent ID on Outgoing?` field on the **CDR System Parameters** screen is set to **y** (the default value), then the agent ID is used for outgoing calls.

If the `Record Agent ID on Outgoing?` field on the **CDR System Parameters** screen is set to **n**, the physical extension is used.

Call restrictions

By dialing an access code, administrators and attendants have the ability to restrict users from making or receiving certain types of calls. There are five restrictions:

- Outward. The user cannot place external calls.
- Station-to-station. The user cannot place or receive internal calls.
- Termination. The user cannot receive any calls (except priority calls).
- Toll. The user cannot place toll calls but can place local calls.
- Total. The user can neither place nor receive any calls.

Calling party/billing number

Allows the system to transmit calling party number/billing number (CPN/BN) information to an ISDN-PRI trunk group. The CPN is the calling party telephone number. BN is the calling party billing number. The CPN/BN may contain international country codes. It is used with an adjunct application.

Class of Restriction

See Class of Restriction on page 195.

Class of Service

Defines whether or not telephone users have permission to access features and functions. Examples of these features and functions are:

- Automatic callback
- Call forwarding
- Data privacy
- Priority calling
- Restrict call forwarding off-net
- Call forward busy/do not answer
- Extended forwarding and busy/do not answer
- Personal station access
- Trunk-to-trunk transfer restriction override
- Off-hook alert
- Console permission
- Client room

Classless Interdomain Routing

See Classless Interdomain Routing on page 151.

Concurrent user sessions

In order to increase the efficiency of administration and maintenance functions, the Communication Manager accommodates multiple concurrent administration and maintenance user sessions. Three or more devices (management terminals or operation support systems) can be connected to the switch to perform administration and/or maintenance tasks simultaneously.

Communication Manager supports eight concurrent administration and maintenance users. Five can perform concurrent administration, and three can perform concurrent maintenance. The eight concurrent sessions can be in any combination of local and remote connections.

Customer-provided equipment alarm

See Customer-provided equipment alarm on page 195.

Customer telephone activation

Enables customers to install their own telephones, eliminating the need for a service technician to do the installation. This feature is based on the TTI feature and allows the customer to associate a physical telephone with a station translations switch.

CTA is a streamlined version of TTI; it has a fixed feature-access code but does not require a security code. In addition, CTA allows only for “merging” of telephones with station translations, whereas TTI allows for both “merging” and “unmerging” of telephones with station translations.

CTA applies only to DCP and analog touch-tone, circuit-switched telephones.

DCS automatic circuit assurance

Allows a user or attendant at one node to activate or deactivate automatic circuit assurance referral calls for the entire DCS network. This transparency allows the referral calls to originate at a node other than the node that detects the problem.

External device alarming

Allows you to assign analog ports to alarm interfaces for external devices. You can specify a port location, information to identify the external device, and the alarm level to report when a contact closure occurs.

Facility busy indication

Allows users of multi-appearance telephones to see which lines, trunk groups, terminating extension groups, hunt groups, or paging zones (called resources or facilities) are busy. When the lamp associated with the resource is lit, the resource is busy.

You can store extension numbers, trunk group access codes, and loudspeaker paging access codes in a facility busy indication button. The facility busy indication button provides direct access to any of the facilities.

Facility restriction levels and traveling class marks

Allows certain calls to specific users, while denying the same calls to other users. For example, certain users may be allowed to use central office (CO) trunks to other corporate locations while other users may be restricted to less expensive private-network lines. **You can administer up to eight** levels of restriction for users of AAR and ARS.

Facility test calls

Allows telephone users to make test calls to access specific trunks, dual tone multifrequency receivers, time slots, and system tones. The user dials an access code and makes the test call to make sure the facility is operating properly. Security measures are included to prevent unauthorized use.

Firmware download

The firmware download feature allows you to download an image from a remote or local source into the system running Communication Manager, and use that image to reprogram the application code of a port circuit pack. This feature makes updating firmware more cost effective. This feature also reduces the expense of servicing the system port circuit packs because it eliminates the need for a technician to be involved when a board is updated. Firmware download is achieved using the TN799C CLAN interface.

Note:

Circuit packs that can be updated with the firmware download feature have a “P” at the end of their TN number.

Five EPN maximum in MCC1 Media Gateways

Note:

This feature is for MCC1 Media Gateways when used with an S8700 Media Server or DEFINITY® Server R configurations only.

This optional software feature allows customers that require high calling traffic capacities to have from two to five expansion port networks (EPN) in a single MCC1 Media Gateway. Only two port networks (PN) are generally available unless a specialized cable was purchased from Avaya and work-arounds were performed in software administration to make additional carriers function as EPNs.

When this feature is activated, Communication Manager enables administration of up to five carriers as EPNs and no custom cables are necessary. This means that the full bandwidth of the TDM bus is available to each carrier while still enabling the customer to have the footprint of an MCC1 Media Gateway. This is especially appealing to call centers without IPSI/PNC duplication, where systems can be quite large and heavily utilized.

The hardware limitation of the MCC1 Media Gateway is five port carriers. All five can be expansion port carriers, although traffic considerations may dictate some number less than that which is optimum. For example, a customer may choose to have three EPN carriers and two standard port carriers.

There is only one maintenance board, which is placed in carrier A. This is the only maintenance board in the cabinet.

Note:

Only two PNs are physically supported in S8700 Media Server IPSI-enabled systems when high/critical reliability options are desired. Only two PNs are physically supported in DEFINITY Server R systems when critical/ATM Network Duplication reliability is desired.

For more information on this feature, see your local Avaya representative.

Information and reports

- Attendant position report

  The attendant position report lists the following:

  - Attendant usage
  - Number of calls answered
  - Total time the attendant was available to answer a new call
  - Average holding time on calls answered
- Blockage study report
- Call coverage reports

  The call coverage report displays measurements of the distribution of traffic offered to call-coverage groups. Separate reports for all calls and external calls are supplied.
- Coverage points report

  The coverage points report differs based on whether all calls or external calls is selected. For each coverage point in the group, the number of calls offered, abandoned while at that coverage point, and overflowing to the next coverage point are listed.
- Display ARP reports
- Emergency and journal reports

  The emergency and journal report is based on information from all crisis alerts.
- Hunt group measurements report
- IP reports
- Packet error history report

  Provides a 24-hour history of important packet level statistics that indirectly indicate some LAN performance characteristics. The 24-hour history gives the ability to look back at these measures if the trouble cleared.
- Port network and link usage report

- Processor occupancy report

  The processor occupancy report provides summary information on how heavily the processor is loaded.

- Recent change history report

  Allows the system manager to view or print a history report of the most recent administration and maintenance changes on the switch. This report may be used for diagnostic or information purposes.

- Refresh route reports
- Summary report

  The summary report provides a performance summary of your system running Communication Manager.

- Tandem traffic report

  The tandem traffic report provides information on facilities that serve tandem traffic.

- Tracelog

  The Tracelog, among other things, lists:

  - all IP endpoint registrations
  - all IP endpoint unregistrations
  - all Ethernet interfaces coming into service
  - all Ethernet interfaces coming out of service

  These events are tagged as a new log type.

- Traffic reports

  Traffic reports show measurements in the format of switch-based reports for local or remote access, and can be collected for subsequent analysis and reporting by adjuncts and operation support systems using the operation support system interface protocol.

- Trunk group detailed measurements

Parsing capabilities for the history report

The history report provides details about every data command. You can use parsing options to limit the data returned in this report. The following table identifies the parsing options that are available.

Note:

You can display these options by entering the command **`list history`**, then clicking **HELP** or pressing **F5**.

Option	Description
date	Specify the month (**MM**) or day (**MM/DD**) for which to display history data.
time	Specify the hour (**HH**) or minute (**HH:MM**) for which to display history data.
login	Specify the login for which you wish to display history data.
action	Specify the command action (the first word of the command string) for which you wish to display history data. You can view the list of available command actions by clicking **HELP** or pressing **F5** at the command line.
object	Specify the command object for which you wish to display history data.
qualifier	Specify the command qualifier for which you wish to display history data.

To limit the data displayed in the history report, enter the command **`list history`** followed by a space and the appropriate parser and, if applicable, format. Only the data for the specified parsers will appear in the report.

You can include multiple parsers, but only a single instance of any parser (for example, you may parse for **date**, **time**, and **login,** but not for **date**, **time**, and two different **login**s).

IP asynchronous links

See IP asynchronous links on page 173.

Malicious call trace

See Malicious call trace on page 196.

Malicious call trace logging

See Malicious call trace logging on page 196.

Music-on-hold

Automatically provides music, silence, or tone to a caller. Music lets the caller know that the connection is still valid.

Local music-on-hold

The music on hold feature is supported on the G700 Media Gateway with Communication Manager. The music source is connected to a port on the MM711 Analog Media Module. Local music-on-hold is part of the call center functionality on the S8300 Media Server.

Local music-on-hold allows one music source. To use multiple music sources on a G700 Media Gateway, you must use multiple ports on the MM771 Analog Media Module, one for each music source.

For more information, see the *Installation for Adjuncts and Peripherals for Avaya Communication Manager*, 555-233-116. Also see the *Administrator Guide for Avaya Communication Manager*, 555-233-506.

Multiple music sources

On an MCC1, SCC1, CMC1, or G600 Media Gateway, this feature allows the customer to provide multiple distinct music sources for use with the call vectoring features, calls placed on hole, calls awaiting pickup, and so on. By purchasing the multiple music-on-hold (also called tenant partitioning) feature, you can have up to 100 music sources.

Many different music options can be administered to accommodate different tenants. See Tenant partitioning on page 218.

Restriction - controlled

Allows an attendant or telephone user, with console permission, to activate and deactivate for an individual telephone or a group of telephones, the following restrictions:

- outward
- total
- station-to-station
- termination restrictions

Scheduling

Functional scheduling in Communication Manager allows you to specify the time a command will be executed or to specify that it should be executed on a periodic basis. Only commands that do not require user interaction after being entered on the command line (such as list, display, test) can be scheduled.

Security Violation Notification

See Security Violation Notification on page 200.

Station security codes

See Station security codes on page 200.

Tenant partitioning

Allows partitioning of the system in order to lease the system services and features to multiple tenants. This provides attractive services and revenue for “virtual” landlords. It provides the robust features of a large system at affordable rates to small business tenants. Communication Manager supports up to 100 partitions and 27 attendant groups.

Multiple attendant groups can be assigned to each partition. Stations, hunt groups, and other endpoints assigned to a Class of Service (COS) can be partitioned. Network routing pattern preferences also support the assigned tenant partitioning. Tenant partitioning also allows you to assign a unique music source for each tenant partition for customers who are put on hold.

See Music-on-hold on page 216.

Terminal Translation Initialization

See Terminal Translation Initialization on page 120.

Time of day clock synchronization through a LAN source

Customers need accurate and common time of day time source across multiple switches in a network. This is especially important when customers are using a central Avaya Call Management System (CMS) to report events coming from multiple servers running Communication Manager.

The time of day clock synchronization through a LAN source feature is implemented on two different platforms:

- Linux
- UNIX

Linux platforms

Communication Manager that is running on Linux-based media servers synchronizes time directly from a LAN source.

UNIX platforms

Communication Manager running on DEFINITY servers which use an Oryx/Pecos operating system (proprietary UNIX-based OS) receives a command from Avaya site administration to adjust the time. Avaya site administration is synchronized to the LAN PC's clock.

Trunk group circuits

Trunks provide the communications links between Communication Manager and other switches, including central office switches and other premises switches. Trunks that perform the same function are grouped together and administered as trunk groups. Trunks interface with Communication Manager through port circuit packs.

Variable length ping

See Variable length ping on page 161.

Variable Length Subnet Mask

See Variable Length Subnet Mask on page 161.

Avaya Integrated Management

Avaya Integrated Management is a systems management software suite that contains applications to manage a converged voice and data network. The applications include:

- network management
- fault management
- performance management
- configuration management
- directory management
- policy management functionality

ATM WAN Spare Processor Manager

See ATM WAN Spare Processor Manager on page 123.

Avaya Communication Manager configuration manager

Avaya configuration manager provides centralized management of distributed network and campus environments, using a single point of entry and graphical Web-based interface for configuration and administration of multiple Avaya media servers.

Avaya Communication Manager fault/performance manager

Communication Manager fault/performance manager integrates with Avaya multiservice network manager to provide a system view of your converged network. Fault manager displays a hierarchical view of devices and their status, allowing you to view and isolate alarms and errors. Performance manager provides a comprehensive set of performance reports for trending and isolation of performance issues.

Avaya site administration

Avaya site administration is a Microsoft Windows-based graphical user interface for making changes, adding or moving users, and performing basic traffic analysis.

Voice Announcement over LAN manager

Avaya Voice Announcement over LAN (VAL) Manager is part of the Avaya Integrated Management suite of products. It enables you to the use of a LAN to transfer recorded announcements to Avaya media servers.

Announcements can be stored in **.wav** files, which can be sent to a voice announcement over LAN board without conversion. The voice announcement over LAN manager also provides a repository to backup and restore announcement files, and simplifies administration. With voice announcement over LAN manager, you can view the current status of announcements, easily add, change, and remove announcements, and copy and backup announcement files from Avaya media servers to the voice announcement over LAN manager and back, through the LAN.

Avaya VoIP Monitoring Manager

Avaya VoIP monitoring manager (VMON) provides the ability to monitor voice over IP (VoIP) network quality. This web-based application receives QoS statistics from Avaya IP end points and displays the data via graphs and reports, so administrators can isolate voice quality problems and send traps when poor voice quality is detected.

Directory

Allows users with display-equipped telephones to access the system database, use the touch-tone buttons to enter a name, and retrieve an extension number from the system directory. The directory contains the names and extensions assigned to all telephones on the system.

Administration change notification

Enables Communication Manager to communicate with the Avaya Directory Enabled Management (DEM) client. This feature enables the client to have real-time, integrated, directory-based, read/write access to Communication Manager administration data based on rules defined by the customer. Administration change notification enables the client to subscribe to notifications of changes to administration data in Communication Manager. It thus provides real-time updates whenever administration changes occur in a particular object (for example, a station).

Avaya Directory Enabled Management

Avaya Directory Enabled Management (DEM) is part of the Avaya Integrated Management suite of products It provides real-time, integrated, directory-based read/write access to Avaya media servers and INTUITY AUDIX messaging servers. It streamlines workflow and information management in an electronic environment using converged networks.

DEM creates a meta-directory for converged voice and data networks. It synchronizes directory information with data from Communication Manager and INTUITY devices, and stores the information in an LDAP-compliant directory service, for example, eDirectory from Novell, or active directory from Microsoft. Directory-enabled applications can then use the DEM to implement workflow processes that automate various system management functions and speed business operations.

Lightweight Directory Access Protocol

Lightweight Directory Access Protocol version 3 (LDAPv3) is an industry compliant protocol for accessing online directory services. A directory is like a database, but tends to contain more description information. Communication Manager integrates with LDAP datastores through the use of the administration change notification feature and Avaya directory enabled management client application to provide real-time, integrated, directory-based read/write access to Communication Manager and INTUITY AUDIX messaging servers.

Chapter 20: Telecommuting and remote office

Coverage of calls redirected off-net

Coverage of calls redirected off-net (CCRON) allows calls that have been redirected to locations outside of the switch to return to the switch for further processing.

For example, an employee that telecommutes can have two coverage paths. One coverage path is used when the employee is in the office and the other coverage path is used when the employee is working from home. The coverage path used from home takes a call to the employee work telephone, and covers the call to the employee home telephone. If the employee does not answer the call or is busy on another call, the call is redirected back to the switch for further processing, such as coverage to voice mail.

Remote call coverage and call forwarding off-net allow calls to be redirected to a remote location. This allows you to have calls placed to your on-site office redirected to your home office. You can administer the system to either monitor calls and bring them back for additional processing if not answered or to leave calls at the remote (off-net) location.

Extended user administration of redirected calls (telecommuting access)

Extended user administration of redirected calls (also called telecommuting access) allows you to change the lead call coverage path or forwarding extension from any on-site or off-site location. Thus you can change the path or extension from your home office, for example.

IP Softphone and IP Agent - RoadWarrior mode

See IP Softphone and IP Agent - RoadWarrior mode on page 86.

IP Softphone and IP Agent - Shared Control mode

See IP Softphone and IP Agent - Shared Control mode on page 86.

IP Softphone and IP Agent - Telecommuter mode

See IP Softphone and IP Agent - Telecommuter mode on page 86.

IP Softphone

See Avaya IP Softphone on page 85.

Off-premises station

A trunk-data module connects off-premises private-line trunk facilities and Communication Manager. The trunk-data module converts between the RS-232C and the DCP, and can connect to DDD modems as the DCP member of a modem pool.

See Call redirection on page 229 and Call vectoring on page 63.

Remote access

Permits authorized callers from remote locations to access the system via the public network and then use its features and services.There are a variety of ways of accessing the feature. After gaining access, you hear a system dial tone, and, for system security, may be required to dial a barrier code.

Chapter 21: Telephony

Abbreviated Dialing

Abbreviated Dialing (AD) provides lists of stored numbers you can use to:

- Place local, long-distance, and international calls
- Activate features
- Access remote computer equipment

You simply dial the list number and the one-digit, two-digit, or three-digit number associated with the telephone number you want. The number is then automatically dialed by the system. A frequently called number can be stored on an abbreviated dialing button that you need only press once to make the call.

Abbreviated dialing labeling

An administrator can type personalized labels for the Abbreviated Dialing (AD) System List entries. Users of the 2420 DCP telephone, as well as the 4600-series, 6400-series, and 8400-series display telephone sets, can administer labels for the AD softkey buttons. These personalized labels appear on the menu display.

These personalized labels can be administered in the standard supported languages (English, French, Italian, Spanish, and a user-defined language). If a personalized label has not been administered for the AD system list entry, the feature button label that is downloaded to the telephone is `AD`***`nn`***, where ***`nn`*** is the abbreviated dialing number.

Note:

This enhancement applies only to the AD system list.

Abbreviated dialing on-hook programming

On-hook programming allows users of the 2420 DCP telephone, as well as the 4600-series, 6400-series, and 8400-series telephone sets with enabled speakers, to access the programming mode without going off-hook during available call appearances. Signaling changes from DTMF to the S-channel, allowing the use of a longer (60 seconds) time-out period. Signaling will remain DTMF and the current time-out period of 10 seconds will still apply to non-display telephone sets.

Active dialing

6400-series and 4600-series telephone sets have a dialing option where the set will send S-channel button codes when the user presses a number on the dial pad when on-hook.

Administrable timeout on call timer

Enhances the call timer feature on the 6400-series telephones. The call timer feature measures the duration of a call, starting a timer when the call is answered and stopping the timer when the call is dropped.

Previously, the call timer feature displayed the duration of the call for only five seconds after the call was dropped. The administrable timeout on call timer feature allows the user to specify how long to display the duration of the call.

Alphanumeric dialing

See Alphanumeric dialing on page 89.

Automatic Call Back

Automatic Call Back (ACB) allows internal users who placed a call to a busy or unanswered internal telephone to be called back automatically when the called telephone becomes available.

When a user activates automatic callback, the system monitors the called telephone. When the called telephone becomes available to receive a call, the system originates the automatic callback call. The originating party receives priority ringing. The calling party then lifts the handset and the called party receives the same ringing provided on the original call.

Automatic Call Back for analog telephones

When a person, using an analog telephone, places a call and the line is busy, an announcement prompts the caller to enter the digit 1 to activate ACB, or to enter the digit 2 to route the call to a hunt group extension.

Automatic hold

Allows attendants and multi-function telephone users to alternate easily between two or more calls. For example, with automatic hold, selection of a second call automatically puts the active call (if any) on hold and makes the second call active. This feature can be activated on a system-wide basis only. When automatic hold is not activated, the selection of the second call drops the first call.

Bellcore calling name ID

Allows the system to accept calling name information from a Local Exchange Carrier (LEC) network that supports the Bellcore calling name specification. The system can send calling name information in the format if Bellcore calling name ID is administered. The following caller ID protocols are supported.

- Bellcore (default) - US protocol (Bellcore transmission protocol with 212 modem protocol)
- V23-Bell - Bahrain protocol (Bellcore transmission protocol with V.23 modem protocol).

Bridged call appearance - multi-appearance telephone

Allows calls made to or from a primary telephone extension to be handled from more than one telephone. A bridged call appearance is set up by administering a primary extension and the button number associated with it on a multi-lamp button on another telephone. This feature is most often used by secretaries or assistants who answer or handle calls to the primary extension (an executive, for example).

When the primary extension receives a call, the bridged call appearance flashes or rings on all telephones administered with this feature. The call can be answered by anyone having a telephone with this feature and handled as if the primary extension user was answering it. The maximum number of bridged appearances is 64.

Bridged call appearance - single-line telephone

Allows single-line telephones users to have a bridged appearance on a multi-appearance telephone.

Call coverage

Call coverage provides automatic redirection of calls that meet specified criteria to alternate answering positions in a call coverage path. A coverage path can include any of the following:

- a telephone
- an attendant group
- a Uniform Call Distribution (UCD) hunt group
- a Direct Department Calling (DDC) hunt group
- an Automatic Call Distribution (ACD) hunt group
- a voice messaging system
- a Coverage Answer Group (CAG) established to answer redirected calls

Alphanumeric field designation

In addition to numeric designations for key system lists and groups of related information, the system administrator can specify alphanumeric designations, 0-15 characters in length, for the following:

- abbreviated dial lists
- abbreviated dial groups
- call pickup groups
- call routing patterns

Changeable coverage paths

Changeable coverage paths allows the end user to modify the coverage points by using a feature access code (FAC).

Time of day

This feature allows a user to have multiple coverage paths depending on the time of day, and day of the week.

Call redirection

Call forward busy/do not answer

Allows calls to be forwarded when the called extension is busy or when the call is not answered after an administrable interval. If the extension is busy, the call forwards immediately. If the extension is not busy, the incoming call rings the called extension, then forwards only if it remains unanswered longer than the administered interval.

Call forwarding all calls

Allows calls to be forwarded to an internal extension, external (off-net) number, an attendant, or an attendant group. You can include an access code or special characters, like pause characters, in the forwarding destination.

Call forwarding override

Allows the user at the forwarded-to extension to override call forwarding and either initiate a call or transfer a call back to the forwarded-from extension.

Call redirection intervals

Communication Manager allows the system administrator to specify the number of times that a call rings at each call coverage point before the call proceeds to the next coverage point.

Call park

Allows you to put a call on hold and then retrieve a call from any other telephone on the system. This is helpful when you are on a call and need to go to another location for information. It also allows you to answer a call from any telephone after being paged by a telephone user or an attendant.

Call pickup

Along with directed call pickup, allows you to answer calls for other telephones within your specified call pickup group. Directed call pickup allows you to pick up any call on the system. With this feature, you do not have to leave your telephone to answer a call for a nearby telephone. You simply dial an access code or press a call pickup button.

Group call pickup

Allows you to dial a feature access code (FAC) and a pickup group number to answer a call from a different group. For example, marketing would be able to pickup calls in the sales group when the sales group is unavailable. This feature is ideal for offices that are not divided by partitions and generally have the departments on the same floor.

Caller ID on analog trunks

See Caller ID on analog trunks on page 143.

Caller ID on digital trunks

See Caller ID on digital trunks on page 143.

Circular station hunt group

See Circular station hunt group on page 66.

Conferencing

See Conferencing on page 75.

Consult

Allows a covering user, after answering a call received through call coverage, to call the called party for private consultation. Consult can be used to let a covering user ask the principal if they want to speak with the calling party.

Coverage callback

Allows a covering user to leave a message for the called party to call back the person who called.

Coverage incoming call identification

Allows multi-appearance telephones users without a display in a coverage answer group to identify an incoming call to that group.

Disconnecting unanswered calls

Disconnects unanswered outgoing calls after a predetermined amount of time. When any of the following timers expire during an outgoing local, toll, or international call attempt, the switch disconnects the call and applies busy tone, which may or may not be followed by howler tone:

- Pre-dialing and interdigit timer
- Outgoing seizure acknowledge timer
- Answer supervision timer
- 60-, 90-, and 120-second no-answer disconnect timers, based on ARS call type
- 120-second timer used for calls without a call type, such as calls to trunk access codes

Distinctive ringing

Rings or activates alerting on your telephone in such a way that you are aware of the type of incoming call before answering it. This feature operates in a Distributed Communications System (DCS) environment the same as it does within a single system.

By default, internal calls are identified by a 1-burst ringing pattern, external calls by a 2-burst ringing pattern, and priority calls by a 3-burst ringing pattern. You can administer these patterns.

Emergency calls from unnamed IP endpoints

With the Emergency calls from unnamed IP endpoints feature, an IP telephone can register without an extension number. The Emergency calls from unnamed IP endpoints feature places the IP telephone into Terminal Translation Initialization (TTI) service. Users can dial a feature access code (FAC) to either associate an extension number with a telephone, or to dissociate an extension number from a telephone.

If Communication Manager is appropriately administered, a user can use an IP telephone that is in TTI service to make emergency or other calls.

The Emergency Calls from Unnamed IP Endpoints feature requires IP telephone software R2.3 or later. IP telephone software R2.3 or later requires TN799C hardware vintage 3 or later circuit packs. TN799C hardware vintage 1 and 2 circuit packs do not work with IP telephone software R2.3. All versions of TN799DP circuit packs are compatible with IP telephone software R2.3. Therefore, the Emergency Calls from Unnamed IP Endpoints feature requires TN799DP or TN799C hardware vintage 3 or later circuit packs.

Enhanced abbreviated dialing

Supplements abbreviated dialing by providing one enhanced number per system. Enhanced number lists can contain any number or dial access code. System administrators designate privileges for group number lists, system number lists and enhanced number lists. With privileged lists, users can access otherwise restricted numbers (for example, telephones without long-distance access can be programmed to access specified long-distance numbers).

The S8700 Media Server supports 20,000 entries within the enhanced abbreviated dialing system list. This second enhanced abbreviated dialing list doubles the capacity to from 10,000 entries to 20,000 entries.

Future increases to the enhanced abbreviated dialing list can be performed easily by increasing the number of lists. Increasing the number of lists increases the overall capacity by multiples of 10,000 entries.

Enhanced telephone display

The enhanced telephone display feature allows you to choose the character set that you want to see in Communication Manager softkeys and display telephones. In addition to the standard Roman character set, you can choose either the Katakana or characters used for most European languages.

Enterprise Mobility User

Use the Enterprise Mobility User (EMU) feature to associate the features and permissions of the primary telephone of a user to a telephone of the same type anywhere within the customer enterprise. With EMU, the Class of Restriction (COR) and Class of Service (COS) permissions of the primary telephone are not associated with other telephones in the system.

Note:

Any telephone that is not the primary telephone is referred to as the visited telephone. Any server that is not the home server of the primary telephone is referred to as the visited server.

The following list outlines the requirements for the EMU feature:

- QSIG must be the private networking protocol in the network of Communication Manager systems. This requirement also includes QSIG MWI.

- Communication Manager Release 3.1 and later software must be running on the home server and all visited servers.
- All servers must be on a Linux platform. EMU is not supported on DEFINITY servers.
- The visited telephone must be the same model type as the primary telephone to enable a optimal transfer of the image of the primary telephone. If the visited telephone is not the same model type, only the call appearance (**call-appr**) buttons and the message waiting light are transferred.
- All endpoints must be self-designating terminals.
- Uniform Dial Plan (UDP).

Calls made from a visited telephone can be processed by either the home server or the visited server. Which server processes the call depends on how the user originates the call. The home server processes any calls that are a result of a user depressing one of the buttons that were downloaded to the visited telephone. The visited server processes any calls that are placed on the visited telephone using the dial pad.

Enterprise Wide Licensing

Enterprise Wide Licensing (EWL) is a technology within Communication Manager release 3.0 and a Communication Manager license file. EWL is used to partially support a developing offer known as Enhanced Software License Program (ESLP). ESLP gives customers the ability to bulk purchase and then share license capacities across multiple locations.

Note:

Launch of the ESLP offer will be announced at a later date.

Go to cover

Allows users who call another internal extension to send the call directly to coverage.

Hold

Allows you to disconnect from a call temporarily, use your telephone for other call purposes, and then return to the original call.

Intercom - automatic answer

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 telephones with built-in speaker, headphones, or adjunct speakerphone.

Internal automatic answer

Allows specific telephones to answer incoming internal calls automatically. This feature is intended for use with telephones that have speakerphones or headsets. You simply press an internal automatic answer feature button, and calls are automatically answered when the telephone is idle. Internal and Distributed Communications System (DCS) calls can be answered using automatic answer, but only attendants can use automatic answer to answer external calls directed to the attendant.

Last number dialed

Allows you to automatically redial the last number dialed. The system saves the first 24-digits of the last number dialed, whether the call attempt was manually dialed or dialed using abbreviated dialing. When you press the last number dialed button or dial the last number dialed feature access code, the system places the call again.

Local call timer automatic start/stop

Automatically starts the local timer of a 6400-series telephone when a call is received. The timer is stopped automatically when a call is ended. When a call is placed on hold the timer continues to run, but is not displayed. When the call comes off hold, the total elapsed call-time displays.

Long hold recall

Visual and audible warnings are sent to the telephone where a call has been on hold past a specified period of time. Both visual and audible warnings are used if the telephone is on-hook. If the telephone is off-hook, a “priority ring” is used. This is an optional feature at the system level.

Manual originating line service

Connects single-line telephone users to the attendant automatically when the user lifts the handset. The attendant number is stored in an abbreviated dialing list. When the telephone user lifts the handset, the system automatically routes the call to the attendant using the hot line service feature.

Misoperation handling

Defines how calls are handled when a misoperation occurs. A misoperation is when calls are left on hold when the controlling station goes on hook.

For example, a misoperation can occur under either of the following conditions:

- If you hang up prior to completing a feature operation (in some cases, hanging up completes the operation, as in call transfer). If, for example, you place a call on hold, begin to transfer the call, dial an invalid extension number, and then hang up, that is a misoperation.
- When the system enters night service while attendant consoles have calls on hold.

The system administrator can alter the standard misoperation handling to ensure that an external caller is not left on hold indefinitely, or dropped by the system after a misoperation with no way to reach someone for help.

Note:

This feature is required only in France and Italy, but it can be used at any location where the feature has been turned on.

Multiappearance preselection and preference

Provides options for placing or answering calls on selected call appearances.

- Ringing appearance *preference* automatically connects you to the incoming ringing call when the user picks up the handset.
- *Idle appearance preference* automatically connects you to an idle appearance.
- *Preselection* allows the user to manually select an appearance. Preselection is used, for example, when you want to reconnect with a held call or activate a feature.

  Preselection can be used with a feature button. For example, if you press an abbreviated dialing button, the call appearance is automatically selected and, if you pick up the handset within five seconds, the call is automatically placed. The preselection option overrides both of the other preference options.

Multiple level precedence and preemption

Multiple level precedence and preemption (MLPP) is an optional group of features that provide users the ability to interface to and operate in a Defense Switched Network (DSN). The DSN is a highly secure and standards-based communication system of the US Government Department of Defense (DoD).

CAUTION:

MLPP is currently designed to meet only DoD GSCR requirements for connection to a DSN by federal, state, or local government agencies. As such, MLPP is not currently designed for use in commercial enterprise environments. Activation of this feature in any other kind of network environment could result in unexpected and unwanted feature operations.

The MLPP features allow users to request priority processing of their calls during critical situations. The MLPP features include:

- Announcements for precedence calling
- Dual homing
- End office access line hunting
- Line load control
- Precedence call waiting
- Precedence calling
- Precedence routing

- Preemption
- Worldwide numbering and dialing plan

Announcements for precedence calling

In certain situations, precedence calls are blocked because of unavailable resources or improper use. When this occurs, recorded announcements are used to identify what went wrong. The announcements used for MLPP include:

- Blocked precedence call
- Unauthorized precedence level attempted
- Service interruption prevented call completion
- Busy, not equipped for preemption or precedence call waiting
- Vacant code

Dual homing

Dual homing allows a user to dial a telephone number and, if the initial route is unavailable, have the call route to its destination over alternate facilities.

End office access line hunting

End office access line hunting automatically hunts for an idle trunk over end office access lines, based on the precedence level of the call.

Line load control

Line load control is a feature that restricts a predefined set of station users from originating calls during a crisis or emergency. Through administration, users are assigned to a line load control level based on their relative importance. When an emergency occurs, the administrator manually enables the feature to restrict calling by users of lower importance. When the emergency is over, the administrator manually disables the feature.

For example, if a situation occurs that threatens national defense, station users in the defense department will not be restricted from originating calls, but stations in other departments, such as the accounting department, will be restricted. When the crisis is over, the system can be returned to normal operation by the administrator.

Precedence call waiting

After a precedence call is routed, the called party may already be busy on another call. Precedence call waiting allows the caller to “camp on” to the line of the called party and wait for the user to answer the call. The caller hears a special ringback tone and the called party hears a call waiting tone.

Depending on the type of telephone being used, the called party can put the current call on hold and answer the call, or the called party must hang up on their current call to answer the incoming call.

Precedence calling

Precedence calling is the centerpiece of the MLPP features. Precedence calling allows users, on a call-by-call basis, to select a level of priority for each call based on their need and importance (rank). The call receives higher-priority routing, whether the call is local or going around the world.

Users may access five levels of precedence when placing calls:

- Flash Override (the highest precedence level)
- Flash
- Immediate
- Priority
- Routine (the default, and lowest precedence level)

Each station user is administered with a maximum precedence level. The more important or higher in rank of the user, the higher the precedence level. Users cannot originate calls at precedence levels higher than their maximum administered level. Non-MLPP calls are treated as routine level precedence calls.

Precedence routing

When precedence calls are destined for other switches in a private network, the precedence routing feature is used to route the calls. The precedence routing feature routes calls based on three main criteria:

- Routing based on the destination number
- Routing based on the precedence level
- Routing based on the time of day

These routing criteria are administrable and can be changed as required. Two related features are dual homing and end office access line hunting.

Preemption

Preemption works with the precedence routing feature to further extend the call routing capabilities of the MLPP features. Preemption, when allowed through administration, can actually tear down an existing lower priority call in order to complete a more important precedence call. Even non-MLPP calls are treated as routine level precedence calls and can be preempted.

When this occurs, the callers on the existing call hear a tone indicating that the call is about to be preempted. The callers have three seconds to end the call before the call is automatically disconnected. After the existing call is disconnected, the new call is placed using preempted facility.

Worldwide numbering and dialing plan

The worldwide numbering and dialing plan (WNDP) feature allows Communication Manager to conform to the standard numbering system established by the Defense Communications Agency (DCA). WNDP defines its own format for the precedence dialing. The capability to operate with this numbering plan must be incorporated into all new switches introduced into the Defense Switched Network (DSN).

Night service

There are five night service features:

- Hunt group night service allows an attendant or a split supervisor to assign a hunt group or split to night service mode. All calls for the hunt group then are redirected to the hunt group designated night service extension. When a user activates hunt group night service, the associated button lamp lights.
- 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. To activate and deactivate this feature, the attendant typically presses the night button on the principal attendant console or designated console.

- Night station service directs incoming calls for the attendant to designated extensions. Attendants can activate night station service by pressing the night button on the principle console if there is not an active night console. If the night station is busy, calls (including emergency attendant calls) receive a busy tone. They do not queue for the attendant.
- Trunk answer from any station allows telephone users to answer all incoming calls to the attendant when the attendant is not on duty and when other telephones have not been designated to answer the calls. The incoming call activates a gong, bell, or chime and a voice-terminal user dials an access code to answer the call.
- Trunk group night service allows an attendant or a designated telephone user to individually assign a trunk group or all trunk groups to the night service mode. Specific trunk groups individually assigned to the service are in Individual trunk night service mode. Calls coming into these trunk groups are redirected to designated night service extensions. Incoming calls on other trunk groups are processed normally.

Enhanced night service

Communication Manager informs a voice mail system (VMS) that it is in night service, allowing the VMS to perform different actions and call handling for out-of-hours operation. For example, the VMS may be administered to provide recorded announcements after hours. The enhancement is made to the mode code voice mail interface.

License modes

The three modes of license operation for your system are:

- License-normal mode
- License-error mode
- No-license mode

License-normal mode

The license-normal mode is the desired mode of operation of a stable system. During this mode of operation, a license is properly installed, the license contains a serial number that matches the processors, the license is not expired, and feature usage does not exceed limits.

License-error mode

The license-error mode is a warning mode. During this mode, call processing is supported, the system declares a major alarm, and a 6-day countdown timer is running. If this timer is allowed to expire, the no-license mode is invoked.

The license-error mode is entered as a result of one of the following conditions:

- The serial number of the active processor does not match the license file.
- The standby processor cannot be contacted or the serial number of the standby processor does not match the serial numbers in the license.
- The license has expired.
- Feature usage exceeds limits. For example, there are more ports translated than permitted by the port limit in the license.
- A WAN spare or survivable remote processor enters License-Error mode when it is providing primary service.
- A switch has initialized after a software upgrade and a new license has not yet been installed.

The license-error mode is cleared by correcting the error that caused the system to enter into license-error mode, or by installing a valid license that is consistent with the configuration of the switch.

No-license mode

The no-license mode is a state in which all new call originations, except alarm calls and calls to an administered emergency number, are denied. All incoming calls, except calls to an administered number, are also denied.

The no-license mode state is entered as the result of one of the following reasons:

- No license is installed in the system.
- The License-Error timer has expired.
- A survivable remote processor detects a port board in its port network other than an Expansion Interface board.
- A reset system 3 preserve-license command is executed and the offer category in translation does not match the offer category in the license.

Starting with Communication Manager release 2.1, no-license mode not only protects customers from loss of call processing, but also provides software copy protection. The result of no-license mode is an error message on telephone displays, and blocked system administration.

No-license mode is cleared by correcting the error that caused the system to enter into no-license mode, or by installing a valid license that is consistent with the configuration of the system.

Personalized ringing

Allows users of certain telephones to uniquely identify their own calls. Each user can choose one of a number of possible ringing patterns. The eight ringing patterns are tone sequences consisting of different combinations of three tones. With this feature, users working closely in the same area can each specify a different ringing pattern in order to better identify their own calls.

Posted messages

In most situations, after a few rings when no one answers a call, the calling party usually hears an announcement saying that the called party is not available and to please leave a message. At this point, the calling party has no clue when the called party would return the call.

The posted messages feature provides Communication Manager users with the capability of indicating the reason of their unavailability to calling parties. The system provides 30 messages for a user to choose from, such as “on vacation,” or “at lunch.” Of the 30 messages, 15 messages are fixed system messages, and the remaining 15 messages are administrable (custom messages). After a user has chosen one of the messages and thus activated the feature, the message is immediately sent to calling parties who have terminal displays.

The system provides two ways to activate/deactivate this feature: using button pushes and feature access codes. The system allows users to use the feature access codes from their own display telephone, from another station/attendant, or from a remote access trunk.

Priority calling

Allows you to ring another telephone with a distinctive signal that tells the called party the incoming call requires immediate attention. The called party can then handle the call accordingly. You activate priority calling by dialing a priority calling access code or pressing a feature button, followed by the extension number. You can use priority calling only if your telephone has been administered with the required class of service.

Processor Ethernet

The Processor Ethernet (PE) interface is one way of connecting VoIP and IP-based devices to Communication Manager. The PE interface is the appearance of the computer's native ethernet interface inside of the Communication Manager application.

An Enterprise Survivable Server (ESS) and a Local Survivable Processor (LSP) registers with the main server when it is first configured, and every time it receives a file sync from the main server. An ESS server and an LSP uses the PE interface to register with the main server. You do not have to administer the PE interface for registration purposes. The system software enables the use of the PE interface on servers configured as an LSP or and ESS server.

CAUTION:

Do not disable the PE interface on an LSP or an ESS server. Disabling the PE interface disables the LSP or ESS server's ability to register with the main server. The LSP or ESS server will not work if the PE interface is disabled.

On simplex main servers, you can administer the server where IP endpoints may register through:

- The PE interface only
- CLAN interfaces only (requires the configuration to have CLANs)
- Either the PE interface or specified CLAN interfaces (requires the configuration to have CLANs) That is, both interfaces must be able to be enabled at the same time with some endpoints registering through the PE interface and some through CLANs.

Adjuncts

If the main server is a simplex server, adjuncts that connect to the simplex main server using a CLAN can also connect to the simplex main server using the server's PE interface.

Three adjuncts are supported for connectivity to the PE interface of a LSP or an ESS server:

- CMS
- CDR
- Application Enablement Services (AES)

An adjunct link is established between the LSP or the ESS server. Administration that allows dedicated and shared connections between the adjuncts and the servers must allow for the link to remain active at all times. When the LSP or ESS server is not active, the adjunct does not receive data from that server.

For information on how to a administer the adjunct link, see the adjunct documentation that is specific to your adjunct.

H.248 and H.323 registration

The system uses the PE interface on an LSP to register H.248 gateways and H.323 endpoints. Starting with Communication Manager Release 3.1, the use of the PE interface to register H.248 gateways and H.323 endpoints has been expanded to include the simplex main server.

S8500 Media Servers

With the increased functionality of the PE interface, the role of the S8500 Media Server has been expanded. This expansion includes an S8500 configured as an LSP, and an S8500 configured as a main server in an IP configuration with no port networks.

Note:

The PE interface is not supported for H.248, H.323, and adjunct connectivity on a duplex (S8700-series) server.

Pull transfer

Allows *either* the party who was originally called, *or* the party to whom the held call will be transferred, to complete the transfer. This is a convenient way to connect a party with someone better qualified to handle the call. Attendant assistance is not required and the call does not have to be redialed. It interfaces with satellite workstations through TGU/TGE trunks and is always available for calls that use TGU/TGE trunks.

Recall signaling

Recall signaling allows the user of an analog station to place a call on hold, use the telephone for other call purposes, and then return to the original call.

Recorded telephone dictation access

Allows telephone users, including remote access and incoming tie trunk users, to access dictation equipment. The dictation equipment is accessed by dialing an access code or extension number. The start/stop function can be voice or dial controlled. Other functions such as initial activation and playback are controlled by additional dial codes.

Reset shift call

If a called number is busy and does not have coverage, or the called number and the coverage are both busy, you have an opportunity to replace the last digit that was entered. This allows you to call another extension without having to hang up and redial. Reset shift call is a feature that is active for station to station (internal) calls and for private network calls. The private network trunks must signal busy using out-of-band signaling.

Ringback queuing

Places calls in an ordered queue (first in, first out) when all trunks are busy. The telephone user who is trying to make a call is automatically called back when a trunk becomes available, and hears a distinctive three-burst signal when called back.

Ringer cutoff

Allows the user of a multi-appearance telephone to turn audible ringing signals on and off. Visual alerting is not affected by this feature. When this feature is enabled, only priority (three-burst) ring, redirect notification, intercom ring, and manual signaling ring at the telephone. Internal and external calls do not ring.

Ringing - abbreviated and delayed

Allows you to manually or automatically assign one of four ring types to each call appearance on a telephone. Whatever treatment you assign to a call appearance is automatically assigned to each of its bridged call appearances.

Ringing options

Provides multi-appearance telephone users with different ringing patterns. This feature primarily affects audible ringing for calls directed to telephones that are off hook, or calls directed to idle and active CALLMASTER telephones.

Send all calls

Allows users to temporarily direct all incoming calls to coverage regardless of the assigned call-coverage redirection criteria. Covering users can temporarily remove their telephones from the coverage path. The feature is activated and deactivated via a button or access code.

Special dial tone

Provides the ability to play a special dial tone whenever an analog set is not able to receive calls. When such conditions as call forward all calls, call forward busy/no answer, send all calls, or do not disturb are activated on a telephone set, a special dial tone lets you know that you cannot receive any calls.

Station hunting

Routes calls made to a busy extension to another extension. To use station hunting, you create a station hunting chain that governs the order in which a call routes from one extension to the next when the called extension is busy. Each extension in the chain links *to* only one subsequent extension. However, an extension may be linked *from* any number of extensions.

Station hunt before coverage

This feature changes the interaction that occurs between station hunting and call coverage. Station hunt before coverage causes a call going to a busy station to go through a station hunting process before going to coverage. If all the stations in the hunt group are busy, the call will go to the coverage path.

Station self display

Station self display shows the extension number of the telephone set when a user either dials the feature access code while off-hook, or depresses the **INSPECT** button when on-hook. The dialed number will be displayed once the user starts to dial. This feature is helpful to people who move from one desk to another while they are working. This feature is also used by maintenance personnel to ensure that an extension number is correctly administered.

Station used as a virtual extension

Allows a customer to assign multiple, individual, virtual extensions to one physical telephone. The physical telephone must be analog and on the local switch. The administrator can set each virtual extension with a unique ring pattern to identify the extension for which the incoming call is intended. For example, an administrator could assign three virtual extensions, each with a unique ring pattern, to a single telephone shared by three roommates in a college dormitory. This feature affects incoming calls only; all outgoing calls are associated with the physical extension.

Support for the Hewlett Packard DL380G2 server

Communication Manager is supported on Hewlett Packard (HP) DL380G2 servers in an S8700 IP-connect system configuration (an S8700 Media Server with a G600 Media Gateway).

Telephone display

Provides multi-appearance telephone users with updated call and message information. This information is displayed on a display-equipped telephone. The information displayed depends upon the display mode selected by the user. Information that allows personalized call answering is available on many calls.

Users may select any of the following as the display message language: English (default), French, Italian, or Spanish. In addition, messages can be administered on the system in a fifth language. The language for display messages is selected by each user.

Telephone self administration

The telephone self administration capability allows you to program feature buttons on the telephone yourself.

Temporary bridged appearance

Allows multi-appearance telephone users in a terminating extension group or personal central office line group to bridge onto an existing group call. If a call has been answered using the call pickup feature, the originally called party can bridge onto the call. This feature also allows a called party to bridge onto a call that redirects to coverage before the called party can answer it.

Terminating extension group

Allows an incoming call to ring (either audible or silent alerting) as many as four telephones at the same time. Any user in the group can answer the call. Any telephone can be administered as a group member. Only a multi-appearance telephone can be assigned a feature button with an associated status lamp, however.

The feature button allows the user to select a terminating extension group call appearance for answering or bridging onto an existing call but not for call origination. For example, a department in a large store might have three telephones. Anyone in the department can answer the call. The salesperson most qualified to answer the call can bridge onto the call.

Time of day routing

Provides the most economical routing of **ARS** and **AAR** calls. This routing is based on the time of day and day of the week that each call is made. Up to eight TOD routing plans may be administered, each scheduled to change up to six times a day for each day in the week.

This allows you to take advantage of lower calling rates during specific times of the day and week. In addition, companies with locations in different time zones can use different locations that have lower rates at different times of the day or week. This feature is also used to change patterns during the times an office is closed in order to reduce or eliminate unauthorized calls.

Timed call disconnection for outgoing trunk calls

This feature provides the capability to automatically disconnect an outgoing trunk call after an administrable amount of time. Warning tones are applied to all parties on the call prior to the disconnection.

The amount of time that can elapse before the trunk is dropped can be specified, and can vary between 2-999 minutes. If the timer field is blank, which is the default value, then the feature is disabled and the trunk will not be automatically disconnected.

Timed call disconnection applies to all outgoing trunk calls initiated by a party belonging to a specified Class of Restriction (COR).

Prior to disconnecting the trunk, warning tones are applied to all parties on the call. The first warning tone occurs when one minute is remaining on the call. The second warning tone occurs when 30 seconds are remaining on the call.

Transfer

Allows telephone users to transfer trunk or internal calls to other telephones within the system without attendant assistance. This feature provides a convenient way to connect a party with someone better qualified to handle the call.

Abort transfer

Allows a user to abort a transfer attempt by pressing a non-idle line appearance. The call being transferred would be taken off a transfer-type hold and be put on a traditional hold. The transfer will also be aborted when you hang up (going on-hook), unless transfer upon hang-up is activated on the switch. This is an optional feature at the system level.

Transfer - outgoing trunk to outgoing trunk

Allows a user or attendant to initiate two or more outgoing trunk calls and then transfer the trunks together. The transfer operation removes the original user from the connection and conferences the outgoing trunks. Alternatively, the controlling party can establish a conference call with the outgoing trunks and then drop out of the conference, leaving only the outgoing trunks on the conference. This is an optional enhancement to trunk-to-trunk transfer and requires careful administration and use. DCS trunk turnaround may be a safer alternative to this feature.

Transfer recall

Returns the unanswered transfer calls back to the person who transferred the call. Transfer recall uses a priority alerting signal, and the display on the telephone shows “rt”, which indicates a returned call from a failed transfer operation.

Transfer upon hang-up

Provides you with the ability to transfer a call by hanging up instead of having to press the transfer button a second time. You would press the transfer button, dial the number the call is being transferred to and then hang up. This is an optional feature at the system level. You will still be able to transfer a call by pressing the transfer button a second time.

Trunk-to-trunk transfer

Allows the attendant or telephone user to connect an incoming trunk call to an outgoing trunk call. This feature is particularly useful when a caller outside the system calls a user or attendant and requests a transfer to another outside number. For example, a worker, away on business, can call in and have the call transferred elsewhere. The system assures that incoming central office (CO) trunks without disconnect supervision are not transferred to outgoing trunks or other incoming central office trunks without disconnect supervision.

Trunk flash

Trunk flash allows a feature or function button on a multifunction telephone or attendant console to be assigned as a flash button. Pressing this button while connected to a trunk (which must have been administered to allow trunk flash) causes the system to send a flash signal out over the connected trunk.

Trunk flash enables multifunction telephones to access central office customized services that are provided by the central office to which the system running Communication Manager is connected. **These** services are electronic features, such as conference and transfer, that are accessed by a sequence of flash signal and dial signals from the system station on an active trunk call.

The trunk flash feature can help to reduce the number of trunk lines connected to the system. “Digit 1 as flash” as used in Italy, and the United Kingdom will not serve as the flash button in this application.

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