Avaya Aura™ Communication Manager Screen Reference

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Screen Reference

This document contains descriptions of Communication Manager screens that are used in performing administrative tasks. These are most often screens that are invoked using commands such as **add**, **change**, and **remove**.

For maintenance-related screens that are invoked using commands such as **list**, **display**, and **status**, see *Maintenance Commands for Avaya Aura™ Communication Manager, Media Gateways and Servers*, 03-300431.

AAR and ARS Digit Analysis Table

Avaya Aura™ Communication Manager compares dialed numbers with the dialed strings in this table and determines the route pattern for the number.

Note:

Typing the command **change aar analysis** or **change ars analysis** displays an all-locations Digit Analysis screen. To access a per-location screen, type **change aar analysis location *n*** or **change ars analysis location *n***, where ***n*** represents the number of a specific location. For details on command options, see online help, or *Maintenance Commands for Avaya Aura™ Communication Manager, Media Gateways and Servers*, 03-300431.

Field descriptions for page 1

Figure 1: AAR Digit Analysis Table screen

```
change aar analysis n                                          Page 1 of X

                             AAR DIGIT ANALYSIS TABLE
                                   Location:All         Percent Full:

     Dialed          Total    Route     Call   Node  ANI
     String         Min Max  Pattern    Type   Num   Reqd
_________________   __  __    _____     ____   ___    n
_________________   __  __    _____     ____   ___    n
_________________   __  __    _____     ____   ___    n
_________________   __  __    _____     ____   ___    n
_________________   __  __    _____     ____   ___    n
_________________   __  __    _____     ____   ___    n
_________________   __  __    _____     ____   ___    n
_________________   __  __    _____     ____   ___    n
_________________   __  __    _____     ____   ___    n
_________________   __  __    _____     ____   ___    n
```

Figure 2: ARS Digit Analysis Table screen

```
change ars analysis                                          Page 1 of X

                             ARS DIGIT ANALYSIS TABLE
                                   Location: All         Percent Full:

     Dialed          Total    Route     Call   Node  ANI
     String         Min Max  Pattern    Type   Num   Reqd
_________________   __  __    _____     ____   ___    n
_________________   __  __    _____     ____   ___    n
_________________   __  __    _____     ____   ___    n
_________________   __  __    _____     ____   ___    n
_________________   __  __    _____     ____   ___    n
_________________   __  __    _____     ____   ___    n
_________________   __  __    _____     ____   ___    n
_________________   __  __    _____     ____   ___    n
_________________   __  __    _____     ____   ___    n
_________________   __  __    _____     ____   ___    n
```

ANI Reqd

Valid entries	Usage
y/n	Enter **y** if ANI is required on incoming R2-MFC or Russian MF ANI calls. This field applies only if the **Request Incoming ANI (non-AAR/ARS)** field on the Multifrequency-Signaling-Related System Parameters screen is **n**.
r	Allowed only if the **Allow ANI Restriction on AAR/ARS** field on the Feature-Related System Parameters screen is **y**. Use to drop a call on a Russian Shuttle trunk or Russian Rotary trunk if the ANI request fails. Other types of trunks treat **r** as **y**.

Call Type (for AAR only)

Enter the call type associated with each dialed string. Call types indicate numbering requirements on different trunk networks. ISDN Protocols are listed in the table below.

Valid entries	Usage
aar	Regular AAR calls
intl	The Route Index contains public network ISDN trunks that require international type of number encodings.
pubu	The Route Index contains public network ISDN trunks that require unknown type of number encodings.
lev0 to lev2	Specify ISDN Private Numbering Plan (PNP) number formats. (See Numbering — Private Format on page 692 for more information.)
unku	The **unku** AAR Call Type makes it easier to set up an Implicit (Unknown) Numbering Plan, in which users dial each other by extension (optionally preceded by a node number), without an ARS or AAR Access Code (for example, “9” or “8”).

ISDN Protocol

Call Type	Numbering Plan Identifier	Type of Numbering
aar	E.164(1)	national(2)
intl	E.164(1)	international(1)
pubu	E.164(1)	unknown(0)
lev0	PNP(9)	local(4)

Call Type	Numbering Plan Identifier	Type of Numbering
lev1	PNP(9)	Regional Level 1(2)
lev2	PNP(9)	Regional Level 2(1)

Call Type (for ARS only)

Valid entries	Usage	China # 1 Call Type
alrt	alerts attendant consoles or other digital telephones when an emergency call is placed	normal
emer	emergency call	normal
fnpa	10-digit North American Numbering Plan (NANP) call (11 digits with Prefix Digit "1")	attendant
hnpa	7-digit NANP call	normal
intl	public-network international number	toll-auto
iop	international operator	attendant
locl	public-network local number	normal
lpvt	local private	normal
natl	non-NANP	normal
npvt	national private	normal
nsvc	national service	normal
op	operator	attendant
pubu	public-network number (E.164)-unknown	normal
svcl	national(2)	toll-auto
svct	national(2)	normal
svft	service call, first party control	local
svfl	service call, first party control	toll

Dialed String

User-dialed numbers are matched to the dialed string entry that most closely matches the dialed number. For example, if a user dials 297-1234 and the AAR or ARS Digit Analysis Table has dialed string entries of 297-1 and 297-123, the match is on the 297-123 entry.

An exact match is made on a user-dialed number and dialed string entries with wildcard characters and an equal number of digits. For example, if a user dials 424, and there is a 424 entry and an X24 entry, the match is on the 424 entry.

Valid entries	Usage
0 to **9**	Enter up to 18 digits that the call-processing server analyzes.
***, x, X**	wildcard characters

Location

This is a display-only field. Typing the command **`change aar analysis`** ***`n`*** or **`change ars analysis`** ***`n`*** displays the all-locations screen, and populates this field with **all**. The ***n*** specifies that dialed strings beginning with the value **`n`** are displayed first. To access a per-location screen, type **`change aar analysis location`** ***`n`*** or **`change ars analysis location`** ***`n`***, where ***n*** represents the number of a specific location. This field then displays the number of the specified location. For details on command options, see online help, or *Maintenance Commands for Avaya Aura™ Communication Manager, Media Gateways and Servers*, 03-300431.

Valid entries	Usage
1 to **64**	Defines the location of the server running Communication Manager that uses this AAR/ARS Digit Analysis Table. On the **System Parameters Customer-Options (Optional Features)** screen, the **Multiple Locations** field must be set to **y** for values other than **all** to appear. For ARS, the **ARS** field must also be set to **y** on the System Parameters Customer-Options (Optional Features) screen.
all	Indicates that this AAR/ARS Digit Analysis Table is the default for all port network (cabinet) locations. Appears only if the **Multiple Locations** field is **n** on the **System Parameters Customer-Options (Optional Features)** screen.

Max

Valid entries	Usage
Between Min and **28**	Enter the maximum number of user-dialed digits the system collects to match to the dialed string.

Min

Valid entries	Usage
1 to Max	Enter the minimum number of user-dialed digits the system collects to match to the dialed string.

Node Number

Valid entries	Usage
1 to **999** or blank	Enter the number of the destination node in a private network if you are using node number routing or DCS. If you complete this field, leave the **Route Index** field blank.

Percent Full

Displays the percentage (**0** to **100**) of the system's memory resources that have been used by AAR/ARS.

Route Pattern

Enter the route number you want the server running Communication Manager to use for this dialed string.

Valid entries	Usage
p1 to **p2000**	Specifies the route index number established on the Partition Routing Table
1 to **640**	Specifies the route pattern used to route the call.
1 to **999**	Specifies the route pattern used to route the call. For S8300 Servers only.

Valid entries	Usage
r1 to **r32**	Specifies the remote home numbering plan area table. Complete this field if RHNPA translations are required for the corresponding dialed string.
node	Designates node number routing
deny	Blocks the call

AAR and ARS Digit Conversion Table

Your system uses the AAR or ARS Digit Conversion Table to change a dialed number for more efficient routing. Digits can be inserted or deleted from the dialed number. For instance, you can tell the server running Communication Manager to delete a 1 and an area code on calls to one of your locations, and avoid long-distance charges by routing the call over your private network.

Note:
Typing the command **`change aar digit-conversion`** or **`change ars digit-conversion`** displays the all-locations Digit Conversion Table screen. To access a per-location screen, type **`change aar digit-conversion location`** ***n*** or **`change ars digit-conversion`** ***n***, where ***n*** represents the number of a specific location. For details on command options, see online help, or *Maintenance Commands for Avaya Aura™ Communication Manager, Media Gateways and Servers*, 03-300431.

Field descriptions for page 1

Figure 3: AAR Digit Conversion Table screen

```
change aar digit-conversion                                       Page 1 of 2

                       AAR DIGIT CONVERSION TABLE
                              Location:All                Percent Full:


  Matching Pattern     Min   Max   Del   Replacement String   Net  Conv ANI Req
  __________________   __    __    __    __________________   ___   _        _
  __________________   __    __    __    __________________   ___   _        _
  __________________   __    __    __    __________________   ___   _        _
  __________________   __    __    __    __________________   ___   _        _
  __________________   __    __    __    __________________   ___   _        _
  __________________   __    __    __    __________________   ___   _        _
  __________________   __    __    __    __________________   ___   _        _
  __________________   __    __    __    __________________   ___   _        _
  __________________   __    __    __    __________________   ___   _        _
  __________________   __    __    __    __________________   ___   _        _
```

Figure 4: ARS Digit Conversion Table screen

```
change ars digit-conversion                                       Page 1 of 2

                        ARS DIGIT CONVERSION TABLE
                              Location:All                Percent Full:

  Matching Pattern     Min   Max   Del   Replacement String   Net Conv ANI Req
  __________________   __    __    __    __________________   ___   _        _
  __________________   __    __    __    __________________   ___   _        _
  __________________   __    __    __    __________________   ___   _        _
  __________________   __    __    __    __________________   ___   _        _
  __________________   __    __    __    __________________   ___   _        _
  __________________   __    __    __    __________________   ___   _        _
  __________________   __    __    __    __________________   ___   _        _
  __________________   __    __    __    __________________   ___   _        _
  __________________   __    __    __    __________________   ___   _        _
  __________________   __    __    __    __________________   ___   _        _
  __________________   __    __    __    __________________   ___   _        _
  __________________   __    __    __    __________________   ___   _        _
  __________________   __    __    __    __________________   ___   _        _
```

Note:
When you access the screen with the `display` or `change` command, the entries are sorted in the order of the matching pattern. Digits appear before characters.

ANI Req

This field applies only if the **Request Incoming ANI (non-AAR/ARS)** field on the Multifrequency-Signaling-Related System Parameters screen is **n**.

Valid entries	Usage
y/n	Enter **y** to require ANI on incoming R2-MFC or Russian MF ANI calls. Must be **y** to enable EC500 origination features.
r	Allowed only if the **Allow ANI Restriction on AAR/ARS** field is **y** on the Feature-Related System Parameters screen. Use to drop a call on a Russian Shuttle trunk or Russian Rotary trunk if the ANI request fails. Other types of trunks treat **r** as **y**.

Conv

Valid entries	Usage
y/n	Enter **y** to allow additional digit conversion.

Del

Valid entries	Usage
0 to Min	Number of digits you want the system to delete from the beginning of the dialed string.

Location

This is a display-only field. Typing the command **`change aar digit-conversion`** ***`n`*** or **`change ars digit-conversion`** ***`n`*** displays the all-locations screen, and populates this field with **all**. The ***n*** specifies that dialed strings beginning with the value **`n`** are displayed first. To access a per-location screen, type **`change aar digit-conversion location`** ***`n`*** or **`change ars digit-conversion location`** ***`n`***, where ***n*** represents the number of a specific location. This field then displays the number of the specified location. For details on command options, see online help, or *Maintenance Commands for Avaya Aura™ Communication Manager, Media Gateways and Servers*, 03-300431.

Valid entries	Usage
1 to **64**	Defines the location of the server running Communication Manager for this AAR/ARS Digit Conversion Table. On the **System Parameters Customer-Options (Optional Features)** screen, the **Multiple Locations** field must be set to **y** for values other than **all** to appear. For ARS, the **ARS** field must also be set to **y** on the System Parameters Customer-Options (Optional Features) screen.
all	Indicates that this AAR/ARS Digit Conversion Table is the default for all port network (cabinet) locations.

Matching Pattern

Valid entries	Usage
0 to **9** (1 to 18 digits)	Enter the number you want the server running Communication Manager to match to dialed numbers. If a Prefix Digit 1 is required for 10-digit direct distance dialing (DDD) numbers, be sure the matching pattern begins with a 1.
***, x, X**	wildcard characters

Max

Valid entries	Usage
Min to **28**	Enter the maximum number of user-dialed digits the system collects to match to this Matching Pattern.

Min

Valid entries	Usage
1 to Max	Enter the minimum number of user-dialed digits the system collects to match to this Matching Pattern.

Net

Enter the call-processing server network used to analyze the converted number.

Valid entries	Usage
ext, aar, ars	Analyze the converted digit-string as an extension number, an AAR address, or an ARS address.

Percent Full

Displays the percentage (**0** to **100**) of the system's memory resources that have been used by AAR/ARS. If the figure is close to 100%, you can free-up memory resources.

Replacement String

Valid entries	Usage
0 to **9** (1 to 18 digits)	Enter the digits that replace the deleted portion of the dialed number. Leave this field blank to simply delete the digits.
*	
#	Use # to indicate end-of-dialing. It must be at the end of the digit-string.
blank	

Abbreviated Dialing List

This screen establishes system-wide or personal lists for speed dialing.

Enhanced List

The Enhanced Abbreviated Dialing List can be accessed by users to place local, long-distance, and international calls; to activate/deactivate features; or to access remote computer equipment.

Note:
Dialing must be enabled in your license file before you can program an Enhanced List. When the feature is enabled, the **Abbreviated Dialing Enhanced List** field on the **System Parameters Customer-Options (Optional Features)** screen displays **y**.

You can define two Enhanced Abbreviated Dialing Lists in the system. Before you assign numbers to a list, you must define whether you want a 3-digit or 4-digit enhanced list on the **Feature-Related System Parameters** screen. If you select 3-digit enhanced list, the list can be up to 10 separate screens numbered from 0 to 9 that allow you to define up to 1000 numbers. If you select a 4-digit enhanced list, a list can include up to 100 separate screens numbered 0 to 99 that allow you to assign up to 10,000 numbers on each list. The two Enhanced Abbreviated Dialing Lists together can support up to 20,000 entries.

If you want your attendants to use abbreviated dialing, you must also administer the **Console Parameters** screen.

Figure 5: Abbreviated Dialing Enhanced List screen

```
display abbreviated-dialing enhanced                              Page 1 of x
                                    ABBREVIATED DIALING LIST
                                        Enhanced List
                     Size (multiple of 5): 5                 Privileged? n
    DIAL CODE
      100: ________________________
      101: ________________________
      102: ________________________
      103: ________________________
      104: ________________________
      105: ________________________
```

DIAL CODE

Enter the number you want the system to dial when users enter this dial code. While the system is waiting, a call progress tone receiver is tied up, and, since there are a limited number of receivers in the system, outgoing calling capability might be impaired.

Vector Directory Number extension can also be assigned.

Valid entries	Usage
Digits **0** to **9**	Up to 24 characters
***** (star)	Part of FAC
# (pound)	Part of FAC
~p	Pause 1.5 seconds
~w	Wait for dial tone
~m	Change to outpulse DTMF digits at the end-to-end rate
~s	Start suppressing display of the digits being outpulsed
~W	Wait indefinitely for dial tone. Use this only if network response time is more than 30 seconds. Not available for S8300 Servers.

Privileged

Indicates whether users of this list can dial any number in the list, regardless of the COR of the station from which they dial.

Valid entries	Usage
y/n	Set this field to **n** if you want the system to verify that this station is allowed to dial this number.

Size (multiple of 5)

The number of dial code list entries you want in this list.

Valid entries	Usage
5 to **100**, in multiples of 5	Up to 100 entries per screen

Group List

This screen implements the Abbreviated Dialing Group List. The Group Lists are controlled by the System Administrator. Up to 100 numbers can be entered per group list that can be accessed by users to place local, long-distance, and international calls; to activate/deactivate features; or to access remote computer equipment.

Figure 6: Abbreviated Dialing Group List screen

```
change abbreviated-dialing group                                    Page 1 of X
                            ABBREVIATED DIALING LIST
                               Group List: ____
       Size (multiple of 5):   5  Program Ext: ________   Privileged? n
    DIAL CODE
         01: ________________
         02: ________________
         03: ________________
         04: ________________
         05: ________________
```

DIAL CODE

Enter the number you want the system to dial when users enter this dial code. While the system is waiting, a call progress tone receiver is tied up, and, since there are a limited number of receivers in the system, outgoing calling capability might be impaired.

Only 1 through 5 display initially. If you enter a number greater than 5 in the **Size** field, the system increases the number of dial codes to the number you specified.

Vector Directory Number extension can also be assigned.

Valid entries	Usage
Digits **0** to **9**	Up to 24 characters
* (star)	Part of FAC
# (pound)	Part of FAC
~p	Pause 1.5 seconds
~w	Wait for dial tone
~m	Change to outpulse DTMF digits at the end-to-end rate
~s	Start suppressing display of the digits being outpulsed
~W	Wait indefinitely for dial tone. Use this only if network response time is more than 30 seconds. Not available for S8300 Servers.

Group List

This is a display-only field when the screen is accessed using an administration command such as `add` or `change`.

Valid entries	Usage
Display-only field	Enter a group number when completing a paper screen.

Privileged

Valid entries	Usage
y	If **y** is entered, the calling telephone's class of restriction (COR) is never checked and any number in the group list can be dialed.
n	If **n** is entered, the calling telephone's COR is checked to determine if the number can be dialed.

Program Ext

Enter the extension that you want to give permission to program the Group List.

Size (multiple of 5)

Enter the number of abbreviated dialing numbers you want to assign in multiples of 5, up to 100.

Personal List

This screen establishes a personal dialing list for telephone/data module users. The personal list must first be assigned to the telephone by the system administrator before the telephone user can add entries in the list. The lists can be accessed by users to place local, long-distance, and international calls; to activate/deactivate features; or to access remote computer equipment.

Figure 7: Abbreviated Dialing Personal List screen

```
change abbreviated-dialing personal                                 Page 1 of x
                         ABBREVIATED DIALING LIST

                  Personal List: ________ List Number: ___
                  Size (multiple of 5): 5
DIAL CODE
     01: ________________
     02: ________________
     03: ________________
     04: ________________
     05: ________________
     06: ________________
     07: ________________
     08: ________________
     09: ________________
     00: ________________
```

DIAL CODE

Enter the number you want the system to dial when users enter this dial code. While the system is waiting, a call progress tone receiver is tied up, and, since there are a limited number of receivers in the system, outgoing calling capability might be impaired.

Only 1 through 5 display initially. If you enter a number greater than **5** in the **Size** field, the system increases the number of dial codes to the number you specified.

Note:
Although the Abbreviated Dialing Personal List screen shows dial codes with a leading zero (that is, 01, 02, 03), the user enters only the digit following the zero and not the zero itself to successfully access the extension administered on that dial code.

Vector Directory Number extension can also be assigned.

Valid entries	Usage
Digits 0 to 9	Up to 24 characters
***** (star)	Part of FAC
# (pound)	Part of FAC
~p	Pause 1.5 seconds
~w	Wait for dial tone
~m	Change to outpulse DTMF digits at the end-to-end rate

Valid entries	Usage
~s	Start suppressing display of the digits being outpulsed
~W	Wait indefinitely for dial tone. Only use this if network response time is more than 30 seconds.

List Number

A display-only field indicates which of the three personal lists is defined for the telephone.

Personal List

A display-only field indicates the extension of the telephone that uses this list.

Size (multiple of 5)

Enter the number of abbreviated dialing numbers you want to assign in multiples of 5, up to 100.

System List

This screen implements a system abbreviated-dialing list. Only one system list can be assigned and is administered by the System Administrator. The list can be accessed by users to place local, long-distance, and international calls; to activate/deactivate features; or to access remote computer equipment.

Figure 8: Abbreviated Dialing System List screen

```
add abbreviated-dialing system                              Page  1 of x
                        ABBREVIATED DIALING LIST

                                SYSTEM LIST
Size (multiple of 5): 100      Privileged? n      Label Language:english
DIAL CODE                                LABELS FOR 2420/4620 STATIONS
  11:                                 11:*************
  12:                                 12:*************
  13:                                 13:*************
  14:                                 14:*************
  15:                                 15:*************
  16:                                 16:*************
  17:                                 17:*************
  18:                                 18:*************
  19:                                 19:*************
  20:                                 20:*************
  21:                                 21:*************
  22:                                 22:*************
  23:                                 23:*************
  24:                                 24:*************
  25:                                 25:*************
```

DIAL CODE

Enter the number you want the system to dial when users enter this dial code. While the system is waiting, a call progress tone receiver is tied up, and, since there are a limited number of receivers in the system, outgoing calling capability might be impaired.

Only 1 through 5 display initially. If you enter a number greater than 5 in the **Size** field, the system increases the number of dial codes to the number you specified.

Vector Directory Number extension can also be assigned.

Valid entries	Usage
Digits 0 to 9	Up to 24 characters
***** (star)	Part of FAC
# (pound)	Part of FAC
~p	Pause 1.5 seconds
~w	Wait for dial tone
	1 of 2

Valid entries	Usage
~m	Change to outpulse DTMF digits at the end-to-end rate
~s	Start suppressing display of the digits being outpulsed
~W	Wait indefinitely for dial tone. Use this only if network response time is more than 30 seconds.
	2 of 2

Label Language

This field provides administration of personalized labels on the 2420/4620 telephone sets. If this field is changed to another language, all administered labels in the original language are saved and the labels for the new language are read in and displayed.

Valid entries	Usage
English **Italian** **French** **Spanish** **user-defined** **Unicode**	Enter the appropriate language for the 2420/4620 labels. **Note:** Unicode labels are only available for Unicode-supported telephones. Currently, the 4610SW, 4620SW, 4621SW, and 4622SW, Sage, Spark, and 9600-series Spice telephones support Unicode display. Unicode is also an option for the 2420J telephone when **Display Character Set** on the **System Parameters Country-Options** screen is **katakana**. For more information on the 2420J, see *2420 Digital Telephone User's Guide*, 555-250-701. Unicode labels are entered through the Avaya Site Administration (ASA).

LABELS FOR 2420/4620 STATIONS

This field provides the administrative capability to actually customize the labels for the system-wide Abbreviated Dial buttons on the 2420/4620 telephone sets.

Valid entries	Usage
A-Z, a-z, 0-9, and ! & * ? ; ’ ^ ( ) , . : -	Up to 15 alphanumeric characters

Privileged

Valid entries	Usage
y	Enter **y** if the originating party's class of restriction (COR) is never checked and any number in the list can be dialed.
n	Enter **n** if the COR is to be checked to determine if the number can be dialed.

Size (multiple of 5)

Enter the number of abbreviated dialing numbers you want to assign in multiples of 5, up to 100.

The Figure 9 shows the last page of the Abbreviated Dialing System screen when, on the System Parameters Customer-Options (Optional Features) screen, the **A/D Grp/Sys List Dialing Start at 01** field is **n**.

Figure 9: Abbreviated Dialing System List screen

```
add abbreviated-dialing system                               Page  7 of x
                         ABBREVIATED DIALING LIST

                                 SYSTEM LIST
                                        Label Language:english
DIAL CODE                                 LABELS FOR 2420/4620 STATIONS
  01:                                  01:*************
  02:                                  02:*************
  03:                                  03:*************
  04:                                  04:*************
  05:                                  05:*************
  06:                                  06:*************
  07:                                  07:*************
  08:                                  08:*************
  09:                                  09:*************
  10:                                  10:*************
```

Figure 10 shows the last page of the Abbreviated Dialing System screen when, on the System Parameters Customer-Options (Optional Features) screen, the **A/D Grp/Sys List Dialing Start at 01** field is **y**.

Figure 10: Abbreviated Dialing System List screen

```
add abbreviated-dialing system                              Page  7 of x
                         ABBREVIATED DIALING LIST

                                 SYSTEM LIST
                                        Label Language:english
DIAL CODE                                 LABELS FOR 2420/4620 STATIONS
  91:                                  91:*************
  92:                                  92:*************
  93:                                  93:*************
  94:                                  94:*************
  95:                                  95:*************
  96:                                  96:*************
  97:                                  97:*************
  98:                                  98:*************
  99:                                  99:*************
  00:                                  00:*************
```

7103A Button List

This screen assigns abbreviated dialing numbers to the 7103A telephone buttons. The entries can then be accessed by 7103A telephone users to place local, long-distance, and international calls; activate/deactivate features; or to access remote computer equipment. This screen applies only to 7103A fixed feature telephones. Only one 7103A abbreviated dialing list can be implemented in the system and it applies to all 7103A fixed feature telephones in the system. This list is controlled by the System Administrator.

Figure 11: Abbreviated Dialing List — 7103A Button List screen

```
display abbreviated-dialing 7103A-buttons                Page 1 of x
                           ABBREVIATED DIALING LIST
                              7103A Button List

      DIAL CODE (FOR THE 7103A STATION BUTTONS)
       1: ________________________       5. ________________________
       2: ________________________       6. ________________________
       3: ________________________       7. ________________________
       4: ________________________       8. ________________________
```

DIAL CODE

Enter the number you want to assign to each dial code (button). Any additions or changes apply to all 7103A fixed feature telephones. While the system is waiting, a call progress tone receiver is tied up, and, since there are a limited number of receivers in the system, outgoing calling capability might be impaired.

Vector Directory Number extension can also be assigned.

Valid entries	Usage
Digits 0 to 9	Up to 24 characters
***** (star)	Part of FAC
# (pound)	Part of FAC
~p	Pause 1.5 seconds
~w	Wait for dial tone
~m	Mark
~s	Start suppressing display of the digits being outpulsed.
~W	Wait indefinitely for dial tone. Use this only if network response time is more than 30 seconds. Not available for S8300 Servers.

Access Endpoint

This screen administers Access Endpoints and Wideband Access endpoints.

Note:
You can administer Wideband Access Endpoints only if, on the System Parameters Customer-Options (Optional Features) screen, the **Wideband Switching** field is **y**.

An Access Endpoint is a nonsignaling trunk that neither responds to signaling nor generates signaling. Access Endpoints eliminate the need to dedicate an entire trunk group for the access of a single trunk by providing the capability to assign an extension number to a single trunk.

An Access Endpoint can be specified as the Originator or Destination endpoint of an administered connection.

A Wideband Access Endpoint (WAE) is an endpoint application connected to line-side non-ISDN T1 or E1 facilities and, like Access Endpoints, have no signaling interface with the system.

The WAE is defined by a starting port (DS0) and a width specifying the number of adjacent nonsignaling DS0s (positioned within a DS1 facility) that make up the endpoint. This width can be between 2 and 31 adjacent DS0s.

Note:
Access Endpoints and Wideband Access Endpoints consume the same resources that trunks use. Thus, the sum of Access Endpoints and trunks cannot exceed the maximum number of trunks available in your system configuration.

Field descriptions for page 1

Figure 12: Access Endpoint screen

```
add access-endpoint next                                        Page   1 of   x
                              ACCESS ENDPOINT

            Extension: 30001      (Starting) Port:_______
   Communication Type: voice-grade-data      Name:_____________

                  COR: 1                      COS: 1
                   TN: 1                      ITC: restricted
```

Communication Type

Valid entries	Usage
voice-grade-data	For an analog tie trunk access endpoint.
56k-data	For a DS1 access endpoint enter as appropriate (**64k-data** is not allowed for robbed-bit trunks).
64k-data	
wideband	For a Wideband access endpoint

COR

The COR is administered so that only an administered connection (AC) endpoint can be connected to another AC endpoint.

Valid entries	Usage
0 to **995**	Enter the appropriate class of restriction (COR) number.

COS

The COS is administered (see Class of Service on page 131) so that the use of the Call Forwarding All Calls feature for access endpoints is prohibited.

Valid entries	Usage
0 to **15**	Enter the appropriate COS number.

Extension

A display-only field showing the extension number as specified in the command line, or shows the next available extension number if **next** was entered on the command line. This is the extension number assigned to the nonsignaling trunk and used to access the trunk endpoint.

ITC (Information Transfer Capability)

This field is used to determine the type of transmission facilities to be used for ISDN calls originating from this endpoint. Displays when the **Communication Type** field is **56k-data**, **64k-data**, or **Wideband**.

When adding an access endpoint with the ITC administered as unrestricted, its associated port has to be a channel of a DS1 circuit pack with the **Zero Code Suppression** field administered as B8ZS. If the port is not a channel of a DS1 circuit pack with its **Zero Code Suppression** field administered as B8ZS, the end validation fails and the screen submission is rejected. The cursor is moved to ITC with the following error message:

An unrestricted access endpoint can only be from B8ZS DS1 circuit pack.

When adding an access endpoint with the ITC administered as restricted, its associated port can be a channel from a DS1 circuit pack with the **Zero Code Suppression** field administered as ZCS or B8ZS.

For an existing access endpoint, ITC can only be changed from restricted to unrestricted if its associated port is a channel of a DS1 circuit pack with its **Zero Code Suppression** field administered as B8ZS. If the port is not a channel of a DS1 circuit pack with its **Zero Code Suppression** field administered as B8ZS, the end validation fails and the screen submission is rejected. The cursor is moved to ITC with the following error message:

```
An unrestricted access endpoint can use only B8ZS DS1 circuit pack
```

Without this end validation, a user could administer an access endpoint as unrestricted when in fact it is restricted, that is, its associated port is a member of a DS1 circuit pack that uses ZCS data transmission.

Valid entries	Usage
unrestricted	When **unrestricted**, only unrestricted transmission facilities (b8zs) is used to complete the call. An unrestricted facility is a transmission facility that does not enforce 1's density digital transmission (that is, digital information is sent exactly as is). For Wideband Access Endpoints, enter **unrestricted**.
restricted	When **restricted**, either restricted (zcs-ami) or unrestricted transmission facilities is used to complete the call. A restricted facility is a transmission facility that enforces 1's density digital transmission (that is, a sequence of eight digital zeros is converted to a sequence of seven zeros and a digital one) via zcs coding on DS1 circuit pack.

Name

Enter an name for the endpoint.

Note:
BRI stations support ASCII characters only. They do not support non-ASCII characters, such as Eurofont or Kanafont. Therefore, if you use non-ASCII characters in any Communication Manager **Name** field, such characters do not display correctly on a BRI station.

(Starting) Port

Enter the necessary characters.

Valid entries	Usage
01 to **03** (DEFINITY CSI) or **1** to **64** (S87XX/S8300 Servers)	First and second characters are the cabinet number.
A to **E**	Third character is the carrier.
0 to **20**	Fourth and fifth characters are the slot number.

Valid entries	Usage
01 to **04** (Analog TIE trunks) **01** to **31**	Six and seventh characters are the circuit number.
1 to **80** (DEFINITY CSI) or **1** to **250** (S87XX/S8300 Servers)	Gateway
V1 to **V9**	Module
01 to **31**	Circuit

For example, 01A0612 is in cabinet 01, carrier A, slot 06, and circuit number (port) 12.

Note:
For Wideband Access Endpoints, analog tie trunks cannot be used and the DS1 Interface circuit pack, Version C or later, must be used.

The DS1 circuit number corresponds to the channel that carries the data traffic. Channels 1 through 31 (DS1 Interface only) or channels 1 through 24 (DS1 Tie Trunk, DS1 Interface, or DS1 Interface (32) circuit packs) can be used when the **DS1 Signaling Type** field is **robbed-bit** or **isdn-ext**. For Common Channel or ISDN-PRI signaling, channel use is limited to channels 1 through 30 (DS1 Interface circuit pack only) or channels 1 through 23 (DS1 Interface (32) or DS1 Interface). A channel can be administered as an access endpoint regardless of the DS1 signaling type.

TN

Valid entries	Usage
1 to 100	Enter the Tenant Partition number.

Width

Appears if the **Communication Type** field is **wideband**. This field cannot be blank.

Valid entries	Usage
2 to **31**	Enter the number of adjacent DS0 ports beginning with the specified Starting Port, that make up the WAE.
6	A width of **6** defines a 384 Kbps WAE.

Administered Connection

This screen assigns an end-to-end Administered Connection (AC) between two access endpoints or data endpoints. The AC is established automatically by the system whenever the system restarts or the AC is due to be active. See Administered Connections in *Avaya Aura™ Communication Manager Feature Description and Implementation*, 555-245-205, and Access Endpoint on page 24 for additional information.

Field descriptions for page 1

Figure 13: Administered Connection screen

```
change administered-connection                                 Page 1 of x
                         ADMINISTERED CONNECTION
  Connection Number:  1                        Enable? y
         Originator:  ________
        Destination:  __________________________________
               Name:  __________________

AUTHORIZED TIME OF DAY

         Continuous? n
                     Sun? n Mon? n Tue? n Wed? n Thu? n  Fri? n  Sat? n
         Start Time:  00:00
           Duration:  000:00

  MISCELLANEOUS PARAMETERS

                 Alarm Type:  warning  Alarm Threshold: 5
                                        Retry Interval: 2
                   Priority:  5      Auto Restoration? y
```

Connection Number

This is a display-only field showing an unassigned AC number when the screen is accessed using an administration command such as **change** or **display**.

Destination

Used to route the AC to a desired endpoint. Enter the address of the destination access or data endpoint. This endpoint is the terminating party of the AC and need not be local to the server on which the AC is assigned. The entry must be consistent with the local Communication Manager server's dial plan (that is, the first digits are assigned as an extension, feature access code, or trunk access code, or DDD Number). If a local extension is entered, it must be assigned to either an access or data endpoint. Abbreviated Dialing entries can be used in this field.

Valid entries	Usage
Extension/string	Enter the assigned access endpoint/data module extension or valid dialed string.

Enable

Provides the administered connection.

Valid entries	Usage
y	Indicates an attempt is made to establish the AC when the AC is due to be active.
n	The AC is not made or if it is up, it drops.

Name

Valid entries	Usage
Up to 27 alphanumeric characters. Up to 15 alphanumeric characters (S8300 Server, S87XX IP-PNC only)	Enter a short identification of the AC. **NOTE:** BRI stations support ASCII characters only. They do not support non-ASCII characters, such as Eurofont or Kanafont. Therefore, if you use non-ASCII characters in any Communication Manager **Name** field, such characters do not display correctly on a BRI station.

Originator

Enter the assigned access endpoint/data module extension.

Data Line circuit pack

- Asynchronous EIA 232C compatible equipment

Digital Line circuit pack connections, including:

- MPDM (700D), MTDM (700B, 700C, 700E), 7400D data module
- 7400A, 7400B, 7400C HSL, 8400B data module
- 7401D telephone with 7400B or 8400B data module
- 7403D/7405D/7407D/7410D/7434D telephone with DTDM or 7400B or 8400B data module
- 7404D or 7406D telephone
- 510D personal terminal
- 515 BCT, 615 BCT, or 715 BCT terminal
- Connection between PC and the server running Communication Manager

ISDN-BRI Line circuit pack connections, including:

- 7500 data module
- 7505D/7506D/7507D telephone with ADM

Valid entries	Usage
Assigned access endpoint/ data module extension	The endpoint must be local to the server on which the AC is administered. Nonsignaling DS1 trunk or analog tie trunk.

Authorized Time of Day

Continuous

The connection is up all the time or re-established if the connection goes down.

Valid entries	Usage
y	Indicates that the AC is continuous (that is, not scheduled to be active at a certain time). If **y** is entered, the seven **Start Days** and associated **Duration** fields do not appear.
n	Displays the **Start Days** fields.

Duration

Enter the period of time that the scheduled AC should remain active. This period is specified in two fields separated by a colon. The maximum duration is 167 hours and 59 minutes (that is, 1 minute less than 1 week). Only appears if the **Continuous** field is **n**.

Valid entries	Usage
000 through **167**	For the **hour** field.
00 through **59**	For the **minute** field.

Start Days (Sun through Sat)

These fields indicate only the days on which an attempt is made to establish the AC and not necessarily the days it is active. A scheduled AC might be active over a number of days, and, in this situation, these fields should be used only to specify the days on which the AC starts and not other days on which the AC might be active. Only appears if the **Continuous** field is **n**.

Valid entries	Usage
y	Enter **y** in each of the required days of the week fields to indicate that an attempt is made to establish the AC.
n	Displays the day fields.

Start Time

Only appears if the **Continuous** field is **n**.

Valid entries	Usage
00:00 through **23:59**	Enter the time of the day when an attempt should begin to establish a scheduled AC. The time is specified in two fields separated by a colon.

Miscellaneous Parameters

Alarm Threshold

Only appears if an entry in the **Alarm Type** field is other than **none**. Enter the number of times an attempt to establish or reestablish an AC must fail consecutively before an AC alarm generates. (An alarm is generated after the fourth retry has failed; thus, with the retry interval of 2 minutes, an alarm is generated approximately 8 minutes after the first failure occurs.)

Valid entries	Usage
1 through **10**	An alarm generates on the first failure if this field is **1**.

Alarm Type

Enter the type of alarm to be generated if the AC cannot be initially established, or fails and cannot be reestablished, and the number of consecutive failures equals the alarm threshold. All AC alarms and the errors that caused the alarms are recorded in the system's alarm and error log. In addition, a status lamp associated with an attendant console or telephone feature button (**ac-alarm**) can be used to indicate the AC alarm.

Valid entries	Usage
major	Failures that cause critical degradation of service and require immediate attention.
minor	Failures that cause some degradation of service, but do not render a crucial portion of the system inoperable. This condition requires action, but its consequences are not immediate. Problems might be impairing service to a few trunks or stations or interfering with one feature across the entire system.
warning	Failures that cause no significant degradation of service or failures in equipment external to the system. Warning alarms are not reported to the attendant console or INADS.
none	The alarm notification is disabled for this AC.

Auto Restoration

Valid entries	Usage
y	Enter **y** to indicate an attempt is to be made to reestablish an AC that failed. Auto restoration is available only for an AC that is established over an ISDN Software Defined Data Network (SDDN) trunk group. A **y** in this field is ignored in all other situations.

Priority

Enter a number that is to be used to determine the order in which ACs are to be established.

Valid entries	Usage
1 to **8**	1 is the highest and 8 the lowest priority.

Retry Interval

Valid entries	Usage
1 to **60**	Enter the number of minutes between attempts to establish or reestablish the AC.

Agent LoginID

Use this screen in an Expert Agent Selection (EAS) environment to add or change agent login IDs and skill assignments. If you add or change skills on the Avaya S8XXX Server, the agent must log out and then log in again before the changes take effect. Note that in non-EAS (basic Automatic Call Distribution) environments, this screen does not appear at all, and agents are assigned directly on the Hunt Group screen. The agent's properties are assigned to the physical telephone extension. For more information, see *Avaya Aura™ Call Center 5.2 Automatic Call Distribution (ACD) Reference*, 07-602568.

Field descriptions for page 1

Figure 14: Agent LoginID screen

```
add agent-loginID 9011                                          Page   1 of   x
                                 AGENT LOGINID

                Login ID: 9011                                  AAS? n
                    Name:                                     AUDIX? n
                      TN: 1                          LWC Reception: spe
                     COR: 1                 LWC Log External Calls? n
           Coverage Path:                 AUDIX Name for Messaging:
           Security Code:
                                      LoginID for ISDN/SIP Display? n
                                                          Password:
                                            Password (enter again):
                                                       Auto Answer: station
                                                 MIA Across Skills: system
                                         ACW Agent Considered Idle: system
                                         Aux Work Reason Code Type: system
                                           Logout Reason Code Type: system
                       Maximum time agent in ACW before logout (sec): system
                                           Forced Agent Logout Time:    :

     WARNING:  Agent must log in again before changes take effect
```

AAS

Enter **y** if this extension is used as a port for an Auto Available Split/Skill. Default is **n**.

Entering **y** in the **AAS** field clears the password and requires execution of the **`remove agent-loginid`** command. To set AAS to **n**, remove this logical agent and add it again. This option is intended for switch adjunct equipment ports only, not human agents.

ACW Agent Considered Idle

Enter **y** to have agents who are in After Call Work included in the Most-Idle Agent queue. This means that ACW is counted as idle time. Enter **n** to exclude ACW agents from the queue. Valid entries are **system** (default), **n**, and **y**. The **system** value indicates that settings assigned on the Feature-Related System Parameters screen apply.

Audix

Enter **y** if this extension is used as a port for AUDIX. Default is **n**. The **AAS** and **AUDIX** fields cannot both be **y**.

Audix Name for Messaging

Do one of the following actions:

- Enter the name of the messaging system used for LWC Reception, or
- Enter the name of the messaging system that provides coverage for this Agent LoginID.

Auto Answer

When using EAS, the agent's auto answer setting applies to the station where the agent logs in. If the auto answer setting for that station is different, the agent's setting overrides the station's setting. The following entries are valid:

- **all** - immediately sends all ACD and non ACD calls to the agent. The station is also given a single ring while a non-ACD call is connected. The **ringer-off** button can be used to prevent the ring when, on the Feature-Related System Parameters screen, the **Allow Ringer-off with Auto-Answer** field is set to **y**.
- **acd** - only ACD split /skill calls and direct agent calls go to auto answer. If this field is **acd**, non ACD calls terminated to the agent ring audibly.
- **none** - all calls terminated to this agent receive an audible ringing treatment. This is the default.
- **station** - auto answer for the agent is controlled by the auto answer field on the Station screen.

Aux Work Reason Code Type

Valid entries	Usage
system	Settings assigned on the Feature-Related System Parameters screen apply. This is the default.
none	Enter **none** if you do not want an agent to enter a Reason Code when entering AUX work.
requested	Enter **requested** if you want an agent to enter a Reason Code when entering AUX mode but do not want to force the agent to do so. To enter **requested** the **Reason Codes** and **EAS** on the System Parameters Customer-Options (Optional Features) screen must be **y**.
forced	Enter **forced** to force an agent to enter a Reason Code when entering AUX mode. To enter **forced**, the **Reason Codes** and **EAS** on the System Parameters Customer-Options (Optional Features) screen must be **y**.

COR

Enter the Class of Restriction for the agent. Valid entries are **0** to **995**. Default is **1**.

Coverage Path

Enter the number of the coverage path used by calls to the LoginID. Valid entries are a path number between **1** and **999**, time of day table **t1** to **t999**, or blank (default). This is used when the agent is logged out, does not answer, or is busy to personal calls when logged in.

Direct Agents Calls First

This field replaces the **Service Objective** field when percent-allocation is entered in the **Call Handling Preference** field. Enter **y** if you want direct agent calls to override the percent-allocation call selection method and be delivered before other ACD calls. Enter **n** if you want direct agent calls to be treated like other ACD calls. For more information, see the *Avaya Business Advocate User Guide*, 07-300653.

Forced Agent Logout Time

This field enables the Forced Agent Logout by Clock Time feature by administering a time of day to automatically log out agents using an hour and minute field. Valid entries for the hour field are **01-23**. Valid entries for the minute field are **00**, **15**, **30**, and **45**. The default is blank (not administered). Examples: 15:00, 18:15, 20:30, 23:45.

Login ID

Display-only field. Contains the identifier for the Logical Agent as entered on the command line.

LoginID for ISDN Display

Enter **y** if the Agent LoginID CPN (Calling Party Number) and Name field is to be included in ISDN messaging over network facilities. If set to **n** (the default), the physical station extension CPN and Name is sent. The **Send Name** field on the ISDN Trunk Group screen prevents sending out the calling party name and number if set to **n**, and may prevent sending it if set to **r** (restricted).

Logout Reason Code Type

Valid entries	Usage
system	Settings assigned on the Feature-Related System Parameters screen apply.This is the default.
none	Enter **none** if you do not want an agent to enter a Reason Code when logging out.
requested	Enter **requested** if you want an agent to enter a Reason Code when logging out but do not want to force the agent to do so. To enter **requested** the **Reason Codes** and **EAS** on the System Parameters Customer-Options (Optional Features) screen must be **y**.
forced	Enter **forced** to force an agent to enter a Reason Code when logging out. Enter **forced** to force an agent to enter a Reason Code when entering AUX mode. To enter **forced**, the **Reason Codes** and **EAS** on the System Parameters Customer-Options (Optional Features) screen must be **y.**

LWC Reception

Enter the name of the messaging system where Leave Word Calling messages for this Agent Login ID is stored. Valid entries are **audix**, **msa**, **spe** (default), and **none**.

Maximum time agent in ACW before logout (sec)

This field is used for setting a maximum time the agent can be in ACW on a per agent basis. Valid entries are:

- **system** - Settings assigned on the Feature-Related System Parameters screen apply. This is the default.
- **none** - ACW timeout does not apply to this agent.
- **30-9999** sec - Indicates a specific timeout period. This setting takes precedence over the system setting for maximum time in ACW.

Messaging Server Name for Messaging

Do one of the following actions:

- Enter the name of the Messaging Server used for LWC Reception.
- Enter the name of the Messaging Server that provides coverage for this Agent LoginID.
- Leave blank (default).

MIA Across Skills

Enter **y** to remove an agent from the MIA queues for all the splits or skills that the agent is available in when the agent answers a call from any of the assigned splits or skills. Enter **n** to exclude ACW agents for the queue.Valid entries are **system** (default), **n**, and **y**. The **system** value indicates that settings assigned on the Feature-Related System Parameters screen apply.

Name

Enter up to a 27-character string naming the agent. Any alpha-numeric character is valid. Default is blank.

Note:
For 4610SW, 4620SW, 4621SW, and 4622SW, Sage, Spark, and 9600-series Spice telephones, the **Name** field has an associated optional native name field that is supported by the Unicode language display. The native name field is accessible through the Integrated Management Edit Tools such as Avaya Site Administration (ASA). Unicode is also an option for the 2420J telephone when **Display Character Set** on the **System Parameters Country-Options** screen is **katakana**. For more information on the 2420J, see *2420 Digital Telephone User's Guide*, 555-250-701.

Avaya BRI stations support ASCII characters only. They do not support non-ASCII characters, such as Eurofont or Kanafont. Therefore, if you use non-ASCII characters in any Communication Manager **Name** field, such characters do not display correctly on a BRI station.

Password

Appears only if both the **AAS** and **AUDIX** fields are **n**. Enter up to nine digits as the password the Agent must enter upon login. Valid entries are the digits **0** through **9**. Enter the minimum number of digits in this field specified by the **Minimum Agent-LoginID Password Length** field on the Feature-Related System Parameters screen. Default is blank. Values entered in this field are not displayed on the screen.

Password (enter again)

Appears only if both the **AAS** and **AUDIX** fields are n. Reenter the same password exactly as it was entered in the **Password** field. Default is blank. Values entered in this field are not displayed on the screen.

Port Extension

Appears only if either the **AAS** or **AUDIX** field is **y**. Enter the assigned extension for the AAS or AUDIX port. This extension cannot be a VDN or an Agent LoginID. Default is blank.

Security Code

Enter the 4-digit security code (password) for the Demand Print messages feature. This field can be blank (default).

TN

Enter the partition number for tenant partitioning. Valid entries are **1** to **20**. Default is **1**.

Field descriptions for page 2

The second page of the Agent LoginID screen contains agent skill information.

Figure 15: Agent Login ID screen

```
add agent-loginID 9011                                            Page 2 of X
                                   AGENT LOGINID
      Direct Agent Skill:                                      Service Objective?
Call Handling Preference:                                  Local Call Preference?


    SN RL SL PA            SN RL SL PA            SN RL SL PA            SN RL SL PA
 1: __ _  __ ___      16:  __ _  __ ___     31:  __ _  __ ___     46:  __ _  __ ___
 2: __ _  __ ___      17:  __ _  __ ___     32:  __ _  __ ___     47:  __ _  __ ___
 3: __ _  __ ___      18:  __ _  __ ___     33:  __ _  __ ___     48:  __ _  __ ___
 4: __ _  __ ___      19:  __ _  __ ___     34:  __ _  __ ___     49:  __ _  __ ___
 5. __ _  __ ___      20:  __ _  __ ___     35:  __ _  __ ___     50:  __ _  __ ___
 6: __ _  __ ___      21:  __ _  __ ___     36:  __ _  __ ___     51:  __ _  __ ___
 7: __ _  __ ___      22:  __ _  __ ___     37:  __ _  __ ___     52:  __ _  __ ___
 8: __ _  __ ___      23:  __ _  __ ___     38:  __ _  __ ___     53:  __ _  __ ___
 9: __ _  __ ___      24:  __ _  __ ___     39:  __ _  __ ___     54:  __ _  __ ___
10. __ _  __ ___      25:  __ _  __ ___     40:  __ _  __ ___     55:  __ _  __ ___
11: __ _  __ ___      26:  __ _  __ ___     41:  __ _  __ ___     56:  __ _  __ ___
12: __ _  __ ___      27:  __ _  __ ___     42:  __ _  __ ___     57:  __ _  __ ___
13: __ _  __ ___      28:  __ _  __ ___     43:  __ _  __ ___     58:  __ _  __ ___
14: __ _  __ ___      29:  __ _  __ ___     44:  __ _  __ ___     59:  __ _  __ ___
15. __ _  __ ___      30:  __ _  __ ___     45:  __ _  __ ___     60:  __ _  __ ___
```

Call Handling Preference

When calls are in queue and an agent becomes available, the **skill-level** setting delivers the highest priority, oldest call waiting for the agent's highest level skill. Other choices are **greatest-need** and **percent-allocation**. **Greatest-need** delivers the oldest, highest priority call waiting for any of the agent's skills. **Percent allocation** delivers a call from the skill that otherwise deviate most from its administered allocation. **Percent-allocation** is available only with Avaya Business Advocate software. For more information, see the *Avaya Business Advocate User Guide*, 07-300653.

Direct Agent Skill

Enter the number of the skill used to handle Direct Agent calls. Valid entries are **1** to **99**, or blank (default).

Local Call Preference

Enter **y** to indicate that for calls queued in more than one skill for a multi-skilled EAS agent, the system should give preference to matching the trunk location number of the queued call to the location number of the previously-busy agent. Valid settings are **n** (default) or **y**. You can only set this field to **y** if the **Call Center Release** field on the Feature-Related System Parameters screen is **3.0** or later, and the **Multiple Locations** field on the **System Parameters Customer-Options (Optional Features)** screen is set to **y**.

PA

Percent Allocation. If the call handling preference is **percent-allocation**, you must enter a percentage for each of the agent's skills. Enter a number between 1 and 100 for each skill. Your entries for all of the agent's skills together must total 100%. Do not use target allocations for reserve skills. Percent Allocation is available as part of the Advocate software.

RL (Reserve Level)

Enter the reserve level to assign to the agent for the skill with the Business Advocate Service Level Supervisor feature or the type of interruption with the Interruptible AUX Work feature. You can assign a reserve level of **1**or **2**, or an interruptible level of auto-in-interrupt (**a**), manual-in-interrupt (**m**), or notify-interrupt (**n**) or blank for no reserve or interruptible level. Changes to this field take effect the next time the agent logs in.

You can enter the reserve levels of 1 and 2 only if Business Advocate feature is enabled. **RL** set to 1 or 2 defines the EWT threshold level for the agent is added to the assigned skill as a reserve agent. When the EWT for the skill reaches the corresponding threshold set on the Hunt Group screen, automatically the assigned skill gets added to the agent logged in skills. The agent delivers calls from this skill until the corresponding threshold drops below the assigned overload threshold for that level.

The Interruptible AUX Work feature is a way to help meet service level targets by requesting agents who are on break to become available when the service level target is not being met.

For more information on Service Level Supervisor, see the *Avaya Business Advocate User Guide*.

For more information on Interruptible AUX Work, see the *Avaya Aura™ Call Center 5.2 Call Vectoring and Expert Agent selection (EAS) Reference*, 07-600780.

Service Objective

Appears only when **Call Handling Preference** is **greatest-need** or **skill-level**. Valid entries are **y** or **n**. Service Objective is administered on the Hunt Group screen and the Agent LoginID screen. The server selects calls for agents according to the ratio of Predicted Wait Time (PWT) or Current Wait Time (CWT) and the administered service objective for the skill. Service Objective is a feature that is part of the Advocate software.

SL

Skill Level. Enter a skill level for each of an agent's assigned skills. If EAS-PHD is not optioned, 2priority levels are available. If EAS-PHD is optioned, 16 priority levels are available. In releases prior to R3V5, level 1 was the primary skill and level 2 was the secondary skill.

SN

Skill Number. Enter the Skill Hunt Group(s) that this agent handles. The same skill cannot be entered twice. Consider the following options:

- If EAS-PHD is not optioned, enter up to four skills.
- If EAS-PHD is optioned, enter up to 20 or 60 skills depending on the platform.
- Assigning a large number of skills to agents can potentially impact system performance. Review system designs with the ATAC when a significant number of agents have greater than 20 skills per agent.

Alias Station

This screen allows you to configure the system so that you can administer new telephone types that are not supported by your system software. This screen maps new telephone models to a supported telephone model. This mapping does not guarantee compatibility, but allows unsupported models to be administered and tracked by their own names.

Some administrators also use this screen to "name" non-telephone devices. For example, you know that you can add a modem to your system by simply administering the extension as the standard analog type 2500. But, if you listed your stations, how would you know which extensions are modems? Instead, you could use the Alias screen to create a ‘modem’ alias to type 2500 and enter modem in the **Type** field for every modem you add to your system.

Tip:
When you upgrade a system that uses an alias set type to a new release, the system determines if the aliased type is supported in the new release (is now a native set type). When you review the Alias Station screen, you might see alias types that have become native. If the type is now native, the last character of the aliased set type becomes a "#."

Field descriptions for page 1

Figure 16: Alias Station screen

```
change alias station                                             Page 1 of x
                                 ALIAS STATION
                        Alias Set Type    Supported Set Type

                             ______             ______
                             ______             ______
                             ______             ______
                             ______             ______
                             ______             ______
                             ______             ______
                             ______             ______
                             ______             ______
                             ______             ______
                             ______             ______
               '#' indicates previously aliased set type is now native
```

Alias Set Type

Enter up to a 5-character name for the non-supported telephone type that you want to alias to a similar supported telephone type. Do not use blank characters.

Supported Set Type

Enter a supported telephone type that you want to map (or alias) to the alias set type. Valid supported telephone types are listed in Telephones on page 840.

Note:
Data Communication Protocol (DCP) telephone types must be aliased to DCP telephone types, hybrid types to hybrid types, and analog to analog types.

Alphanumeric Dialing Table

This screen associates alpha-names to dialed digit strings. This allows telephone users to place a data call by simply typing the alpha-name. Users need only remember far-end alpha-names instead of the actual digit strings.

The screen consists of paired **Alpha-name/Mapped String** fields. Entries can be made in any order on the screen. However, before the screen is displayed for changing or reviewing, the entries in the table are sorted alphanumerically by the alpha-name. All entries are moved to the beginning of the table, leaving all blank entries at the end.

Field descriptions for page 1

Figure 17: Alphanumeric Dial Table screen

```
change alphanumeric-dial-table                                 Page 1 of x

                        ALPHANUMERIC DIALING TABLE
  XXX of XXX administered
  Alpha-name        Mapped String          |  Alpha-name         Mapped String
                                           |
  ____________  ______________________     |  _____________  ____________________
  ____________  ______________________     |  _____________  ____________________
  ____________  ______________________     |  _____________  ____________________
  ____________  ______________________     |  _____________  ____________________
  ____________  ______________________     |  _____________  ____________________
  ____________  ______________________     |  _____________  ____________________
  ____________  ______________________     |  _____________  ____________________
  ____________  ______________________     |  _____________  ____________________
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  ____________  ______________________     |  _____________  ____________________
  ____________  ______________________     |  _____________  ____________________
  ____________  ______________________     |  _____________  ____________________
  ____________  ______________________     |  _____________  ____________________
  ____________  ______________________     |  _____________  ____________________
  ____________  ______________________     |  _____________  ____________________
  ____________  ______________________     |  _____________  ____________________
  ____________  ______________________     |  _____________  ______________________
```

Alpha-name

All alpha-names in the table must be unique and cannot be referenced in their own **Mapped String**. The alpha-names can be used in any other **Mapped String** and multiple times in a **Mapped String**.

Valid entries	Usage
From 1 to 8 alphanumeric characters	The entry must start with an alphabetic character and cannot have blank spaces between characters.

Mapped String

Enter from 1 to 24 characters that might contain alphanumeric, readability, delimiters, and/or special characters. The entry is used to generate the final dialing string and can include Facility Access Codes.

Note:
A **Mapped String** cannot contain an Alpha-Name whose Mapped String also contains an Alpha-Name.

Valid entries	Usage
Digits 0 to 9	Numeric
A through **Z**, **a** through **z**	Alpha (note uppercase entries are mapped to lowercase)
(	Readability character
)	Readability character
/	Readability character
-	Readability character
+	Wait for dial tone
%	Rest of digits are for end to end signaling
"**,**"	Pause for 1.5 seconds
space	Readability character
#	DTMF digit pound
*****	DTMF digit asterisk
^	Readability character

Announcements/Audio Sources

Use this screen to assign announcements to circuit packs and port locations.

Field descriptions for page 1

Figure 18: Announcements/Audio Sources screen

```
add announcement 26451                                              Page 1 of X
                          ANNOUNCEMENTS/AUDIO SOURCES

 Extension: 26451                                 COR: 1
 Annc Name: collect_some_digits                    TN: 1
 Annc Type: integrated                          Queue? y
Group/Port:                              Queue Length?
 Protected? n                                    Rate: 64
```

Annc Name

Valid entries	Usage
up to 27-character alpha-numeric filename	Enter the name of the announcement you are associating with the specified extension.
(no ., /, :, *, ?, <, >, \, .wav, or blanks in this field for VAL circuit packs only)	For VAL announcements, this field is required. The value in this field becomes the filename of the announcement. The **.wav** file extension, which is part of the filename stored on the circuit pack, does not appear. Do not enter **.wav** as part of the filename. Names on a single VAL circuit pack must be unique. The system checks for duplicate filenames on the same VAL circuit pack. **NOTE:** Avaya BRI stations support ASCII characters only. They do not support non-ASCII characters, such as Eurofont or Kanafont. Therefore, if you use non-ASCII characters in any Communication Manager **Name** field, such characters do not display correctly on a BRI station.

Annc Type

Enter the type of announcement you want to assign to this extension number.

If you enter **integrated** or **integ-rep**, complete the **Queue**, **Protected**, **Rate**, and **Port** fields. If you enter **analog**, **ds1-fd**, **ds1-sa**, **ds1-ops**, or **aux-trunk**, complete **Queue Length** (if **Q** is **y**) and **Port**.

Valid entries	Usage
analog	Use to play announcements from an external device for a specific period and hang up when finished. When the device hangs up, the caller hears a click. Connects to the server running Communication Manager through an analog port. Ringing starts playback.
analog-m	Use for continuous playing music or audio source from an external announcement device.
analog-fd	Use to play announcements from an external device for a specific period and hang up when finished. When the device hangs up, the caller hears a click. Connects to the server running Communication Manager through an analog port. Ringing starts playback. Sends forward disconnect signal to stop playback.
aux-trunk	Auxiliary trunk. Use with an external announcement device with a 4-wire "aux" interface.
aux-trk-m	Auxiliary trunk. Use with continuously playing music or audio sources that do not indicate playback is active.
ds1-fd	Assigned to DS1 ports on circuit packs. Callers do not hear a click when the device hangs up. Provides a disconnect to stop playback when the announcement is done.
ds1-ops	Callers do not hear a click when the device hangs up.
ds1-sa	Provides a disconnect to stop playback when the announcement is done. Callers do not hear a click when the device hangs up.
integrated	Stored internally on the Avaya DEFINITY or Avaya S8XXX Server on a special integrated announcement circuit pack. Use for general announcements and VDN of Origin Announcements.
integ-mus	Integrated music source.
integ-rep	Integrated repeating

COR

Valid entries	Usage
0 to **995**	Enter the class of restriction (COR) you want associated with this announcement.

Extension

Valid entries	Usage
1 to **7**	The extension number associated with the announcement being added/ displayed/changed/removed. This field is display-only. It is auto-populated based on the extension entered in the command line.

CAUTION:
When entering a Multi-Location Dial Plan shortened extension in a field designed for announcement extensions, certain administration end validations that are normally performed on announcement extensions are not done, and resultant warnings or submittal denials do not occur. The shortened extensions also do not appear in any display or list that shows announcement extensions. Extra care should be taken to administer the correct type of announcement for the application when assigning shortened extensions.

Group/Port

Use this field to enter the announcement board location or the Audio Group number. If **Annc Type** is **integrated**, then this field displays as **Group/Board**. Also, when **Annc Type** is integrated, the **Queue Length** field does not appear. If **Annc Type** is *not* **integrated**, then the **Group/Port** field displays as **Port**.

Type the group number in one of the following ways:

- **Gnn** where ***nn*** represents a one or two-digit audio group number.
- The location of the VAL or the TN750 announcement circuit pack. Enter the necessary characters in the **aaxss** format (where ***aa*** = the cabinet number, ***x*** = the carrier, and ***ss*** = the slot number).
- **gggv9** for media gateway vVAL, where ***ggg*** is the gateway number of the media gateway (up to 250).

Note:
To administer DID Intercept announcements in a multi-location system where each location or city needs a different announcement, enter an audio group in this field instead of a VAL port.

Protected

Use this field to set the protection mode for an integrated announcement/music extension. When you set this field to **y**, the recording is protected and cannot be deleted or changed via a telephone session or FTP (via SAT or VAL Manager). When you set this field to **n**, the recording can be changed or deleted by users with console permissions to delete or change the recording. Changing or deleting using the telephone recording session requires the **console permissions** class of service (COS). When the **Type** is **analog**, **ds1** or **aux-trunk**, **N/A** appears in this field.

Valid entries	Usage
y	Enter **y** to protect the integrated announcement from being deleted or changed by any user. For VAL, after an announcement file resides on the circuit pack (recorded or FTP transfer), you can set this field to **y** to protect the file (read-only).
n	Enter **n** to allow telephone session users with console permission and/or FTP to change or delete an announcement. Use this value when you initially administer an announcement or subsequently need to change or delete it.

Queue

Valid entries	Usage
y(es)	Enter **y** to queue calls for the announcement if the **Type** field is **integrated**, **integ-rep** or **aux-trunk**. The caller is always connected to the beginning of the announcement. Enter **y** for ACD and vectoring delay announcements. Call centers should always use this option. This is the default.

Valid entries	Usage
n(o)	No queue and no barge-in. The caller is always connected to the beginning of the announcement. The announcement does not play if a port is not available.
b(argein)	Enter **b** to set up barge-in if the **Type** field is **integrated**, **integ-rep** or **aux-trunk**. When **Type** is **integ-mus**, this field defaults to **b**. Callers are connected to the announcement at any time while it is playing. **Note:** The same non-barge-in announcement can be played through more than one port (or all ports) of an integrated circuit pack. The initial request to play an announcement selects an available port on the board on which the announcement resides. If there are additional requests to play the announcement while it is playing on another port(s), another port is selected. If all ports are busy, new requests to play announcements go to the integrated announcement system queue (**Q** field must be **y**). Otherwise, the request to play is denied, and processing continues without the caller hearing the announcement. When a port becomes available, all queued calls (up to the platform "calls connected" limit) are connected at the same time to hear the announcement play from the beginning. A barge-in announcement starts playing when first requested and continues playing through a port, repeating until there are no more requests. Call processing simultaneously connects calls to the playing barge-in announcement. Each call remains connected until the requesting feature operation removes the call (for example, wait step times out). Barge-in type announcements never select another port to play the same announcement once it is playing on a specific port.

Queue Length

The queue length is the number of calls that can queue for this announcement. The maximum number of queues allowed depends on your system configuration.

The **Queue Length** field applies if the **Queue** field is **y** and the **Type** field is **analog**, ds1 or **aux-trunk**. When the **Type** field is **integrated** or **integ-rep**, **N/A** appears in this field. Integrated announcements have a pre-set queue length

.

Valid entries	Usage
The maximum number your system allows	Number of calls that can be queued for this announcement

Rate

Enter the recording rate speed (in 1000 bits/second) for TN750 or ISSPA integrated announcements. A different recording speed can be used for each integrated announcement. With VAL type sources, the default is **64** and cannot be changed. When the **Type** field is **analog**, **ds1** or **aux-trunk**, **N/A** appears in this field.

Valid entries	Usage
16	16 kbps (8 minutes and 32 seconds of announcement time per circuit pack or 1 hour and 24 minutes for 10 circuit packs for the TN750; for the ISSPA, there are 240 minutes of storage time). This rate does not provide a high-quality recording. Avaya does not recommend this for customer announcements, but it is adequate for VDN of Origin announcements.
32	32 kbps (4 minutes and 16 seconds of total announcement time for the TN750; for the ISSPA, there are 120 minutes of storage time).
64	64 kbps (for 2 minutes and 8 seconds of announcement time per circuit pack or 42 minutes for 10 circuit packs for the TN750; for the ISSPA, there are 60 minutes of storage time). This is the default for VAL.

TN

Valid entries	Usage
1 to **100**	Enter the Tenant Partition number, if any.

ARS Toll Table

This screen assigns ARS Toll Tables used by Subnet Trunking. Use it to specify whether calls to CO codes listed on the table are toll or non-toll calls. You specify non-toll calls based on the last 2 digits of the distant-end of the trunk group.

Field descriptions for page 1

Figure 19: ARS Toll Table screen

```
change ars toll
                              ARS TOLL TABLE: __              Page 1 of x

                             OFFICE CODES: x00-x99
   00: y   10: y   20: y   30: y  40: y  50: y  60: y  70: y  80: y  90: y
   01: y   11: y   21: y   31: y  41: y  51: y  61: y  71: y  81: y  91: y
   02: y   12: y   22: y   32: y  42: y  52: y  62: y  72: y  82: y  92: y
   03: y   13: y   23: y   33: y  43: y  53: y  63: y  73: y  83: y  93: y
   04: y   14: y   24: y   34: y  44: y  54: y  64: y  74: y  84: y  94: y
   05: y   15: y   25: y   35: y  45: y  55: y  65: y  75: y  85: y  95: y
   06: y   16: y   26: y   36: y  46: y  56: y  66: y  76: y  86: y  96: y
   07: y   17: y   27: y   37: y  47: y  57: y  67: y  77: y  87: y  97: y
   08: y   18: y   28: y   38: y  48: y  58: y  68: y  78: y  88: y  98: y
   09: y   19: y   29: y   39: y  49: y  59: y  69: y  79: y  89: y  99: y
```

00: through 99:

These fields represent the last 2 digits of the codes within the 100-block of numbers. Designate each as a number toll or non-toll call.

Valid entries	Usage
y/n	Enter **n** to designate a non-toll CO code.

ARS TOLL TABLE

Valid entries	Usage
2 through **9**	Identify the number of the ARS Toll Table.

OFFICE CODES

Valid entries	Usage
200 to **299** through **900** to **999**	Identify the block of numbers on this screen.

Attendant Console

This screen assigns an Attendant Console to the system.

Field descriptions for page 1

Figure 20: Attendant Console screen

```
add attendant n                                                    Page   1 of  x
                             ATTENDANT CONSOLE 1

         Type: console           Name: 27 character attd cons name
    Extension: 1000             Group: 1                Auto Answer: none
 Console Type: principal           TN: 1                Data Module? y
         Port: 01C1106            COR: 1          Disp Client Redir? n
Security Code:                    COS: 1           Display Language: english
                                                  H.320 Conversion? n

DIRECT TRUNK GROUP SELECT BUTTON ASSIGNMENTS (Trunk Access Codes)
    Local Remote             Local Remote              Local Remote
 1: 9                    5:                       9:
 2: 82                   6:                      10:
 3:                      7:                      11:
 4:                      8:                      12:

HUNDREDS SELECT BUTTON ASSIGNMENTS
 1:           5:            9:           13:           17:
 2:           6:           10:           14:           18:
 3:           7:           11:           15:           19:
 4:           8:           12:           16:           20:
```

Attendant Console x

This is a display-only field when the screen is accessed using an administration command such as **add** or **change**.

Auto Answer

Valid entries	Usage
all	Entering **all** indicates an incoming call to an idle attendant is answered automatically without any action (no button presses required) by the attendant.
acd	Entering **acd** indicates only ACD split/skill calls and direct agent calls can auto answer. Non-ACD calls terminated to an attendant console with Auto Answer set to **acd** ring audibly.
none	Entering **none** causes all calls terminated to this attendant console to receive some sort of audible ringing treatment.

Console Type

Enter this console's intended use. There can only be one **night-only** or one **day/night** console in the system unless Tenant Partitioning is administered. Night Service is activated from the principal console or from the one station set per-system that has a **nite-serv** button.

Valid entries	Usage
principal	Puts the attendant console into night service.
day-only	Handles only day service calls.
night-only	Handles only night service calls.
day/night	Handles day or night service calls.

COR

Valid entries	Usage
0 through **95**	Enter the class of restriction that reflects the desired restriction.

COS

Valid entries	Usage
0 through **15**	Enter the class of service (COS) for this attendant console.

Data Module

Valid entries	Usage
y/n	Enter **y** if the console is to be connected to a data terminal via 7400B or 8400 Data Module. If **y** is entered, complete the Data Module screen (page 4).

Disp Client Redir

This field is administrable only if the Hospitality feature has been enabled on the System Parameters Customer-Options (Optional Features) screen. This field affects the station's display on calls originated from a station with Client Room Class of Service.

Valid entries	Usage
y	When the field is **y**, the redirection information for a call originating from a Client Room and terminating to this station displays. **Note:** For stations with an **audix** station type, AUDIX Voice Power ports, or ports for any other type of messaging that needs display information, this field must be **y**.
n	When the field is **n**, then for all calls originating from a Client Room (even redirected calls) that terminate to this station, this station's display does not show the redirection information. Only the client name and extension (or room, depending on what is administered on the Hospitality screen) displays.

Display Language

Enter the language in which you want console messages displayed.

Valid entries	Usage
English	Enter the language in which you want messages to be displayed.
French	

Valid entries	Usage
Italian	
Spanish	
user-defined	
Unicode	Unicode display is only available for Unicode-supported telephones. Currently, the 4610SW, 4620SW, 4621SW, and 4622SW, Sage, Spark, and 9600-series Spice telephones support Unicode display. Unicode is also an option for the 2420J telephone when **Display Character Set** on the **System Parameters Country-Options** screen is **katakana**. For more information on the 2420J, see *2420 Digital Telephone User's Guide*, 555-250-701.

Extension (Optional)

Enter the extension for the individual attendant console. Individual attendant extensions allow attendants to use features that an attendant group cannot use. For example, extensions can be members of a DDC or UCD group. An individual attendant extension can have its own Class of Restriction and Class of Service.

If you give your attendants an individual extension, users can call the attendant by dialing the extension or you can assign them an abbreviated-dialing button for fast access to the attendant.

Valid entries	Usage
An unassigned extension or blank	If an extension is not assigned, the attendant can only be addressed as a member of the attendant group. If the attendant has a data module, the **Extension** field cannot be blank.

Group

Valid entries	Usage
1 to **128**	Enter the Attendant Group number.

H.320 Conversion

Allows H.320 compliant calls made to this telephone to be converted to voice-only. Because the system can handle only a limited number of conversion calls, you might need to limit the number of telephones with H.320 conversion.

Valid entries	Usage
y/n	Enter **y** for H.320 compliant calls.

Name

Enter the name of this console.

Valid entries	Usage
Up to 27 alphanumeric characters	Any entry is accepted. **NOTE:** Avaya BRI stations support ASCII characters only. They do not support non-ASCII characters, such as Eurofont or Kanafont. Therefore, if you use non-ASCII characters in any Communication Manager **Name** field, such characters do not display correctly on a BRI station.

Port

Enter the necessary characters.

Valid entries	Usage
01 to **03** (DEFINITY CSI) or **1** to **64** (S87XX/S8300 Servers)	First and second characters are the cabinet number.
A to **E**	Third character is the carrier.
0 to **20**	Fourth and fifth characters are the slot number.
01 to **04** (Analog TIE trunks) **01** to **31**	Six and seventh characters are the circuit number.
1 through **80** (DEFINITY CSI) or **1** through **250** (S87XX/S8300 Servers)	Gateway
V1 through **V9**	Module
	1 of 2

Valid entries	Usage
01 through **31**	Circuit
ip	SoftConsole IP attendant. You also must have the **Type** field as **302B** and enter a security code. ip is allowed only if, on the System Parameters Customer-Options (Optional Features) screen, the **IP Attendant Consoles** field is **y**.
x	Indicates that there is no hardware associated with the port assignment. An individual attendant extension must be assigned in the **Extension** field.
	2 of 2

For example, 01A0612 is in cabinet 01, carrier A, slot 06, and circuit number (port) 12.

Each attendant console requires a port on a digital line circuit pack. For reliability, the attendant consoles should not be assigned to ports on the same digital line circuit pack. For example, if three attendant consoles are to be provided, assign each console to a port on three different digital line circuit pack, if possible. However, if required, all attendant consoles can be assigned to ports on the same digital line circuit pack.

Security Code

Does not apply to S87XX Series IP-PNC. Enter the security code required by the SoftConsole IP attendant. The required security code length is determined by **Minimum Security Code Length** on the Feature-Related System Parameters screen.

TN

Valid entries	Usage
1 to **100**	Enter the Tenant Partition number.

Type

Valid entries	Usage
console	Indicates the type of attendant console being administered.
302	Use for 302B/C/D or SoftConsole IP attendant.

DIRECT TRUNK GROUP SELECT BUTTON ASSIGNMENTS (Trunk Access Codes)

Enter the trunk access codes (TACs) for local and remote servers. (There are fields for one local TAC and one remote TAC per button labeled **Local** and **Remote**.) The local TAC (1 to 4 digits) refers to a trunk group or Loudspeaker Paging zone on this server. Remote TACs are only useful in a private network (including DCS) network. The remote TAC (1 to 3 digits) refers to a trunk group on the remote server. If a remote TAC is given, then the local TAC must see a trunk group that connects directly to the remote server running Communication Manager and is also limited to 1 to 3 digits.

Avaya recommends a DCS trunk be specified as the local TAC between the local and remote servers. If the TAC specified as local between the local and remote servers is not a DCS trunk, the remote trunk cannot be monitored by the local server running Communication Manager.

Valid entries	Usage
1 to 4 digit number	Enter the trunk access codes (TACs) for local and remote servers.
***** or #	Can be used as first digit

HUNDREDS SELECT BUTTON ASSIGNMENTS

Enter in the appropriate field (1 through 20), the hundreds group to be associated with a **Hundreds Group Select** button located on an optional selector console.

Valid entries	Usage
1 to 5 digit hundreds group (plus prefix, if needed)	Fields 1 through 8 are used when the selector console is a 24A-type console and fields 1 through 20 are used for a 26A-type console. Enter a hundreds group number that represents all but the last two digits of an extension number (for example, the **Hundreds Select** Button — on the selector console — for extension 3822 would be "38").

Field descriptions for page 2

Figure 21: Attendant Console screen (page 2)

```
change attendant 1                                              Page   2 of   4
                              ATTENDANT CONSOLE

VIS FEATURE OPTIONS

              Auto Start? y
      Echo Digits Dialed? y
```

VIS FEATURE OPTIONS

Use these fields to administer Visually Impaired Service option.

Auto Start

Valid entries	Usage
y/n	Enter **y** to allow an attendant to press any key on the keypad to start a call without the need to first press the **Start** button.

Echo Digits Dialed

Valid entries	Usage
y/n	Enter **y** to provide voiced confirmation of dialed digits.

Field descriptions for page 2 (SoftConsole IP Attendant)

Figure 22: Attendant Console Data Module screen (page 2)

```
change attendant n                                              Page 2 of x
                                   ATTENDANT
IP FEATURE OPTIONS


Remote Softphone Emergency Calls: as-on-local  Direct IP-IP Audio Connections?y
Emergency Location Ext: 1001      Always use? n          IP Audio Hairpinning?n
```

Always Use

This field does not apply to SCCAN wireless telephones, or to extensions administered as type **h.323**.

Valid entries	Usage
y	When this field is **y**: ● The **Remote Softphone Emergency Calls** field is hidden. A softphone can register no matter what emergency call handling settings the user has entered into the softphone. If a softphone dials 911, the **Emergency Location Extension** administered on the Station screen is used. The softphone's user-entered settings are ignored. ● If an IP telephone dials 911, the **Emergency Location Extension** administered on the Station screen is used.
n	For more information, see the description for the **Emergency Location Extension** field on the Station screen. This is the default.

Direct IP-IP Audio Connections

Allows direct audio connections between IP endpoints.

Valid entries	Usage
y/n	Enter **y** to save on bandwidth resources and improve sound quality of voice over IP transmissions.

Emergency Location Ext

The **Emergency Location Ext** field defaults to the telephone's extension. This extension is the starting point for identifying the street address or nearby location when an emergency call is made. The entry in this field is manipulated by CAMA Numbering Format before being sent over CAMA trunks; or similarly by Numbering — Public/Unknown Format before being sent over ISDN trunks. For more information about this field, see the **Usage** description for the **Remote Softphone Emergency Calls** field on the next page.

Valid entries	Usage
0 to **9**	Enter the Emergency Location Extension for the SoftConsole IP Attendant.

IP Audio Hairpinning

Allows IP endpoints to be connected through the server's IP circuit pack.

Valid entries	Usage
y/n	Enter **y** to allow IP endpoints to be connected through the IP circuit pack for Communication Manager in IP format, without going through the Communication Manager TDM bus. Default is **n**.

Remote Softphone Emergency Calls

Use this field to tell Communication Manager how to handle emergency calls from the IP telephone. This field appears when the **IP Softphone** field is set to **y** on the Station screen.

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

Valid entries	Usage
as-on-local	Type **as-on-local** to achieve the following results: ● If the administrator populates the IP Address Mapping screen with emergency numbers, the value **as-on-local** functions as follows: ● If the **Emergency Location Extension** field in the Attendant Console screen is the same as the **Emergency Location Extension** field in the IP Address Mapping screen, the value **as-on-local** sends the extension to the Public Safety Answering Point (PSAP). ● If the **Emergency Location Extension** field in the Attendant Console screen is different from the **Emergency Location Extension** field in the IP Address Mapping screen, the value **as-on-local** sends the extension in the IP Address Mapping screen to the Public Safety Answering Point (PSAP).
block	Enter **block** to prevent the completion of emergency calls. Use this entry for users who move around but always have a circuit-switched telephone nearby, and for users who are farther away from the Avaya S8xxx Server than an adjacent area code served by the same 911 Tandem office. When users attempt to dial an emergency call from an IP Telephone and the call is blocked, they can dial 911 from a nearby circuit-switched telephone instead.
	1 of 2

Valid entries	Usage
cesid	Enter **cesid** to allow Communication Manager to send the CESID information supplied by the IP Softphone to the PSAP. The end user enters the emergency information into the IP Softphone. Use this entry for IP Softphones with road warrior service that are near enough to the Avaya S8XXX Server that an emergency call routed over the it's trunk reaches the PSAP that covers the server or switch. If the Avaya S8XXX Server uses ISDN trunks for emergency calls, the digit string is the telephone number, provided that the number is a local direct-dial number with the local area code, at the physical location of the IP Softphone. If the Avaya S8XXX Server uses CAMA trunks for emergency calls, the end user enters a specific digit string for each IP Softphone location, based on advice from the local emergency response personnel.
option	Enter **option** to allow the user to select the option (extension, block, or cesid) that the user selected during registration and the IP Softphone reported. Use this entry for extensions that can be swapped back and forth between IP Softphones and a telephone with a fixed location. The user chooses between **block** and **cesid** on the softphone. A DCP or IP telephone in the office automatically selects **extension**.
	2 of 2

Field descriptions for Attendant Console Data Module screen

This page displays as page 3 if the **Data Module** field on Page 1 is **y**.

Figure 23: Attendant Console Data Module screen

```
change attendant n                                                  Page 3 of x
ATTENDANT
DATA MODULE
     Data Extension: ____         Name: _______________   BCC: 2
                                   COS: 1_
                                   COR: 1_
                ITC: restricted     TN: 1_

ABBREVIATED DIALING
 List1: ________

SPECIAL DIALING OPTION:


ASSIGNED MEMBER (Station with a data extension button for this data module)
        Ext      Name
     1:
```

DATA MODULE

BCC

A display-only field that appears when the **ISDN-PRI** or **ISDN-BRI Trunks** field is enabled on the System Parameters Customer-Options (Optional Features) screen.

Note:
The **BCC** value is used to determine compatibility when non-ISDN facilities are connected to ISDN facilities (ISDN Interworking feature).

COR

Valid entries	Usage
0 to **995**	Enter the desired class of restriction (COR) number.

COS

Valid entries	Usage
0 to **15**	Enter the desired (COS) number to designate allowed features. See Class of Service on page 131 for additional information on the allowed features.

Data Extension

Enter the extension number assigned to the data module.

Valid entries	Usage
1 to 5-digit number	Must agree with the system's dial plan

Name

Enter the name of the user associated with the data module. The name is optional; it can be left blank.

Note:
Avaya BRI stations support ASCII characters only. They do not support non-ASCII characters, such as Eurofont or Kanafont. Therefore, if you use non-ASCII characters in any Communication Manager **Name** field, such characters do not display correctly on a BRI station.

TN

Valid entries	Usage
1 to **100**	Enter the Tenant Partitioning number.

ABBREVIATED DIALING

List1

Valid entries	Usage
s	System
g	Group. If **g** is entered, a group number is also required.
p	Personal. If **p** is entered, a personal list number also is required.
e	Enhanced

SPECIAL DIALING OPTION

Valid entries	Usage
hot-line	Enter one of the dialing options that are available. This identifies the destination of all calls when this data module originates calls.
default	

HOT LINE DESTINATION — Abbreviated Dialing Dial Code

Only displays when the **Special Dialing Option** field is **hot-line** or **default** (S87XX Series IP-PNC only). The associated AD number is dialed when the user goes off-hook on a Data Hot Line call.

Hot Line Service allows single-line telephone users, by simply lifting the handset, to automatically place a call to a preassigned destination (extension, telephone number, or feature access code).

The Hot Line Service destination number is stored in an Abbreviated Dialing List.

A Direct Department Calling (DDC), a Uniform Call Distribution (UCD), a Terminating Extension Group (TEG) extension, or any individual extension within a group can be a Hot Line Service destination. Also, any extension within a DDC group, UDC group, or TEG can have Hot Line Service assigned.

Use Hot Line Service when very fast service is required and when you use a telephone only for accessing a certain facility. Loudspeaker Paging Access can be used with Hot Line Service to provide automatic access to paging equipment.

Valid entries	Usage
A dial code	Within the range of the abbreviated dialing list type

DEFAULT DIALING Abbreviated Dialing Dial Code

The associated AD number is dialed when the user goes off-hook and enters a carriage return following the "DIAL" prompt. The data call originator also can perform data terminal dialing by specifying a dial string that might or might not contain alphanumeric names. Only displays when the **Special Dialing Option** field is **default**.

Valid entries	Usage
A dial code	Within the range of the abbreviated dialing list type

Ext

This is the extension number of a previously administered user who has an associated **Data Extension** button and who shares the use of the module.

Name

Contains the name assigned to the above extension number.

Field descriptions for page 3

Figure 24: Attendant Console screen

```
change attendant n                                              Page 3 of x
                                ATTENDANT CONSOLE
FEATURE BUTTON ASSIGNMENTS
  1: split_____                          13: __________
  2: __________                          14: __________
  3: __________                          15: __________
  4: __________                          16: __________
  5: __________                          17: __________
  6: hold _____  *                       18: __________
  7: __________                          19: forced-rel
  8: aux-work      RC:    Grp:           20: __________
  9: __________                          21: __________
 10: __________                          22: __________
 11: __________                          23: night-serv *
 12: __________                          24: pos-busy__ *
```

If this is a non-IP attendant console this is page 3 of the Attendant Console screen.

FEATURE BUTTON ASSIGNMENTS

Enter the feature buttons from that you want to assign to the attendant console. The fixed buttons that cannot be changed (that is, split and forced release) are shown on the screen. The **hold**, **night-serv**, and **pos-busy** buttons are shown in the system default locations. These buttons can be administered elsewhere on the screen. The following table provides descriptions of feature buttons that are unique to the attendant console.

Valid entries	Usage
Audible Tones On/Off	
cw-ringoff	Call waiting ringer off; turns on/off the audible tone for call waiting on attendant console (1 per console).
in-ringoff	Incoming call ringer off; turns on/off the audible tone for incoming call ringer (1 per console).
re-ringoff	Timed reminder ringer off; turns on/off the audible tone for timer reminder ringer (1 per console).
	1 of 4

Valid entries	Usage
alt-frl	Alternate FRL. Alternate facility restriction level; allows the attendant to activate or deactivate the AFRL feature. When activated, this allows the originating device (lines or trunks) to use an alternate set of the facility restriction levels to originate a call (1 per console).
Attendant Control of Trunk Group Access	
act-tr-grp	Activate trunk group access; allows the attendant to control a trunk group. All calls going to the trunks are routed to the attendant (1 per console).
deact-tr-g	Deactivate trunk group access; allows the attendant to release control of a trunk group (1 per console).
class-rstr	Display Class of Restriction. Used to display the COR associated with a call (1 per console).
em-acc-att	Emergency Access to the Attendant. The associated status lamp is flashed when there are one or more calls on the emergency attendant queue (1 per console).
hold	Hold. When the **Hold** button is pressed while the attendant is active on a loop, the party on the loop is put on hold and the **call type** button associated with the loop is lit (1 per console).
pos-busy	Position Busy. When this button is pushed, the attendant is put into position busy mode, the "Pos Avail" light is turned off, and the light associated with the **pos-busy** button is lit. Pushing the **pos-busy** button a second time takes the console out of "position busy" mode, turns on the "Pos Avail" light and turns off the light associated with the **pos-busy** button. If the **pos-busy** button is administered on a 2-LED button, the top LED flashes when the last attendant goes into "Position Busy" mode. Otherwise, if the button has only one LED, the single LED associated with the **pos-busy** button flashes (1 per console).
serial-cal	Serial Call. This button allows the attendant-extended calls to return to the same attendant if the trunk remains off-hook (1 per console).
override	Attendant Override. This button enables the attendant to override diversion features such as, Call Forwarding, Call Coverage, and so on (1 per console).
intrusion	Call Offer. Depression of this button allows the attendant to extend a call when the called party is active on another call (1 per console).
dont-split	Don't Split. This button allows the attendant to not split away a call when dialing (1 per console).
	2 of 4

Valid entries	Usage
vis	Visually Impaired Attendant Service (**vis**) — This button activates visually impaired service for the attendant. When this service is activated, the attendant can listen to console status or messages by pressing buttons that have been translated as follows: ● "con-stat" repeats the console status. ● "display" calls out display contents. ● "dtgs-stat" calls out the DTGS status. ● "last-mess" repeats the last message. ● "last-op" calls out the last operation.
Trunk Group Select — In addition to the 12 Direct Trunk Group Selection (DTGS) Button Assignments on Field descriptions for page 1, up to 12 single lamp **DTGS** buttons can be administered on this page. The status lamp associated with the feature button is used to monitor the busy/idle status of the trunk. Trunk groups administered on these buttons cannot be controlled using **Attendant Control of Trunk Group Select** buttons. The single lamp **DTGS** buttons can be administered as follows:	
local-tgs	Local trunk group select; allows the attendant to access trunk groups on the local server running Communication Manager (combination of 12 local-tgs/remote-tgs per console).
remote-tgs	Remote trunk group select; allows the attendant to access trunk groups on a remote server running Communication Manager (combination of 12 local-tgs/remote-tgs per console).
hundrd-sel	Hundreds group select; in addition to the fixed **HGS** buttons on Field descriptions for page 1, a user can administer hundreds group select feature buttons on this page. When a feature button is administered as "hundrd-sel," a subfield appears that must then be administered in the same manner as the fixed **HGS** button fields (a 1 to 3 digit hundreds group plus prefix, if needed). Administered **hundrd-sel** feature buttons operate in the same manner as fixed **HGS** buttons. The total number of hundreds group select buttons (fixed and administered) allowed on a console is 20. Thus, if all 20 fixed **HGS** buttons have been administered, no **hundrd-sel** feature buttons can be administered. **Note:** If no fixed **HGS** buttons are administered, 19 **hundrd-sel** feature buttons are available. This is because 5 of the 24 feature buttons must be used for required feature buttons (**hold**, **pos-busy**, **night-serv**, **forced-rel**, and **split**)
group-disp	Group Display. Allows the attendant to see a display of extensions currently being tracked on the DXS module.
group-sel	Group Select. Allows the attendant to select a specific group of hundreds by dialing the first 2 or 3 digits of the hundreds group.
	3 of 4

Valid entries	Usage
Attendant Room Status	
occ-rooms	Occupied rooms; allows the attendant to see which rooms are occupied.
maid-stat	Maid status; allows the attendant to see which rooms are in one of six specified states.
vu-display	VuStats (**vu-display**) — This button allows users with display telephones and attendants to turn on the VuStats display. The limit to the number of VuStats feature buttons you can administer depends on how many feature buttons are available on the attendant console you are administering. The system is designed to allow you to set up a separate VuStats display format for each feature button. Therefore, agents can change the type of measurements on their display by selecting a different VuStats feature button.
	4 of 4

- If 12 HGS buttons are assigned on field descriptions for page 2, Avaya recommends that the **night**, **pos-busy**, and **hold** buttons be reassigned to locations 20, 21, and 3, respectively. The **HGS** buttons should then be assigned to the right-most three columns, as required.

Field descriptions for page 4

Figure 25: Attendant Console screen

```
change attendant n                                              Page 4 of x
                                  ATTENDANT CONSOLE
DISPLAY BUTTON ASSIGNMENTS
  1: normal____                              5: delete-msg
  2: inspect___                              6: call-disp_
  3: cov-msg-rt                              7: date-time_
  4: next______                              8: timer_____
```

DISPLAY MODULE BUTTON ASSIGNMENTS

Display-type buttons obtain display functions on the associated alphanumeric display. These buttons are noted as [display button] in the **Feature** or **Function** column on the table. Also, several feature buttons can be administered so that their associated status lamps can be used to provide visual indications of the associated feature or function. In some cases, the button itself is not operational. These buttons are noted as [status lamp]. If a Call Cover Msg Rt (**cov-msg-rt**) button is assigned, a Leave Word Calling Delete Msg (**delete-msg**) button and a Next (**next**) button must also be assigned.

Audio Group

Use the Audio Group screen to add, change, or display a specified audio group. An audio group is a collection of recorded audio sources that have been placed in a group to facilitate their selection. The three pages of this screen provide for administering up to 260 audio source locations for an audio group.

Field descriptions for page 1

Figure 26: Audio Group screen

```
add audio-group next                                         Page 1 of x
                              Audio Group 2

                            Group Name:

AUDIO SOURCE LOCATION
     1:         16:         31:         46:         61:         76:
     2:         17:         32:         47:         62:         77:
     3:         18:         33:         48:         63:         78:
     4:         19:         34:         49:         64:         79:
     5:         29:         35:         50:         65:         80:
     6:         21:         36:         51:         66:         81:
     7:         22:         37:         52:         67:         82:
     8:         23:         38:         53:         68:         83:
     9:         24:         39:         54:         69:         84:
    10:         25:         40:         55:         70:         85:
    11:         26:         41:         56:         71:         86:
    12:         27:         42:         57:         72:         87:
    13:         28:         43:         58:         73:         88:
    14:         29:         44:         59:         74:         89:
    15:         30:         45:         60:         75:         90:
```

Audio Source Location

Enter the board location for this audio group: cabinet(1-64):carrier(A-E):slot(1-20):OR gateway(1-250):module(V1-V9).

Group Name

Enter an alpha-numeric name of the audio group for identification.

Note:
Avaya BRI stations support ASCII characters only. They do not support non-ASCII characters, such as Eurofont or Kanafont. Therefore, if you use non-ASCII characters in any Communication Manager **Name** field, such characters do not display correctly on a BRI station.

Audix-MSA Node Names

Field descriptions for page 1

Figure 27: Audix-MSA Node Names screen

```
change node-names audix-msa                                     Page 1 of x

                        AUDIX-MSA NODE NAMES

 Audix Name         IP Address     MSA Names      IP Address
 audixA_            __.__.__.__    _______        __.__.__.__
 audixB_            __.__.__.__    _______        __.__.__.__
 _______            __.__.__.__    _______        __.__.__.__
 _______            __.__.__.__    _______        __.__.__.__
 _______            __.__.__.__    _______        __.__.__.__
 _______            __.__.__.__    _______        __.__.__.__
 _______            __.__.__.__    _______        __.__.__.__
 _______            __.__.__.__    _______        __.__.__.__
```

Audix Names

Identifies the name of the AUDIX node.

Valid entries	Usage
1 to 7 character string	Used as a label for the associated IP address. The node names must be unique on each server running Communication Manager.

IP Address

The IP address associated with the node name.

MSA Names

Identifies the name of the MSA node.

Valid entries	Usage
1 to 7 character string	Used as a label for the associated IP address. The MSA names must be unique on each server running Communication Manager.

Authorization Code — COR Mapping

You use this screen to assign authorization codes and the class of restriction (COR) that is associated with a given authorization code. See Authorization Codes and Class of Restriction in *Avaya Aura™ Communication Manager Feature Description and Implementation,* 555-245-205, for more information on how Authorization Codes work with COR.

To maximize the security of your system:

- Administer authorization codes to the maximum length allowed by the system
- Create random (nonconsecutive) authorization codes
- Change authorization codes at least quarterly
- Deactivate authorization codes immediately if a user leaves the company or changes assignments
- Assign each authorization code the minimum level of calling permissions required

Field descriptions for page 1

Figure 28: Authorization Code - COR Mapping screen

```
change authorization-code                                       Page 1 of x
                       Authorization Code - COR Mapping
      Note:    XX codes administered.  Use "list" to display all codes.
     AC     COR    AC     COR    AC     COR    AC     COR    AC      COR    AC     COR

    _______ __  _______ __  _______ __  _______ __   _______ __   _______ __
    _______ __  _______ __  _______ __  _______ __   _______ __   _______ __
    _______ __  _______ __  _______ __  _______ __   _______ __   _______ __
    _______ __  _______ __  _______ __  _______ __   _______ __   _______ __
    _______ __  _______ __  _______ __  _______ __   _______ __   _______ __
    _______ __  _______ __  _______ __  _______ __   _______ __   _______ __
    _______ __  _______ __  _______ __  _______ __   _______ __   _______ __
    _______ __  _______ __  _______ __  _______ __   _______ __   _______ __
    _______ __  _______ __  _______ __  _______ __   _______ __   _______ __
    _______ __  _______ __  _______ __  _______ __   _______ __   _______ __
    _______ __  _______ __  _______ __  _______ __   _______ __   _______ __
    _______ __  _______ __  _______ __  _______ __   _______ __   _______ __
    _______ __  _______ __  _______ __  _______ __   _______ __   _______ __
    _______ __  _______ __  _______ __  _______ __   _______ __   _______ __
    _______ __  _______ __  _______ __  _______ __   _______ __   _______ __
    _______ __  _______ __  _______ __  _______ __   _______ __   _______ __
```

AC

Valid entries	Usage
Any combination of between 4 and 13 digits	The number of digits must agree with the number assigned to the **Authorization Code Length** field on the Feature-Related System Parameters screen. To enhance system security, choose Authorization Codes of 13 random digits.

COR

Valid entries	Usage
0 to **995**	When a user dials the associated authorization code, this is the COR that the telephone or other facility assumes for that call.

Number of Codes Administered

Displays the number of Authorization Codes already administered using the Authorization Codes screen. There is a maximum number of authorization codes that you can use. To find out what this number is for your system, type **display capacity**, and page down to find the authorization code information. For details on the System Capacity screen, see *Maintenance Commands for Avaya Aura™ Communication Manager, Media Gateways and Servers*, 03-300431.

Authorization Code - PIN Checking for Private Calls

This feature restricts users from making private calls (internal or external) by forcing them to enter Personal Identification Number (PIN) code after dialing PIN feature access code and only when the PIN is valid, the user can dial the destination digits to make a call.

PINs are administered on the same screen as Authorization codes. There is no way looking at the administration to distinguish whether particular entry is PIN or Auth Code. So, if the user is assigned with some Auth Code then it can be used instead of PIN.

Figure 29: Authorization Code Screen

```
 change authorization-code 1234567                                Page 1 of 1
                Authorization Code - COR Mapping
       NOTE: 1      codes administered. Use 'list' to display all codes

     AC       COR         AC       COR         AC       COR         AC       COR
1234567       1
2345678       2
```

Figure 29 shows administration of PIN ‘1234567’and Auth Code ‘2345678’ in same screen. There is no way to distinguish that 1234567 is PIN and 2345678 is Auth Code. 2345678 can also be used as PIN if the COR assigned to that is administered with proper privileges.

For more information on PIN Checking for Private Calls, see *Avaya Aura™ Communication Manager Feature Description and Implementation*, 555-245-205.

Best Service Routing

This screen administers the routing numbers for each location associated with a particular application. This allows the Avaya DEFINITY server or Avaya S8XXX Server to compare specified skills, identify the skill that provides the best service to a call, and deliver the call to that resource.

For information on setting up Best Service Routing (BSR), see *Avaya Aura™ Call Center 5.2 Automatic Call Distribution (ACD) Reference*, 07-602568.

Field descriptions for page 1

Figure 30: Best Service Routing screen

```
change best-service-routing n                                 Page 1 of x

                            BEST SERVICE ROUTING

Number: 1     Name: ARS              Maximum Suppression Time: 30     Lock? n

Num    Location Name  Switch Node    Status Poll VDN    Interflow VDN  Net Redir?
1      st10     auto                 95022011           3035552121         y
4      st10     auto                 95022014           3035551110         n
```

Interflow VDN

Valid entries	Usage
0 to **9**, *****, #, **~p** (pause) **~w/~W** (wait) **~m** (mark) **~s** (suppress) blank (DEFINITY CSI)	When a given remote Avaya server is the best available, the origin Avaya server interflows the call to this vector on the remote server. Each remote Avaya server in a given application has to have a dedicated interflow server.

Location Name

Indicates the location.

Valid entries	Usage
Up to 15 alphanumeric characters. (DEFINITY CSI)	Enter a name for the location.

Lock

Indicates whether this application is locked.

Valid entries	Usage
y/n (DEFINITY CSI)	Set to **y** to prevent this application from being sent to Call Management System (CMS).

Maximum Suppression Time

Prevents callers from connecting to a VDN within a certain time period after receiving a busy signal.

Valid entries	Usage
0 to **60** (DEFINITY CSI)	Enter time in seconds.

Name

Contains the name assigned to the BSR number.

Valid entries	Usage
Up to 15 alphanumeric characters. (DEFINITY CSI)	Assign a descriptive name for the physical location. **NOTE:** Avaya BRI stations support ASCII characters only. They do not support non-ASCII characters, such as Eurofont or Kanafont. Therefore, if you use non-ASCII characters in any Communication Manager **Name** field, such characters do not display correctly on a BRI station.

Net Redir

Valid entries	Usage
y/n (DEFINITY CSI)	Set to **y** for each to location to which calls are to be redirected using Network Call Redirection.

Num

This field corresponds to the "consider location x" step from the Call Vector screen.

Valid entries	Usage
1 to **255** (DEFINITY CSI)	Enter the number.

Number

This display-only field corresponds to the **BSR Application** field on the Vector Directory Number screen.

Status Poll VDN

This field specifies the AAR or ARS pattern that routes over an IP trunk. The status poll vector on the remote Avaya server compares resources on that server and replies to the origin server with information on the best of these. Each remote Avaya server in a given application has to have a dedicated status poll vector.

Valid entries	Usage
0 to **9**, *****, **#**, **~p** (pause) **~w/~W** (wait) **~m** (mark) **~s** (suppress) or blank (DEFINITY CSI)	Specify the AAR or ARS pattern that routes over an IP trunk

Switch Node

Enter a number unique to the switch in a network of switches.

Valid entries	Usage
1 to **32767** or blank (DEFINITY CSI)	This number is an important part of the UCID tag and must be unique to the server running Communication Manager.

Bulletin Board

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

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

Field descriptions for page 1

Figure 31: Bulletin Board screen

```
change bulletin-board                                          Page 1 of x

              Message (* indicates high-priority)                   Date
          *Avaya is in the process of                             03/02/93
          *investigating your trunk lockup problem.               03/02/93
          *The Bulletin Board will be updated as                  03/02/93
          *we find information.                                   03/02/93
          *  We have identified the problem.                      03/04/93
          *The trunk you added does not provide                   03/04/93
          *disconnect supervision. However, the                   03/04/93
          *trunk group was administered as such.                  03/04/93
          *Please call Pat J. for details.                        03/04/93
          We recently added a new trunk group (14)                03/02/93
          and have had many of the members getting                03/02/93
          locked up.                                              03/02/93
          We see the error - thanks for checking.                 03/05/93
```

Date

This display-only field contains the date the information was entered or last changed.

Lines 1 through 10

These lines are reserved for high priority messages and are noted with an asterisk (*) in the first column on the left. If you have an *init* or *inads* login you can enter high-priority information to trigger the high-priority message at login time.

Valid entries	Usage
A to **Z** **a** to **z** Blank **0** to **9** !@#$%^&*()_+-=[]{}|\'˜;:’,"<.>/?	Enter any information.

Lines 11 through 19

These lines can be used by anyone with access.

<table>
<tr><th>Valid entries</th><th>Usage</th></tr>
<tr><td>A to Z
a to z
Blank
0 to 9
!@#$%^&*()_+-=[]{}|\’˜;:’,"<.>/?</td><td>Enter any information.</td></tr>
</table>

Field descriptions for pages 2 and 3

Date

This display only field contains the date the information was entered or last changed.

Lines 1 through 20

These lines can be used by anyone with access.

<table>
<tr><th>Valid entries</th><th>Usage</th></tr>
<tr><td>A to Z
a to z
Blank
0 to 9
!@#$%^&*()_+-=[]{}|\’˜;:’,"<.>/?</td><td>Enter any information.</td></tr>
</table>

Button Type Customization Restrictions

Use this screen to restrict button label customization of up to 50 specified button types for users who are not considered to be VIP users. This helps you to manage the usage of your system’s allocation of customized button labels to ensure that VIP users have the button label customization resource available to them.

Field descriptions for page 1

Figure 32: Button Type Customization Restrictions screen

```
 change button-restriction                                        Page 1 of x

                        Button Type Customization Restrictions

Restrict Customization Of Button Types? y

Restrict Customization Of Labels For The Following Button Types:

   abr-spchar  ~p ___________  ___________  ___________  ___________
   abrv-dial   s  ___________  ___________  ___________  ___________
   ___________    ___________  ___________  ___________  ___________
   ___________    ___________  ___________  ___________  ___________
   ___________    ___________  ___________  ___________  ___________
   ___________    ___________  ___________  ___________  ___________
   ___________    ___________  ___________  ___________  ___________
   ___________    ___________  ___________  ___________  ___________
   ___________    ___________  ___________  ___________  ___________
   ___________    ___________  ___________  ___________  ___________

Caution: Adding new button types to this list will cause all button labels
         associated with that button type to be lost
```

Restrict Customization Of Button Types

Use this field to enable/disable restriction of feature button label customization.

Valid entries	Usage
y	Enter **y** to enable restriction of feature button label customization. This is the default.
n	When you enter **n** in this field, users have the capability to customize labels for all buttons on their telephones.

Restrict Customization Of Labels For The Following Button Types

This field appears when **Restrict Customization of Button Types** is **y**.

Valid entries	Usage
valid button type from the list of entries	Enter the button type you want to restrict from label customization. **Note:** When you enter the special button types **abr-spchar** or **abrv-dial**, an additional field appears to the right of the button type as shown in Figure 32. Use this special field to specify the special character associated with the **abr-spchar** button type or the Abbreviated Dialing List associated with the **abrv-dial** button type.

Call Type Digit Analysis Table

Use the Call Type Digit Analysis Table **`(change calltype analysis)`** to tell Communication Manager how to modify telephone numbers dialed from a telephone's call log from internal contacts, or a corporate directory. There must be at least one entry in the Call Type Digit Analysis Table for Call Type Digit Analysis to take place. Call Type Digit Analysis allows users to automatically place outgoing calls based on the telephone number information in the phone's call log, without the user having to modify the telephone number.

Field descriptions for page 1

Figure 33: Call Type Digit Analysis Table screen

```
change calltype analysis                                          Page 1 of x
                         CALL TYPE DIGIT ANALYSIS TABLE
                                      Location:   all
    Dialed String            Delete Insert     Type   Delete Insert       Type
Match: _____________________ 1: ___  __________      2: ___  ___________
Length:Min    Max      __    3: ___   _________      4: ___   ____________
Match: _____________________ 1: ___  __________      2: ___  ___________
Length:Min    Max      __    3: ___   _________      4: ___   ____________
Match: _____________________ 1: ___  __________      2: ___  ___________
Length:Min    Max      __    3: ___   _________      4: ___   ____________
Match: _____________________ 1: ___  __________      2: ___  ___________
Length:Min    Max      __    3: ___   _________      4: ___   ____________
Match: _____________________ 1: ___  __________      2: ___  ___________
Length:Min    Max      __    3: ___   _________      4: ___   ____________
Match: _____________________ 1: ___  __________      2: ___  ___________
Length:Min    Max      __    3: ___   _________      4: ___   ____________
Match: _____________________ 1: ___  __________      2: ___  ___________
Length:Min    Max      __    3: ___   _________      4: ___   ____________
Match: _____________________ 1: ___  __________      2: ___  ___________
Length:Min    Max      __    3: ___   _________      4: ___   ____________
```

Location

This field is display-only. Its value is copied from the location specified in the command line, or if no location is entered, displays **all**.

Valid entries	Usage
numeric value all	Phones dialing from this location use the entries on this screen. If there are matching entries in the telephone's location, those entries are used. If there are no matching entries in the phone's location, Communication Manager tries the entries in location **all**.

Dialed String Match

Valid entries	Usage
numeric value x X Blank	Communication Manager compares this digit string to the original digit string, looking for a match to complete analysis and routing. **x**, **X**= wildcard digits. Use to match anything, as in ARS analysis administration. Blank=default. Cannot be blank if other fields on the row pair contain data.

Dialed String length (Min, Max)

Valid entries	Usage
numeric value	Communication Manager compares digit strings of this length to the original digit string, looking for a match to complete analysis and routing.

Delete

Valid entries	Usage
numeric value	Communication Manager deletes this number of digits in the original digit string, from the left-hand side of the original digit string, to complete analysis and routing.

Insert

Valid entries	Usage
numeric value	Communication Manager inserts these digits into the left-hand side of the original digit string to complete analysis and routing.

Type

Valid entries	Usage
aar ars ext udp	Administered call type for this dialed string. Communication Manager tests the modified digit string against the administered call type. **aar** = Automatic Alternate Routing, digit analysis algorithm commonly used for private network calls. **ars** = Automatic Route Selection, digit analysis algorithm commonly used for public network calls. **ext** = extension entries in the dialplan analysis tables of type **ext**. **udp** = extension entries in the uniform-dialplan tables.

Call Vector

This screen programs a series of commands that specify how to handle calls directed to a Vector Directory Number (VDN). See *Avaya Aura™ Call Center 5.2 Call Vectoring and Expert Agent selection (EAS) Reference*, 07-600780, for additional information.

Note:
If the **Call Center Release** field is set to **4.0** or later, the Call Vector screen includes additional pages to support up to 99 vector steps.

Field descriptions for page 1

Figure 34: Call Vector screen

```
 change vector nnnn                                              Page 1 of x
                                CALL VECTOR
   Number: nnnn           Name: _______________

Multimedia? n     Attendant Vectoring? n     Meet-me Conf? y            Lock? n
     Basic? y  EAS? n   G3V4 Enhanced? y    ANI/II-Digits? y   ASAI Routing? n
 Prompting? y  LAI? n  G3V4 Adv Route? y    CINFO? y   BSR? n      Holidays? n

  01 ____________
  02 ____________
  03 ____________
  04 ____________
  05 ____________
  06 ____________
  07 ____________
  08 ____________
  09 ____________
  10 ____________
  11 ____________

                    Press Esc f6 for Vector Editing
```

01 through XX

Enter vector commands as required (up to the maximum allowed in your configuration). For more information, see *Avaya Aura™ Call Center 5.2 Call Vectoring and Expert Agent selection (EAS) Reference*, 07-600780.

.

Valid entries	Usage
adjunct routing	Causes a message to be sent to an adjunct requesting routing instructions based on the CTI link number.
announcement	Provides the caller with a recorded announcement.
busy	Gives the caller a busy signal and causes termination of vector processing.
check	Checks the status of a split (skill) for possible termination of the call to that split (skill).
	1 of 2

Valid entries	Usage
collect	Allows the user to enter up to 16 digits from a touch-tone telephone, or allows the vector to retrieve Caller Information Forwarding (CINFO) digits from the network.
consider	Defines the resource (split, skill, or location) that is checked as part of a Best Service Routing (BSR) consider series and obtains the data BSR uses to compare resources.
converse-on	Delivers a call to a converse split (skill) and activates a voice response script that is housed within a Voice Response Unit (VRU).
disconnect	Ends treatment of a call and removes the call from the server running Communication Manager. Also allows the optional assignment of an announcement that plays immediately before the disconnect.
goto	Allows conditional or unconditional movement (branching) to a preceding or subsequent step in the vector.
messaging	Allows the caller to leave a message for the specified extension or the active or latest VDN extension.
queue-to	Unconditionally queues a call to a split or skill and assigns a queueing priority level to the call in case all agents are busy.
reply-best	Used only in status poll vectors in multi-site Best Service Routing applications, where it "returns" best data for its location to the primary vector on the origin server.
return	Returns vector processing to the step following the **goto** command after a subroutine call has processed.
route-to	Routes calls either to a destination that is specified by digits collected from the caller or an adjunct (route-to digits), or routes calls to the destination specified by the administered digit string (route-to number).
set	Performs arithmetic and string operations and assigns values to a vector variable or to the digits buffer during vector processing.
stop	Halts the processing of any subsequent vector steps.
wait-time	Delays the processing of the next vector step if a specified delay time is included in the command's syntax. Also provides feedback (in the screen of silence, ringback, or music) to the caller while the call advances in queue.
	2 of 2

ANI/II-Digits

A display-only field indicating whether you can use ANI and II-Digits Vector Routing Commands. ANI/II-Digits Routing requires that the **G3V4 Enhanced** field be **y**.

ASAI Routing

A display-only field indicating whether, on the System Parameters Customer-Options (Optional Features) screen, the **CallVisor Adjunct/Switch Applications Interface (ASAI Link Core Capabilities)** field is **y**.

Attendant Vectoring

This field appears only if, on the System Parameters Customer-Options (Optional Features) screen, the **Attendant Vectoring** field is **y**. **Attendant Vectoring** and **Meet-me Conference** cannot be enabled at the same time. Use this field to indicate attendant vectoring. If **Basic Vectoring** and **Vector Prompting** are both set to **n**, then the **Attendant Vectoring** field defaults to **y** and no changes are allowed to the field. When attendant vectoring is indicated for VDNs and vectors, all call center-associated fields (such as Skills and BSR) are removed.

Valid entries	Usage
y	Enter **y** so the vector is an attendant vector.
n	Default.

Basic

A display-only field indicating whether, on the System Parameters Customer-Options (Optional Features) screen, the **Vectoring (Basic)** field is **y**.

BSR

A display-only field indicating that on the System Parameters Customer-Options (Optional Features) screen, the **Vectoring (Best Service Routing)** field is **y**. Thus, you can use BSR commands and command elements in your vectors. An **n** indicates that the BSR option is not enabled.

CINFO

A display-only field indicating whether, on the System Parameters Customer-Options (Optional Features) screen, the **Vectoring (CINFO)** field is **y**.

EAS

A display-only field indicating whether, on the System Parameters Customer-Options (Optional Features) screen, the **Expert Agent Selection (EAS)** field is **y**.

Note:
When **Expert Agent Selection (EAS)** field is **y**, the help messages and error messages associated with this screen reflects a terminology change from “Split” to “Skill.” In addition, the vector commands entered also are affected by this terminology change (for example, *check backup split* becomes *check backup skill* when EAS is enabled).

G3V4 Adv Route

A display-only field indicating whether you can use the G3V4 Advanced Vector Routing commands.

G3V4 Enhanced

A display-only field indicating whether you can use G3V4 Enhanced Vector Routing commands and features.

Holidays

A display-only field that appears when, on the screen, the **Vectoring (Holidays)** field is **y**.

LAI

A display-only field indicating whether **Look-Ahead Interflow** is enabled.

Lock

This field controls access to the vector from Avaya CentreVu products.

Note:
Always lock vectors that contain secure information (for example, access codes).

Valid entries	Usage
y	You do not want this vector to be accessible to these client programs. Locked vectors can only appear and be administered through the SAT or a terminal emulator. If **Meet-me Conference** is **y**, the **Lock** field also must be **y**.
n	Gives CentreVu CMS and CentreVu Control Center users the ability to administer this vector from these client programs.

Meet-me Conf

This field appears only if, on the System Parameters Customer-Options (Optional Features) screen, the **Enhanced Conferencing** field is **y**. This field designates the VDN as a Meet-me Conference VDN.

Valid entries	Usage
y/n	Enter **y** to enable Meet-me Conference for this vector. If **Meet-me Conference** is **y**, the **Lock** field also must be **y**. When the **Lock** field is **y**, the vector cannot be changed by adjunct vectoring programs such as Visual Vectors. **Attendant Vectoring** and **Meet-me Conference** cannot be enabled at the same time.

Multimedia

Indicates whether the vector should receive early answer treatment for multimedia calls. This only applies if the **Multimedia Call Handling** field is **y**. This field does not appear for S87XX Series IP-PNC.

Valid entries	Usage
y/n	If you expect this vector to receive multimedia calls, set this field to **y**. If this value is **y**, the call is considered to be answered at the start of vector processing, and billing for the call starts at that time.

Name

Represents the vector name.

Valid entries	Usage
Up to 27 alphanumeric characters. Up to 15 alphanumeric characters (for S8300, S8400, S87XX IP-PNC Servers only)	This is an optional field. If ~r can be used to activate Network Call Redirection if, on the System Parameters Customer-Options (Optional Features) screen, the **ISDN Network Call Redirection** field is **y**. **Note:** For 4610SW, 4620SW, 4621SW, and 4622SW, Sage, Spark, and 9600-series Spice telephones, the **Name** field has an associated optional native name field that is supported by the Unicode language display. The native name field is accessible through the Integrated Management Edit Tools such as Avaya Site Administration (ASA). Unicode is also an option for the 2420J telephone when **Display Character Set** on the **System Parameters Country-Options** screen is **katakana**. For more information on the 2420J, see *2420 Digital Telephone User's Guide*, 555-250-701. Avaya BRI stations support ASCII characters only. They do not support non-ASCII characters, such as Eurofont or Kanafont. Therefore, if you use non-ASCII characters in any Communication Manager **Name** field, such characters do not display correctly on a BRI station.

Number

Represents the vector number. A display-only field when the screen is accessed using a **change** or **display** administration command.

Prompting

A display-only field indicating whether, on the System Parameters Customer-Options (Optional Features) screen, the **Vectoring (Prompting)** field is **y**.

CAMA Numbering Format

This screen administers the Centralized Automatic Message Accounting (CAMA) trunks and provides Caller's Emergency Service Identification (CESID) information to the local community's Enhanced 911 system through the local tandem office.

This screen provides the CESID format by extension number or number blocks. This allows for multiple CESID formats to be sent over multiple CAMA trunk groups allowing for mixed station numbering plans and some limited conversion from non-DID to DID numbers typically required by the Private Switch/Automatic Location Interface (PS/ALI) database.

The default CESID defines the CESID for all extensions that are not defined in the **Ext Code** field.

There are 446 CESID entries over 15 pages. The first page contains the Default CESID and 26 extensions to CESID entries. The second through fifteenth pages each contain 30 extensions to CESID entries.

Field descriptions for page 1

Figure 35: CAMA Numbering Format screen

```
change cama-numbering                                                Page 1 of x
                         CAMA NUMBERING - E911 FORMAT

System CESID Default: _____________

Ext Ext                             Total
Len Code             CESID          Length
__  _______________  ________________ __
__  _______________  ________________ __
__  _______________  ________________ __
__  _______________  ________________ __
__  _______________  ________________ __
__  _______________  ________________ __
__  _______________  ________________ __
__  _______________  ________________ __
__  _______________  ________________ __
__  _______________  ________________ __
```

CESID

Valid entries	Usage
1 to 11 digits or blank 1 to 16 digits or blank (S8300, S87XX IP-PNC Servers)	Enter the number that is used to identify the calling terminal within an emergency service system. This field can represent a prefix to an extension or the entire CESID.

Ext Code

Valid entries	Usage
Leading extension digits or blank	Enter the leading digits or all of the digits in the extension for the specified CESID. If the extension length is greater than the number of digits in the extension code, the extension code is interpreted as a block of digits. For example, if the extension length is 4 and the extension code is 11, the CESID serves extensions 1100 through 1199. The Ext Code 11 is for a DID block. An Ext Code of 126 might point a non-DID block to a nearby DID extension 5241666.

Ext Len

Valid entries	Usage
1 to **13** or blank	Enter the number of digits in the extension.

System CESID Default

Valid entries	Usage
1 to 16 digits	Enter a default CESID. This number is sent over the CAMA trunk if the **Ext Code** field does not have an entry.

Total Length

Valid entries	Usage
1 to **16** or blank	Enter the total number of digits to send.

Capacities

The **System Capacity** screen (command `display capacity`) is described in *Maintenance Commands for Avaya Communication Manager, Media Gateways and Servers*, 03-300431. Detailed system capacity information can be found in *Avaya Aura™ Communication Manager System Capacities Table*, 03-300511.

CDR System Parameters

Use the Call Detail Recording (CDR) System Parameters screen to set parameters for the types of calls you want to record and how to format the information. You can use CDR records to determine call costs, diagnose problems, detect abuse, and optimize your network.

Field descriptions for page 1

Figure 36: CDR System Parameters screen

```
change system-parameters cdr                                   Page   1 of   x
                           CDR SYSTEM PARAMETERS

 Node Number (Local PBX ID):                      CDR Date Format: month/day
      Primary Output Format:
    Secondary Output Format:
           Use ISDN Layouts? n                  Enable CDR Storage on Disk? n
       Use Enhanced Formats? n      Condition Code 'T' For Redirected Calls? n
      Use Legacy CDR Formats? y                Remove # From Called Number? n
Modified Circuit ID Display? n                            Intra-switch CDR? n
                  Record Outgoing Calls Only? n     Outg Trk Call Splitting? y
  Suppress CDR for Ineffective Call Attempts? y       Outg Attd Call Record? y
      Disconnect Information in Place of FRL? n      Interworking Feat-flag? n
 Force Entry of Acct Code for Calls Marked on Toll Analysis Form? n
                                    Calls to Hunt Group - Record: member-ext
Record Called Vector Directory Number Instead of Group or Member? n
Record Agent ID on Incoming? n       Record Agent ID on Outgoing? y
     Inc Trk Call Splitting? n
  Record Non-Call-Assoc TSC? n           Call Record Handling Option: warning
      Record Call-Assoc TSC? n   Digits to Record for Outgoing Calls: dialed
   Privacy - Digits to Hide: 0               CDR Account Code Length: 15
```

Calls to Hunt Group — Record

Valid entries	Usage
member-ext	Enter **member-ext** to record the extension of the telephone or data terminal where the call terminated.
group-ext	Enter **group-ext** to record the extension that was dialed.

CDR Account Code Length

Valid entries	Usage
1 to **15**	Enter the number of digits to record when a user enters an account code. For some record formats, a long account code overwrites spaces on the record that are usually assigned to other fields.

CDR Date Format

Use this field to select the format for the date stamp that begins each new day of call records.

Valid entries	Usage
month/day **day/month**	Choose the format that is most appropriate for your situation. If your company has many different sites, you might need to use the same format as the other locations.

Condition Code 'T' for Redirected Calls

You can elect to identify CDR records of calls that have been redirected automatically off the server running Communication Manager.

Valid entries	Usage
y	The Condition Code of both CDR records for the call is 'T.'
n	The Condition Codes normally associated with the **Record Outgoing Calls Only** field are generated.

Digits to Record for Outgoing Calls

Valid entries	Usage
dialed	Use **dialed** to record the digits a user actually dials.
outpulsed	Use **outpulsed** to record the digits that Communication Manager actually sends out over the trunk, *including any additions or deletions that take place during routing*.

Disconnect Information in Place of FRL

Valid entries	Usage
y	Enter **y** to replace the **Facility Restriction Level (FRL)** field with information about why a call disconnects.
n	Enter **n** to record the call's FRL.

Enable CDR Storage on Disk

Valid entries	Usage
y/n	Enter **y** to enable the Survivable CDR feature. Default is **n**.

Force Entry of Acct Code for Calls Marked on Toll Analysis Form

Specifies whether an account code is required when making a toll call. This will not necessarily be all chargeable calls and it might even include some non-chargeable calls.

Valid entries	Usage
y	Enter **y** to deny all toll calls unless the user dials an account code. Forced Entry of Account Codes must be **y** on the System Parameters Customer-Options (Optional Features) screen.
n	Enter **y** to allow calls without an account code. *This does not override other calling restrictions.*

Inc Attd Call Record

Appears when **Inc Trk Call Splitting** is **y**.

.

Valid entries	Usage
y/n	Enter **y** to enable separate recording of attendant portions of outgoing calls that are transferred or conferenced.

Inc Trk Call Splitting

Appears when the **Record Outgoing Calls Only** field on the System Parameters CDR screen is **n**.

Valid entries	Usage
y/n	Enter **y** to create separate records for each portion of incoming calls that are transferred or conferenced.

Interworking Feat-flag

See Call Detail Recording in *Avaya Aura™ Communication Manager Feature Description and Implementation,* 555-245-205, for more information.

Valid entries	Usage
y	Enter **y** if you want the feature flag to indicate interworked outgoing ISDN calls. *An interworked call is one that passed through more than one ISDN node.*
n	Enter **n** if you want the feature flag to indicate no answer supervision for interworked calls.

Intra-Switch CDR

Valid entries	Usage
y/n	Enter **y** to record calls within Communication Manager. If you choose this option, you must complete the **Intra-Switch CDR** screen to indicate which extensions should be monitored.

Modified Circuit ID Display

This affects the "printer," "teleseer," and "59-character" output formats.

Valid entries	Usage
y	Enter **y** to display the circuit ID in its actual format (100's, 10's, units). For example, circuit ID 123 displays as 123. *You might need to verify that your output device can accept this format.*
n	Enter **n** to display the circuit ID in its default format (10's, units, 100's). For example, circuit ID 123 appears as 231.

Node Number (Local PBX ID)

A display-only field indicating the DCS switch node number in a network of switches.

Outg Attd Call Record

Only appears if **Outg Trk Call Splitting** is **y**.

Valid entries	Usage
y/n	Enter **y** to enable separate recording of attendant portions of outgoing calls that are transferred or conferenced.

Outg Trk Call Splitting

See Call Splitting in *Avaya Aura™ Communication Manager Feature Description and Implementation,* 555-245-205, for more information.

Valid entries	Usage
y/n	Enter **y** to create separate records for each portion of outgoing calls that are transferred or conferenced.

Primary Output Endpoint

This field determines where the server running Communication Manager sends the CDR records, and is required if you specify a Primary Output Format.

Valid entries	Usage
eia	If you use the EIA port to connect the CDR device, enter **eia**.
Extension number	This is the extension of the data module (if used) that links the primary output device to the server running Communication Manager.
CDR1, CDR2	Use this value if the CDR device is connected over a TCP/IP link, and this link is defined as either CDR1 or CDR2 on the **IP Services** screen.

Primary Output Format

Controls the format of the call records sent to the primary output device.

Valid entries	Usage
customized	Use this option if you have special call accounting needs that standard record formats do not accommodate. If you use a customized record format, you need to have call accounting software that is also customized to receive these records. Consult with your call accounting vendor before using this option.
printer	Use **printer** if you are sending the call detail records to a printer rather than to a record collection or call accounting system.
59-char **expanded** **lsu** **lsu-expand** **int-direct** **int-isdn** **int-process** **teleseer** **unformatted**	The remaining formats are standard record formats. The one you use must be compatible with your call accounting software. Verify this through your vendor or the accounting system documentation.

Privacy — Digits to Hide

If you enable **CDR Privacy** on the Station screen for a given telephone, use this field to indicate how much of the dialed number to hide on the CDR record.

Valid entries	Usage
0 to **7**	Enter the number of digits to hide, counting from the end (right to left). For example, if you enter **4** in this field and the user dials 555-1234, only "555" would appear in the **Dialed Number** field of the CDR record.

Record Agent ID on Incoming

Only displays if the **Expert Agent Selection (EAS)** field is **y** on the **System Parameters Customer-Options (Optional Features)** screen. You cannot use both the **Called VDN** and the **Agent Login ID Instead of Group or Member**. Only one of these fields can be **y**.

Valid entries	Usage
y/n	Enter **y** to include the EAS agent's LoginID instead of the physical extension in the **Dialed Number** field of a CDR record.

Record Agent ID on Outgoing

Only displays if the **Expert Agent Selection (EAS)** field is **y** on the **System Parameters Customer-Options (Optional Features)** screen.

Valid entries	Usage
y/n	Enter **y** to include the EAS agent's LoginID instead of the physical extension in the **Dialed Number** field of a CDR record.

Record Call-Assoc TSC

Valid entries	Usage
y/n	Enter **y** to create records for call-associated temporary signaling connections. *If you have a lot of data connections this could increase the number of records. You might want to consider the capacity of your call collection device*.

Record Called Vector Directory Number Instead of Group or Member

If this option is enabled, the called VDN overrides the group or member information that normally appears in the **Dialed Number** field of the CDR record. If a call is directed through more than one VDN, the first VDN used for the call is stored. This applies only to calls routed to a hunt group by a vector, not to calls routed directly to an extension by a vector.

You cannot use both the **Called VDN** and the **Agent Login ID instead of Group or Member**. Only one of these fields can be **y**.

Valid entries	Usage
y/n	Enter **y** to include the Vector Directory Number (VDN) in the **Dialed Number** field of a CDR record.

Record Non-Call-Assoc TSC

A temporary signaling channel (TSC) is a virtual connection established within an ISDN D-channel.

Valid entries	Usage
y/n	Enter **y** to create records for non-call-associated temporary signaling connections. *If you have a lot of data connections this could increase the number of records. You might want to consider the capacity of your record collection device*.

Record Outgoing Calls Only

Valid entries	Usage
y	Enter **y** to record only outgoing calls. *This can save space if you are only concerned with charges for outbound calls.*
n	Enter **n** to record both outgoing and incoming calls.

Remove # From Called Number

Valid entries	Usage
y	Enter **y** to have the "#" (or "E") symbol removed from the **Dialed Number** field of the call detail record. *You might need to verify that your output device can accept this format.*
n	Enter **n** to have the trailing "#" (or "E") symbol appear in the **Dialed Number** field whenever inter-digit time out occurs or users dial # to indicate the end of dialing.

Secondary Output Endpoint

Appears when the **Secondary Output Format** field is administered.

Valid entries	Usage
eia	Use this if the secondary output device is connected to the eia port.
Extension number	This is the extension of the data module (if used) that links the secondary output device to the server running Communication Manager.
CDR1, CDR2	Use this value if the CDR device is connected over a TCP/IP link, and this link is defined as either CDR1 or CDR2 on the IP Services screen.

Secondary Output Format

Controls the format of the call records sent to the secondary output device.

CAUTION:
Only qualified (Avaya) service personnel should administer a secondary output device. This option might cause loss of data when the buffer contains large amounts of data.

Valid entries	Usage
customized **int-direct** **int-process** **lsu** **unformatted**	These are the only formats you can use for a secondary output device. The format must be compatible with your call accounting software. Verify this through your vendor or the accounting system documentation.

Suppress CDR for Ineffective Call Attempts

Ineffective call attempts are calls that are blocked because the user did not have sufficient calling privileges or because all outgoing trunks were busy. This includes the unavailable incoming or outgoing trunks due to trunk usage allocation for ISDN Call-by-Call Service Selection trunks, incoming calls rejected by Communication Manager due to NSF mismatch, and ISDN calls that did not complete at the far end, if a cause value was provided. These calls appear on the CDR record with a condition code "E."

Valid entries	Usage
y	Enter **y** to ignore ineffective call attempts. *Use this if you have limited storage space for CDR records and records often overrun the buffer.*
n	Enter **n** to report ineffective call attempts. *This can tell you if your users are often unable to place outgoing calls, or if a large number of incoming calls are not completed.* You can also use this if you need to have records of attempts to contact a client, and are using ISDN trunks. Using this option requires more space for records.

Use Enhanced Formats

Enhanced formats provide additional information about time in queue and ISDN call charges, where available. This affects the "expanded," "teleseer," "lsu," "printer," and "unformatted" output formats.

Valid entries	Usage
y/n	Enter **y** to enable the use of the Enhanced version of the specified primary output format. You cannot use Enhanced formats and ISDN formats at the same time.

Use ISDN Layouts

ISDN Layouts provide more accurate information about the inter-exchange carrier and ISDN network services used for a call. This affects "lsu" and "printer" output formats, as well as any format with ISDN layouts, such as "teleseer."

Valid entries	Usage
y/n	Enter **y** to enable the use of the ISDN version of the specified primary output format. You cannot use ISDN formats and Enhanced formats at the same time.

Use Legacy CDR Formats

Use this field to specify the use of pre-Communication Manager 4.0 (“legacy”) Call Detail Recording (CDR) formats in the CDR records the system produces, instead of the formats used in Communication Manager 4.0 and later. Listed below are the CDR formats that are impacted by the **Use Legacy CDR Formats** field. All other CDR formats remain unchanged.

CDR Format	Communication Manager 3.1 and earlier length	Communication Manager 4.0 and later length
ISDN Teleseer	80	82
Enhanced Teleseer	81	83
ISDN Printer	84	86
Enhanced Printer	85	87
ISDN LSU	59	61
Enhanced LSU	59	61
Expanded	135	139
Enhanced Expanded	151	155
Unformatted	105	109
Enhanced Unformatted	119	123
Int-ISDN	136	140

Valid entries	Usage
y	Enter **y** to use pre-Communication Manager 4.0 (“legacy”) CDR formats for CDR records. Default is **y**.
n	Enter **n** to use CDR formats for Communication Manager 4.0 and later. When this field is set to **n**, the **INS** field in the CDR records is increased from three to five characters, and the **Attendant Console** field is increased from two to four characters.

Field descriptions for page 2

This page appears only if the **Primary Output Format** field is **customized**.

Figure 37: CDR System Parameters screen

```
change system-parameters cdr                                    Page 2 of x
                             CDR SYSTEM PARAMETERS
      Data Item - Length           Data Item - Length       Data Item - Length
 1: time____________ - 4_  17: ________________ - __  33: ______________ - __
 2: space___________ - 1_  18: ________________ - __  34: ______________ - __
 3: duration________ - 4_  19: ________________ - __  35: ______________ - __
 4: return__________ - 1_  20: ________________ - __  36: ______________ - __
 5: line-feed_______ - 1_  21: ________________ - __  37: ______________ - __
 6: ________________ - __  22: ________________ - __  38: ______________ - __
 7: ________________ - __  23: ________________ - __  39: ______________ - __
 8: ________________ - __  24: ________________ - __  40: ______________ - __
 9: ________________ - __  25: ________________ - __  41: ______________ - __
10: ________________ - __  26: ________________ - __  42: ______________ - __
11: ________________ - __  27: ________________ - __  43: ______________ - __
12: ________________ - __  28: ________________ - __  44: ______________ - __
13: ________________ - __  29: ________________ - __  45: ______________ - __
14: ________________ - __  30: ________________ - __  46: ______________ - __
15: ________________ - __  31: ________________ - __  47: ______________ - __
16: ________________ - __  32: ________________ - __  48: ______________ - __
                               Record length = 11
```

Data Item

Enter the data items in the order they should appear on the customized record. Only use this screen if you have arranged with your vendor to customize your call accounting system to receive these records.

You must include at least one field in order to have a record. See the table below for valid entries. The last two data items in a the record must be **line-feed** and **return**, in that order.

For more information, see Call Detail Recording in *Avaya Aura™ Communication Manager Feature Description and Implementation,* 555-245-205*.*

Data Item	Length	Data Item	Length
acct-code	15	ixc-code	4
attd-console	2	line-feed	1
auth-code	7	location-from	3
bandwidth	2	location-to	3
bcc	1	in-trk-code	4
calling-num	15	ma-uui	1
			1 of 2

Data Item	Length	Data Item	Length
clg-pty-cat	2	node-num	2
clg-num/in-tac	10	null	1
code-dial	4	out-crt-id	3
code-used	4	ppm	5
cond-code	1	res-flag	1
country-from	3	return	1
country-to	3	sec-dur	5
dialed-num	23	space	1
duration	4	time	4
feat-flag	1	timezone-from	3
frl	1	timezone-to	6
in-crt-id	3	tsc_ct	4
ins	3	tsc_flag	1
isdn-cc	11	vdn	5
			2 of 2

Length

Enter the length of each data item, if different from the default.

Valid entries	Usage
The maximum record length depends on the call accounting system you use. Check with your vendor.	The date field should be six-digits to ensure proper output. Certain fields default to the required length.

Record Length

Display-only field indicating the accumulated total length of the customized record, updated each time the length of a data item changes.

Change Station Extension

This screen allows an administrator to change extensions on the switch from one extension to another all at once. When the screen is filled out and submitted, all administration that was associated with the current extension is now associated with the new extension. Any administration references of the extension being changed, such as references used in a vector, coverage, etc., is now reference the new extension. Once the extension has been changed, all references to the previous extension is removed from the switch.

If an extension is changed that is also administered on an adjunct (such as voice mail or an ASAI link), the extension on the adjunct must also be changed to ensure proper functionality.

Note:
A forwarded extension administered as a button is not handled by the **`change extension-station`** ***`xxxxxxx`*** command. It is recommended that the administrator use the **`list usage`** command prior to changing any extensions.

Field descriptions for page 1

Figure 38: Change Station Extension screen

```
change extension-station xxxxxxx                                Page 1 of x

                     CHANGE STATION EXTENSION

           Station Name: xxxxxxxxxxxxxxxxxxxxxxxxxxxxxxx     Port: xxxxxx

                                FROM EXTENSION                 TO EXTENSION
                                --------------                 -------------
                        Station: xxx-xxxx                        xxx-xxxx
                   Message Lamp: xxx-xxxx                        xxx-xxxx
         Emergency Location Ext.: xxx-xxxx                        xxx-xxxx
IP Parameter Emergency Location: xxx-xxxx                 See IP-Network Map Form
```

Note:
You cannot use the **`change extension-station`** command to change the extension of a station if that station is administered as the emergency location extension for another station. For example, if station A is administered as the emergency location extension for station B, then:

- You cannot change the extension of station A using the **`change extension-station`** command unless you first change station B to assign a different emergency location extension.

- You can change the extension of station B. If you do, the Change Station Extension screen displays station A's extension in the **Emergency Location Ext.** field under the **From Extension** header.

Emergency Location Extension

The Emergency Location Extension from the Station screen associated with the current extension is displayed under **From Extension**.

Valid entries for "To Extension"	Usage
0 to **9**	Type a new extension for the **Emergency Location Ext**. field that appears on the Station screen, any valid and assigned extension number for your dial plan.

IP Parameter Emergency Location

The Emergency Location Extension from the IP Address Mapping screen associated with the current extension is displayed under **From Extension**.

Valid entries for "To Extension"	Usage
n/a	The words **See Ip-network Map Form** display. The administrator can only change this field on the IP Address Mapping screen (using the `ip-network-map` command).

Message Lamp

The Message Lamp Extension associated with the current extension is displayed under **From Extension**.

Valid entries for "To Extension"	Usage
0 to **9**	Type a new extension for the **Message Lamp Ext.** field, any valid and assigned extension number for your dial plan.

Port

This field is read only, and displays the port of the existing extension.

Station

The current extension that is being changed (the extension that was typed in the **`change extension-station`** ***`xxxxxxx`*** command) is displayed under **From Extension**.

Valid entries for "To Extension"	Usage
0 to **9**	Type the new extension that you want the current extension changed to, any valid and assigned extension number for your dial plan.

Station Name

This field is read only, and displays the name of the existing extension (the extension that was typed in the **`change extension-station`** ***`xxxxxxx`*** command).

Note:
Avaya BRI stations support ASCII characters only. They do not support non-ASCII characters, such as Eurofont or Kanafont. Therefore, if you use non-ASCII characters in any Communication Manager **Name** field, such characters do not display correctly on a BRI station.

Circuit Packs

This screen is described in *Maintenance Commands for Avaya Aura™ Communication Manager, Media Gateways and Servers*, 03-300431.

Class of Restriction

Use this screen to establish classes of restriction (COR). Classes of restriction control call origination and termination. Your system might use only one COR or as many as necessary to control calling privileges. You can assign up to 995 different CORs.

Consider the following to enhance your system security:

1. Assign a separate COR to incoming and outgoing trunk groups, then restrict calling between the two groups.
2. Limit the calling permissions as much as possible by setting appropriate Calling Party Restrictions and Facility Restriction Levels (FRLs).

Field descriptions for page 1

Figure 39: Class of Restriction screen (page 1)

```
change cor n                                                Page 1 of x
                             CLASS OF RESTRICTION
               COR Number: n
          COR Description: supervisor

                      FRL: 0                                    APLT? y
  Can Be Service Observed? n              Calling Party Restriction: none
Can Be A Service Observer? y               Called Party Restriction: none
 Partitioned Group Number: 1          Forced Entry of Account Codes? n
        Priority Queuing? n                     Direct Agent Calling? y
     Restriction Override: none             Facility Access Trunk Test? n
     Restricted Call List? n                     Can Change Coverage? n
   Unrestricted Call List? __ __ __ __ __ __ __ __ __ __
            Access to MCT? y                 Fully Restricted Service? n
Group II Category For MFC: 7                 Hear VDN of Origin Annc.? n
         Send ANI for MFE? n_                 Add/Remove Agent Skills? y
            MF ANI Prefix: ______            Automatic Charge Display? n
Hear System Music on Hold? y     PASTE(Display PBX Data on telephone)? n
                             Can Be Picked Up By Directed Call Pickup? n
                                         Can Use Directed Call Pickup? n
                                         Group Controlled Restriction: inactive
```

Access to MCT?

This field refers to Malicious Call Trace.

Valid entries	Usage
y	Enter **y** to allow permissions to activate a request to trace a malicious call.
n	Entering **n** prohibits this user from requesting a malicious call trace, but does not prevent this extension from appearing in the MCT History report, should this extension be the subject of a malicious call trace.

Add/Remove Agent Skills

Valid entries	Usage
y/n	Enter **y** to allow users with this COR to add and remove skills.

APLT

Valid entries	Usage
y/n	Enter **n** to allow access to APLT trunk group Enhanced Private Switched Communications System (EPSCS) or Common Control Switched Arrangement (CCSA) off-net facilities. If fully restricted service is enabled, set this field to **n**.

Automatic Charge Display

Shows the cost of an active outgoing call using Periodic Pulse Metering (PPM) or ISDN Advice of Charge (AOC) on Digital Communications Protocol (DCP) or Avaya BRI stations. Not available in the U.S

.

Valid entries	Usage
y	Displays call charges during and at the end of the call.
n	Call charges can be seen if users press the **disp-chrg** button before the call drops.

Called Party Restriction

Valid entries	Usage
Inward	Blocks the calling party from receiving incoming exchange network calls, attendant originated calls, and attendant completed calls.
Manual	Blocks the called party from receiving all calls except for those originated or extended by the attendant.
Public	Blocks the called party from receiving public network calls. Attendant calls are allowed to go through to the called party as well as attendant-assisted calls if the **Restriction Override** field in the public restricted station's COR is **attd** or **all**.

Valid entries	Usage
Termination	Blocks the called party from receiving any calls at any time.
none	No called party restrictions.

Calling Party Restriction

This field determines the level of calling restriction associated with this COR.

Note:
To enhance system security, limit calling permissions as much as possible.

Valid entries	Usage
Origination	Blocks the calling party from originating a call from the facility at any time. The party can only receive calls. A telephone with this COR can initiate Remote Access calls, if the COR of the barrier code allows it.
Outward	Blocks the calling party from calling outside the private network. Users can dial other users on the same server running Communication Manager or within a private network. To enhance security, Avaya recommends that you use outward restrictions when practical.
All-toll	Blocks the calling party from making ARS and trunk access calls from a facility assigned the COR to certain toll areas as defined in the **Dialed String** field on the Toll Analysis screen. The **Dialed String** field must be marked as being associated with the system's Toll List. The call completes if the facility's COR also is associated with an Unrestricted Call List and whose **Dialed String** field also matches the dialed number.
Tac-toll	Blocks the calling party from making trunk access calls from the facility assigned the COR to certain toll areas as defined in the **Dialed String** field on the Toll Analysis screen. The **Dialed String** field must be marked as being associated with the system's Toll List. The call completes if the facility's COR also is associated with an Unrestricted Call List and whose **Dialed String** field also matches the dialed number. See Toll Analysis on page 961 for additional information.
none	No calling party restrictions.

Can Be a Service Observer

If you want an observer to observe users, set the users' CORs to **y** on the observer's COR Service Observing Permission table.

SECURITY ALERT:
The use of Service Observing features might be subject to federal, state, or local laws, rules, or regulations; or require the consent of one or both of the parties to the conversation. Customers should familiarize themselves with and comply with all applicable laws, rules, and regulations before using these features.

Note:
You cannot enter **y** in the previous two fields unless **Service Observing (Basic)** is enabled on the System Parameters Customer-Options (Optional Features) screen.

Valid entries	Usage
y/n	Enter **y** if users with this COR can service observe other users.

Can Be Picked Up By Directed Call Pickup

Valid entries	Usage
y/n	Enter **y** to allow this Station's or EAS agent's calls to be picked up by using the Directed Call Pickup Up feature. Before you can set this field to **y**, you must set **Directed Call Pickup** on the Feature-Related System Parameters screen to **y**.

Can Be Service Observed

Note that this field allows or denies service observing for not only physical extensions, but also for logical agent IDs and VDNs. If you want an observer to observe users, set the users' CORs to **y** on the observer's COR Service Observing Permission table

Valid entries	Usage
y/n	Enter **y** if users with this COR can be service observed.

Can Change Coverage

Valid entries	Usage
y/n	Enter **y** to allow station users with this COR to select one of two previously administered coverage paths or to activate, change, or deactivate call forward all calls or call forward busy/don't answer from any on-site or off-site location.

Can Use Directed Call Pickup

Valid entries	Usage
y/n	Enter **y** to allow the station, attendant, or EAS agent to pick up calls using the Directed Call Pickup feature. Set **Directed Call Pickup** on the Feature-Related System Parameters screen to **y** to set this field to **y**.

COR Description

Valid entries	Usage
Up to 35 characters	Enter a description of the COR that indicates how you use it. If you make this as clear as possible (for example, Customer Service, Legal Department), it is easier to remember which COR to assign when you add users.

COR Number

This is a display-only field when the screen is accessed via an administration command such as **`change`** or **`display`**. Displays the COR number.

Direct Agent Calling

Valid entries	Usage
y/n	If this is **y**, users can dial an ACD agent’s extension directly, rather than anyone in the agent pool. If the system is in Night Service, the call routes to the Night Service extension. If the extension with this COR belongs to an agent, the agent can receive calls directly.

Facility Access Trunk Test

An associated feature button (**trk-ac-alm**) status lamp lights when a successful test attempt occurs. Pressing one of the alarm buttons (ten maximum) when its associated status lamp is lit turns off all lamps on all buttons whether the access is still in progress or has completed

Valid entries	Usage
y/n	Enter **y** to allow users with this COR to perform Facility Access Trunk Tests.

Forced Entry of Account Codes

FEAC must be enabled on the System Parameters Customer-Options **(Optional Features)** screen and on the CDR System Parameters screen.

Note:
If a COR requiring entry of account codes is assigned a VDN, the route to commands executed by the associated vector cannot be successful.

Valid entries	Usage
y/n	Enter **y** to indicate that an account code must be dialed when making outgoing trunk calls. If this is **y**, any telephone assigned the associated COR must dial an account code before making an outgoing call. If you set this to y for a COR assigned to a trunk group, users must dial account codes before calling out over that trunk group. This might be useful for trunks used in international calls, and those that are more expensive. If a call is being routed by ARS, account code checking is not done on the COR.

FRL

Valid entries	Usage
0 to **7**	Enter an originating FRL number. AAR and/or ARS features use this entry to determine call access to an outgoing trunk group. Outgoing call routing is determined by a comparison of the FRLs in the AAR/ARS Routing Pattern and the FRL associated with the COR of the call originator (typically, a telephone user). An originating FRL of 0 has the least calling privileges. To enhance system security, assign the lowest possible FRL.

Fully Restricted Service

Note:
If this field is enabled, the **APLT** field must be **n**.

Valid entries	Usage
y/n	When **y** entered for a given COR, stations assigned that COR does not have access to the public network for either incoming or outgoing calls.

Group II Category For MFC

This field always controls categories for Russian signaling trunks. It can control categories for R2-MFC signaling trunks, depending on the value of the **Use COR for Calling Party Category** field on the Multifrequency-Signaling-Related System Parameters screen.

The Calling Party Category digit administered in this field is included as part of the ANI information sent to the Central Office on request using R2-MFC signaling

.

Valid entries	Usage
1 to **10**	Enter the value you want the server running Communication Manager to send as the Calling and/or Called Party Category for telephones or trunks that use this COR.

Group Controlled Restriction

A display-only field that determines if the current COR is under controlled restriction. This field can help troubleshoot problems by first checking its value.

.

Valid entries	Usage
active	indicates the COR is controlled restricted
inactive	indicates the COR is not controlled restricted

Hear System Music on Hold

Valid entries	Usage
y/n	If you enter **y**, the Music on Hold feature is activated on the telephone and all calls that the telephone user puts on hold will hear music.

Hear VDN of Origin Announcement

Valid entries	Usage
y/n	Enter **y** if users with this COR can receive VDN of Origin messages.

MF ANI Prefix

Defines the prefix to apply to an extension number when ANI is sent to the CO. This overrides any ANI prefix administered on the Multifrequency Signaling screen. This does not apply when ANI is tandemed through the Communication Manager server on tandem calls. This field also applies to the ANI for the server when the originating side is a trunk and there was no ANI.

Valid entries	Usage
1 to 7 digits or blank	If you want the entire number to display on the receiving end, enter all digits except the extension number.

Partitioned Group Number

This field appears only if **AAR/ARS Partitioning** is **y** and **Time of Day Routing** is **n** on the System Parameters Customer-Options **(Optional Features)** screen.

Valid entries	Usage
1 to **8**	Enter the AAR/ARS partitioned group number associated with this COR.

PASTE (Display PBX Data on telephone)

Valid entries	Usage
y/n	Enter **y** to download all lists. Enter **n** to disallow the PASTE feature.

Priority Queuing

Valid entries	Usage
y	Enter **y** to allow the telephone user’s calls to be placed ahead of non-priority calls in a hunt group queue
n	If you do not use Automatic Call Distribution (**ACD** is not enabled on the System Parameters Customer-Options (Optional Features) screen), this field must be **n**.

Restricted Call List

This list can be used whether the COR is toll restricted. The Restricted Call List (RCL) has priority over the Toll Analysis Unrestricted Call List (UCL). A call attempt from a facility assigned a COR (with **RCL** field set to **y**), whose dialed digit string is on the Toll Analysis screen and is marked as being associated with the RCL, is denied.

Valid entries	Usage
y/n	Enter **y** to specify that this COR will have access to the system's Restricted Call List (see Toll Analysis on page 961).

Restriction Override

Allows the specified users to bypass restriction on conference, transfer or call forwarding operations.

Valid entries	Usage
attendant	A telephone with a COR that is inward restricted cannot receive public network, attendant-originated, or attendant-extended calls. Enter **attendant** to give your attendants the ability to override this restriction.
all	Enter **all** if you want all of the users with this COR to override inward restrictions.
none	Enter **none** if you do not want any users of this COR to bypass the restrictions.

Send ANI for MFE

Only applicable for Spain. Valid for 2/6 signaling, but not 2/5 signaling. The following field appears only if **Expert Agent Selection (EAS)** is enabled on the Feature-Related System-Parameters screen.

.

Valid entries	Usage
y	Enter **y** to enable Automatic Number Identification (ANI). When the value is **y**, Communication Manager sends the calling party's number to the public or IBERCOM network so that charges are broken down by line.
n	If this value is **n**, charges are not itemized by line, and your company receives a single bill for the total number of calls made (block charging).

Time of Day Chart

Appears only if **Time of Day** field is enabled on the System Parameters Customer-Options **(Optional Features)** screen.

Valid entries	Usage
1 to **8**	Enter the AAR/ARS time-of-day-chart number associated with this COR.

Unrestricted Call List

Any entry on the Toll Analysis screen with an **X** in the **Toll List** column is restricted, meaning that the system blocks any attempt to complete a call containing the Dialed String. However, this field overrides that restriction.

For example, if the Toll Analysis screen shows a **Dialed String** entry of 538 and there is an **X** in the **Toll List** column, the 538 number is restricted. To override this restriction, in the Toll Analysis screen, enter **X** in the **5** column under the **Unrestricted Call List** heading. In the Class of Restriction screen, in this field, enter **5** to complete the restriction override.

Valid entries	Usage
1 to **10** or blank	Appears when **Calling Party Restriction** is **all-toll** or **tac-toll**. This field allows a user to complete a toll call with "restricted" dialed digits. This field is associated with the **Dialed String** field on the Toll Analysis screen. An Unrestricted Call List number is denoted on that screen.

Field descriptions for page 2

Figure 40: Class of Restriction screen (page 2)

```
change cor nn                                                       Page 2 of x
                         CLASS OF RESTRICTION

                     MF Incoming Call Trace? n
               Brazil Collect Call Blocking? n
                     Block Transfer Display? n
Block Enhanced Conference/Transfer Displays? y
                     Remote Logout of Agent? n

                           Station Lock COR: 10
  Outgoing Trunk Disconnect Timer (minutes):
       Line Load Control:
 Maximum Precedence Level:       Preemptable?
 MLPP Service Domain:
      Station-Button Display of UUI IE Data?
      Service Observing by Recording Device?
                         ERASE 24xx USER DATA UPON
        Dissociate or unmerge at this phone: none
          EMU login or logoff at this phone: none
           Mask CPN/NAME for Internal Calls:
```

Block Enhanced Conference/Transfer Display

Use this field to add display messages regarding conference and transfer features on digital telephones.

Valid entries	Usage
y/n	Enter **y** to block all the enhanced conference/transfer display messages except "Transfer Completed."

Block Transfer Display

Valid entries	Usage
y/n	Enter **y** to prevent users of DCP, Hybrid, ISDN-BRI, or wireless display telephones from receiving a confirmation message when they transfer a call.

Brazil Collect Call Blocking

For Brazil only.

Valid entries	Usage
y/n	Enter **y** to permit all Brazilian trunks calls that terminate to a station to send back a double answer to the CO. This double answer tells the CO that this particular station cannot accept collect calls. The CO then tears down the call if it is a collect call. Set **Country** on the Trunk Group screen to **23** and set this field to **y**.

Erase 24xx User Data Upon: Dissociate or unmerge this telephone

Use this field to administer what local terminal data items are erased when the 24xx is dissociated or unmerged.

Valid entries	Usage
none	No local terminal data is erased. This is the default.
log	Terminal's local call Log data is erased.
customizations	Call Log, Button labels, Speed Dial List, Local Terminal Options are erased.
all	All local terminal data is erased (Call Log, Button Labels, Speed Dial List, Options, Language).

Erase 24xx User Data Upon: EMU login or logoff at this telephone

Use this field to administer what local terminal data items are erased upon Enterprise Mobility User (EMU) login or logoff.

Valid entries	Usage
none	No local terminal data is erased. This is the default.
log	Terminal's local call Log data is erased.

Valid entries	Usage
customizations	Call Log, Button labels, Speed Dial List, Local Terminal Options are erased.
all	All local terminal data is erased (Call Log, Button Labels, Speed Dial List, Options, Language).

Line Load Control

Valid entries	Usage
1 to **4**	Enter the line load control level for this COR, where **1** has no restrictions, and **4** is most restrictive.

Mask CPN/Name for Internal Calls

Valid entries	Usage
y/n[1]	Enter **y** to hide the display of calling/called party numbers and administered name on internal calls.

1. This feature does not work for SIP stations.

Maximum Precedence Level

Assign a maximum precedence level for extensions with this COR for use with the Multiple Level Precedence and Preemption feature.

Valid entries	Usage
fo	Flash Override
fl	Flash
im	Immediate
pr	Priority
ro	Routine (default)

MF Incoming Call Trace

Valid entries	Usage
y/n	Enter **y** to allow assignment of a Call Trace COR to a station. Communication Manager then generates an MFC backward signal (administered on the System-Parameters Multifrequency-Signaling screen) during call setup instead of the free signal. This triggers the central office to collect trace information before releasing the calling party, if the terminating station's COR has this feature set to **y**.

MLPP Service Domain

Valid entries	Usage
1 to **16777215**	Enter the service domain for users and trunks to which this particular COR is assigned.

Outgoing Trunk Disconnect Timer (minutes)

This feature provides the capability to disconnect an outgoing trunk automatically after an administrable amount of time. This field defaults to blank (outgoing trunk calls are only disconnected when dropped by one or all parties), or you can enter a timer value in number of minutes to apply to outgoing trunk calls if the initiating party belongs to this COR

.

Valid entries	Usage
2 to **999**	Enter a value of as many as 3 characters in number of minutes. A warning tone is given to all parties on the trunk call 1 minute before the administered value (that is, after 1 to 998 minutes have elapsed) and a second warning tone is heard 30 seconds later. The call is automatically disconnected 30 seconds after the second warning tone.

Preemptable

Valid entries	Usage
y/n	Enter **y** to make extensions with this COR preemptable for Multiple Level Precedence and Preemption calls.

Remote Logout of Agent

Use a feature access code to logout an idle ACD or EAS agent without being at the agent's telephone.

Valid entries	Usage
y/n	Enter **y** to allow remote logout of an idle ACD or EAS agent.

Service Observing by Recording Device

Valid entries	Usage
y/n	When set to **y**, the service observer associated with the COR is actually a remote service-observing connection made by an audio recording device such as the Witness product. Default is **n**.

Station-Button Display of UUI IE Data

This field can only be set to **y** if the Call Center release is 3.0 or later.

Valid entries	Usage
y/n	Enter **y** to allow a station user to push a **uui-info** station-button and see up to 32 bytes of ASAI-related User-User-Information Information Element (UUI-IE) data. Pressing the **uui-info** button displaces the incoming call/collected digits display. Pressing **callr-info** redisplays the collected digits. Default is **n**.

Station Lock COR

This field defaults to the current screen COR. Extensions that are assigned this COR can use Station Lock with the access code administered on the FAC screen

Valid entries	Usage
0 to **995**	This field defaults to current COR.

Field descriptions for page 3

Figure 41: Class of Restriction screen (page 3)

```
change cor 1                                                  Page   3 of  x
                              CLASS OF RESTRICTION

               SAC/CF Override by   Team Btn? n  Priority Call? n Dialing? n
   SAC/CF Override Protection for   Team Btn? n  Priority Call? n Dialing? n



                         one-X Server Access? y
Team Btn  Silent if Active? n  Priority Ring? n    Auto Answer? n
Team Btn  Display Name? n      Pick Up by Going Off Hook? n
```

Figure 42: Class of Restriction screen (page 3)

```
Change cor 1                                                 Page   3 of  x
                              CLASS OF RESTRICTION

            SAC/CF Override by   Team Btn? n  Priority Call? n  Dialing? n
SAC/CF Override Protection for   Team Btn? n  Priority Call? n  Dialing? n
Display Names on Bridged Appearance Labels? n
         Remove Caller Id from Set Display? n
_   ___    n
```

SAC/CF Override by Team Btn

This feature allows the user of a station with a **Team** button administered, who is monitoring another station, to directly reach the monitored station by pushing the **Team** button. This overrides any currently active rerouting (for example, Send All Calls, Call Forwarding) on the monitored station.

Valid entries	Usage
y/n	Enter **y** to allow override of active rerouting on a monitored station. Default is **n**.

SAC/CF Override Protection for Team Btn

Valid entries	Usage
y/n	Enter **y** to protect stations in this COR from SAC/CF Override rerouting. Default is **n**.

SAC/CF Override by Priority Call and Dialing

This feature allows the user of a station to enable the SAC/CF override feature depending on call initiation, by pushing the **Priority** button or by using the dial pad (dialing).

Valid entries	Usage
y/n	Enter **y** to allow override of active rerouting on a called station. Default is **n**.

SAC/CF Override Protection for Priority Call and Dialing

This feature allows the user of a station to enable the SAC/CF override protection feature depending on call initiation, by pushing the **Priority** button or by using the dial pad (dialing).

Valid entries	Usage
y/n	Enter **y** to allow override of active rerouting on a called station. Default is **n**.

For more information on Overriding of SAC/CF, see *Avaya Aura™ Communication Manager Feature Description and Implementation,* 555-245-205.

Team Btn Silent if Active?

Valid entries	Usage
y/n	Set the value to **y** to enable audible ringing on team button calls. Default is **y**.

Priority Ring?

Valid entries	Usage
y/n	Set the value to **y** to enable priority ringing for speed dialing on team button calls. Default is **n**.

Auto Answer?

Valid entries	Usage
y/n	Set the value to **y** to enable automatic answer on team button calls. Default is **n**.

Team Btn Display Name?

Valid entries	Usage
y/n	Set the value to **y** to enable display of station name on team button calls. Default is **n**.

Team Pick Up by Going Off Hook?

Valid entries	Usage
y/n	Set the value to **y** to enable pick up by going off hook on team button calls. Default is **n**.

Field descriptions for page 4 to 23

You can use the screens from page 4 to page 13 to assign calling permissions and the screens from page 14 to page 23 to assign service observing permissions to the COR that you administer.

Figure 43: Class of Restriction screen (page 4)

```
change cor nn                                               Page 4 of x
                      CLASS OF RESTRICTION
CALLING PERMISSION (Enter y to grant permission to call specified COR)
  0? n     15? n     30? n     44? n     58? n     72? n    86? n
  1? n     16? n     31? n     45? n     59? n     73? n    87? n
  2? n     17? n     32? n     46? n     60? n     74? n    88? n
  3? n     18? n     33? n     47? n     61? n     75? n    89? n
  4? n     19? n     34? n     48? n     62? n     76? n    90? n
  5? n     20? n     35? n     49? n     63? n     77? n    91? n
  6? n     21? n     36? n     50? n     64? n     78? n    92? n
  7? n     22? n     37? n     51? n     65? n     79? n    93? n
  8? n     23? n     38? n     52? n     66? n     80? n    94? n
  9? n     24? n     39? n     53? n     67? n     81? n    95? n
 10? n     25? n     40? n     54? n     68? n     82? n    96? n
 11? n     26? n     41? n     55? n     69? n     83? n    97? n
 12? n     27? n     42? n     56? n     70? n     84? n    98? n
 13? n     28? n     43? n     57? n     71? n     85? n    99? n
 14? n     29? n
```

CALLING PERMISSION

Users with this COR can call users with the CORs from 0 to 995.

Valid entries	Usage
y/n	If you set the field to **y**, a station user with the COR that you administer can call stations with the other COR. If you set the field to **n**, a station user with the COR that you administer cannot call stations with the other COR.The default value is **y**.

SERVICE OBSERVING PERMISSION

Users with this COR can observe users with the CORs from 0 to 995.

Valid entries	Usage
y/n	If you set the field to **y**, a station user with the COR that you administer can observe stations with the other COR. If you set the field to **n**, a station user with the COR that you administer cannot observe stations with the other COR.The default value is **y**.

Class of Service

This screen administers access permissions for call processing features that require dial code or feature button access.

Note:
Class of Service (COS) does not apply to trunk groups except for the Remote Access feature.

A COS assignment defines whether or not a telephone user can access or use the following features and functions. Up to 16 different COS numbers can be administered (0 to 15). When the **Tenant Partitioning** field is **y** on the System Parameters Customer-Options (Optional Features) screen, you can administer up to 100 COS groups, each with 16 Classes of Service. This can be useful in controlling service to the stations and attendant of different tenants.

Field descriptions for page 1

Figure 44: Class of Service screen

```
change cos-group 1                                                  Page 1 of x
CLASS OF SERVICE       COS Group: 1     COS Name: COS Group 1


                                    0 1 2 3 4 5 6 7 8 9 10 11 12 13 14 15

    Auto Callback                   n y y n y n y n y n  y  n  y  n  y  n
    Call Fwd-All Calls              n y n y y n n y y n  n  y  y  n  n  y
    Data Privacy                    n y n n n y y y y n  n  n  n  y  y  y
    Priority Calling                n y n n n n n n n y  y  y  y  y  y  y
    Console Permissions             n n n n n n n n n n  n  n  n  n  n  n
    Off-hook Alert                  n n n n n n n n n n  n  n  n  n  n  n
    Client Room                     n n n n n n n n n n  n  n  n  n  n  n
    Restrict Call Fwd-Off Net       n y y y y y y y y y  y  y  y  y  y  y
    Call Forward Busy/DA            n n n n n n n n n n  n  n  n  n  n  n
    Personal Station Access (PSA)   n n n n n n n n n n  n  n  n  n  n  n
    Extended Forwarding All         n n n n n n n n n n  n  n  n  n  n  n
    Extended Forwarding B/DA        n n n n n n n n n n  n  n  n  n  n  n
    Trk-to-Trk Restriction Override n n n n n n n n n n  n  n  n  n  n  n
    QSIG Call Offer Originations    n n n n n n n n n n  n  n  n  n  n  n
    Contact Closure Activation      n n n n n n n n n n  n  n  n  n  n  n
    Automatic Exclusion             n n n n n n n n n n  n  n  n  n  n  n
```

The screen lists the default values for each COS/feature combination. For a particular combination, **y** allows access to the feature and **n** denies access. Assign entries on the screen for each COS to be implemented. Default values are shown on the screen.

CAUTION:
Because many hunt groups are set up with COS 1, be careful when you assign restrictions to COS 1.

Automatic Callback

Allows this user to request Automatic Callback. See Automatic Callback in *Avaya Aura™ Communication Manager Feature Description and Implementation,* 555-245-205, for more information.

Automatic Exclusion

Allows a user to activate automatically Exclusion when they go off hook on a station that has an assigned **Exclusion** button. If set to **n**, allows a user manual exclusion when they press the **Exclusion** button before dialing or during a call. Appears when, on the Feature-Related System Parameters screen, the **Automatic Exclusion by COS** field is **y**.

Call Forwarding All Calls

Allows this user to forward all calls to any extension. See Call Forwarding in *Avaya Aura™ Communication Manager Feature Description and Implementation,* 555-245-205, for more information.

Call Forwarding Busy/DA

Allows this user to forward calls to any extension when the dialed extension is busy or does not answer. See Call Forwarding in *Avaya Aura™ Communication Manager Feature Description and Implementation,* 555-245-205, for more information.

Client Room

Allows users to access Check-In, Check-Out, Room Change/Swap, and Maid status functions. In addition, Client Room is required at consoles or telephones that are to receive message-waiting notification. You can administer class of service for Client Room only when you have Hospitality Services and a Property Management System interface.

Console Permissions

Console Permissions allow multiappearance telephone users to control the same features that the attendant controls. You might assign this permission to front-desk personnel in a hotel or motel, or to a call center supervisor. With console permission, a user can:

- Activate Automatic Wakeup for another extension
- Activate and deactivate controlled restrictions for another extension or group of extensions
- Activate and deactivate Do Not Disturb for another extension or group of extensions
- Activate Call Forwarding for another extension
- Add and remove agent skills
- Record integrated announcements

Contact Closure Activation

Allows a user to open and close a contact closure relay.

COS Group

This field appears when, on the System Parameters Customer-Options (Optional Features) screen, the **Tenant Partitioning** field is **y**. The Class of Service group corresponding to the value given in the command line (`cos-group number` [between 1 to 100]. You can administer up to 100 COS groups.

COS Name

This field appears when, on the System Parameters Customer-Options (Optional Features) screen, the **Tenant Partitioning** field is **y**. The identifying name for this COS group.

Extended Forwarding All

Allows a user to administer call forwarding (for all calls) from a remote location. You cannot change a COS to **y** if **Extended Cvg/Fwd Admin** on the System Parameters Customer-Options **(Optional Features)** screen is **n**.

Extended Forwarding B/DA

Allows this user to administer call forwarding (when the dialed extension is busy or does not answer) from a remote location. You cannot change this COS to **y** if **Extended Cvg/Fwd Admin** on the System Parameters Customer-Options **(Optional Features)** screen is **n**.

Off-Hook Alert

See Emergency Access to the Attendant in *Avaya Aura™ Communication Manager Feature Description and Implementation,* 555-245-205, for more information. To enable this option, either the **Hospitality (Basic)** or **Emergency Access to Attendant** field must be enabled in your license file. When enabled, these fields display as **y** on the System Parameters Customer-Options (Optional Features) screen.

Personal Station Access (PSA)

Allows users to associate a telephone to their extension with their programmed services, using a feature access code. This field must be set to **n** for virtual telephones. This field must be set to **y** at a user's home station in order for that user to use the Enterprise Mobility User (EMU) feature at other stations. You cannot change this field to **y** if **Personal Station Access (PSA)** on the System Parameters Customer-Options **(Optional Features)** screen is **n**. See Personal Station Access in *Avaya Aura™ Communication Manager Feature Description and Implementation,* 555-245-205, for more information. For more information about Enterprise Mobility User, see *Administering Avaya Aura™ Communication Manager*, 03-300509.

Priority Calling

Allows user to dial a feature access code to originate a priority call. Such calls ring differently and override send all calls, if active. See Priority Calling in *Avaya Aura™ Communication Manager Feature Description and Implementation,* 555-245-205, for more information.

QSIG Call Offer Originations

Allows this user to invoke QSIG Call Offer services.

Restrict Call Fwd-Off Net

This restricts users from forwarding calls to the public network. For security reasons, this should be enabled for all classes of service except the ones you use for very special circumstances. See Call Forwarding Off-net in *Avaya Aura™ Communication Manager Feature Description and Implementation,* 555-245-205, for more information.

Trk-to-Trk Restriction Override

Users with this COS override any system and/or COR-to-COR calling party restrictions that would otherwise prohibit the trunk-to-trunk transfer operation for users with this COS. See Trunk-to-Trunk Transfer in *Avaya Aura™ Communication Manager Feature Description and Implementation,* 555-245-205, for more information.

SECURITY ALERT:
Use this COS capability with caution. The ability to perform trunk-to-trunk transfers greatly increases the risk of toll fraud.

Field descriptions for page 2

Figure 45: Class of Service screen

```
change cos                                                            Page 2 of x
                            CLASS OF SERVICE

                                    0 1 2 3 4 5 6 7 8 9 10 11 12 13 14 15

VIP Caller                          y y y y y y y y y y  y  y  y  y  y  y

Masking CPN/Name Override           y y y y y y y y y y  y  y  y  y  y  y
Call Forwarding Enhanced            y y y y y y y y y y  y  y  y  y  y  y
Priority Ip Video                   y y y y y y y y y y  y  y  y  y  y  y
Ad hoc Video Conferencing           y y y y y y y y y y  y  y  y  y  y  y
```

Ad hoc Video Conferencing

Enables Ad-hoc Video Conferencing, so that up to six users can participate in a video conference call.

Call Forwarding Enhanced

Allows users to designate different preferred destinations for forwarding calls that originate from internal and external callers.

Masking CPN/Name Override

Allows users to override the MCSNIC capability (that is, masking the display of calling party information and replacing it with a “hard-coded,” system-wide text string, "Info Restricted").

Note:
This feature has no effect when activated on SIP stations because MCSNIC capability is not supported on SIP stations.

Priority Ip Video

Allows priority video calling, where video calls have an increased likelihood of receiving bandwidth and can also be allocated a larger maximum bandwidth per call.

VIP Caller

Enables automatic priority calling when assigned to the originator of a call. A call from a VIP telephone is always a priority call without the use of a feature button or FAC. Default is **n**. For more information on the VIP Caller feature, See Priority Calling in *Avaya Aura™ Communication Manager Feature Description and Implementation,* 555-245-205.

Code Calling IDs

On systems with chime paging, use this screen to assign a unique series of chimes (a *chime code*) to extensions. The chime code assigned to an extension plays over the speakers whenever that extension is paged. You can assign chime codes to up to 125 extensions. Page 1 of this screen provides for the entry of ID Assignments 111-245. Page 2, IDs 251-435, and Page 3, IDs 451-555.

Field descriptions for page 1

Figure 46: Code Calling IDs screen

```
change paging code-calling-ids                                  Page 1 of 3

                                     CODE CALLING IDs
     ID ASSIGNMENTS
              Id   Ext           Id   Ext           Id   Ext
              111:               141:               221:
              112:               142:               222:
              113:               143:               223:
              114:               144:               224:
              115:               145:               225:
              121:               151:               231:
              122:               152:               232:
              123:               153:               233:
              124:               154:               234:
              125:               155:               235:
              131:               211:               241:
              132:               212:               242:
              133:               213:               243:
              134:               214:               244:
              135:               215:               245:
```

Ext

This field assigns extensions to chime codes. Only one extension can be assigned to each chime code.

Valid entries	Usage
An extension	Enter a physical extension, not a VDN, to assign that extension to a code. Otherwise, leave this field blank.

Related topics

See Loudpeaker Paging in *Avaya Aura™ Communication Manager Feature Description and Implementation,* 555-245-205, for more information for a description of the feature.

Command Permission Categories

Beginning with Communication Manager 4.0, there is no longer a Command Permission Categories screen. For details on screens used for login permissions, see *Maintenance Commands for Avaya Aura™ Communication Manager* Media Gateways and Servers, 03-300431, and AAA Services in *Avaya Aura™ Communication Manager Feature Description and Implementation*, 555-245-205.

Communication Interface Processor Channels

See **Processor Channel Assignment**.

Configuration Set

This screen defines a number of call treatment options for Extension to Cellular cellular telephone calls. The Extension to Cellular feature allows the use of up to 99 Configuration Sets, which are already defined in the system using default values.

Field descriptions for page 1

Figure 47: Configuration Set screen

```
change off-pbx-telephone configuration-set 1                    Page   1 of   x



                                     CONFIGURATION SET: 1

                         Configuration Set Description:
                                  Calling Number Style: network
                                   CDR for Origination: phone-number
                    CDR for Calls to EC500 Destination? y
                           Fast Connect on Origination? n
                          Post Connect Dialing Options: dtmf
                         Cellular Voice Mail Detection: timed  (seconds): 4
                                         Barge-in Tone? n
                           Calling Number Verification? y
             Call Appearance Selection for Origination: primary-first
                                      Confirmed Answer? n

 Use Shared Voice Connections for Second Call Answered? n
Use Shared Voice Connections for Second Call Initiated? n
```

Barge-In Tone

The barge-in tone adds security to Extension to Cellular. If a user is on an active Extension to Cellular call and another person joins the call from the Extension to Cellular enabled office telephone, all parties on the call hear the barge-in tone.

Valid fields	Usage
y/n	Default is **n**.

Call Appearance Selection for Origination

Use this field to specify how the system selects a Call Appearance for call origination.

Valid entries	Usage
first-available	If **Bridged Calls** on the Stations With Off-PBX Telephone Integration screen is **y**, the system searches for the first available regular or bridged Call Appearance.
primary-first	This is the default. ● If **Bridged Calls** on the Stations With Off-PBX Telephone Integration screen is **n**, only regular Call Appearances are used for call origination. If a regular call appearance is not available, the call is not allowed. ● If **Bridged Calls** on the Stations With Off-PBX Telephone Integration screen is **y**, the system first searches for a regular Call Appearance for call origination. If a regular Call Appearance is not available, a second search is made that includes both regular and bridged Call Appearances.

Calling Number Style

Determines the format of the caller ID for calls from a local Communication Manager extension to an Extension to Cellular telephone.

Valid entries	Usage
network	Provides a display of only 10-digit numbers. For internal calls, the ISDN numbering tables are used to create the calling number and DCS calls use the ISDN calling number if provided. The externally provided calling number is used when available for externally originated calls.
pbx	Provides a display of less than 10-digits. Extensions sent as the calling number for all internally- and DCS network-originated calls.

Calling Number Verification

Use this field to impose restrictions on calls originating from an Extension to Cellular-enabled cell phone to some other extension.

Valid entries	Usage
y	If you set the field to **y** and the network service provider has not marked the screening indicator as “network provided" or "user provided verified and passed", Communication Manager treats the incoming call as an ordinary incoming trunk call and not as a call from an extension.
n	If you set the field to **n**, Communication Manager treats the incoming call as a call from the corresponding desk phone.

CDR for Calls to EC500 Destination

Determines whether a call detail record is generated for any call to the cell telephone.

Note:
CDR reporting for Extension to Cellular calls relies on the **CDR Reports** field on the Trunk Group screen. If, on the Trunk Group screen, the **CDR Reports** field is **n**, no CDR is generated even if this field is **y**.

Valid entries	Usage
y	Treats calls to the XMOBILE station as trunk calls and generates a CDR.
n	Treats calls to the XMOBILE station as internal calls and does not generate a CDR.

CDR for Origination

You can generate CDR records for a call that originates from an Extension to Cellular cell phone. To generate this CDR, you must enable the Incoming Trunk CDR. The CDR report does not include dialed Feature Name Extensions (FNEs). The entries for this field determine the CDR report format.

Valid entries	Usage
phone-number	The calling party on the CDR report is the 10-digit cell phone number. This is the default.
extension	The calling party on the CDR report is the internal office telephone extension associated with the Extension to Cellular cell phone
none	The system does not generate an originating CDR report.

Cellular Voice Mail Detection

This field prevents cellular voice mail from answering an Extension to Cellular call. When the call server detects that the cell phone is not the entity answering the call, the call server brings the call back to the server.

Valid entries	Usage
none	Default entry is **none**
timed	When you enter **timed**, the system displays the **(seconds)** field, which accepts values from **1** to **9** seconds. The default value is **4** seconds. In the Extension to Cellular enabled environment, if a user answers the call at the cell within the configured time, Communication Manager treats the call as being answered by the cellular voice mail, and disconnects the cellular leg of the call. The call continues to ring at the desk phone. You can use this configuration for any type of network, including GSM, CDMA, and ISDN.
message	Detect carrier voice mail

Configuration Set Description

Describes the purpose of the configuration set.

Valid entries	Usage
Up to 20 alphanumeric characters or blank	For example, Extension to Cellular handsets.

Confirmed Answer

Use this field to require the user to input a digit to confirm receipt of a call sent to a cellular telephone by the Extension to Cellular feature. Upon answering the incoming call on the cellular telephone, the user hears a dial tone.The user must then press any one of the digits on the telephone keypad. Until the system receives a digit, the system does not treat the call as answered. The length of time to wait for the digit can be administered from 5-20 seconds, with a default of 10 seconds. The system plays a recall dial-tone to indicate that input is expected. During the response interval, the original call continues to alert at the desk set and any stations bridged to the call. If the user does not enter a digit before the timeout interval expires, the call is pulled back from the cell phone.

Valid entries	Usage
y/n	Enter **y** to enable Confirmed Answer on Extension to Cellular calls for this station. Default is **n**.

Fast Connect on Origination

Determines whether some additional processing occurs on the server running Communication Manager prior to connecting a call.

Valid entries	Usage
y/n	Enter **y** to send CONNECT messages.

Post Connect Dialing Options

Determines whether additional capabilities, beyond standard ISDN dialing, are available for those incoming ISDN trunk calls that are mapped into XMOBILE stations. These options come into effect after the call has entered the active state (Communication Manager has sent a CONNECT message back to the network).

Valid entries	Usage
dtmf	Expect digits from either in-band or out-of-band, but not simultaneously. The server allocates a DTMF receiver whenever it needs to collect digits. This option normally would be used for Extension to Cellular XMOBILE station calls.
out-of-band	Expect all digits to be delivered by out-of-band signaling only. The server running Communication Manager collects digits that it needs from the out-of-band channel (no touch-tone receiver). In addition, any digits received when the server is not collecting digits are converted to DTMF and broadcast to all parties on the call. This option is in force for DECT XMOBILE station calls.
both	Expect all subsequent digits to be delivered by simultaneous in-band and out-of-band signaling. Out-of-band signaling consists of digits embedded in ISDN INFO messages while the in-band signaling consists of DTMF in the voice path. The server running Communication Manager collects all digits that it needs from the out-of-band channel. No touch tone receive is allocated in order to prevent collecting double digits. End-to-end signaling occurs transparently to the server via in-band transmission of DTMF. This option is in force for PHS XMOBILE station calls.

Console Parameters

This screen administers attendant console group parameters. This includes basic parameters for Centralized Attendant Service (CAS) and Inter-PBX Attendant Service (IAS). A list of the administered attendant consoles also displays on this screen.

Field descriptions for page 1

Figure 48: Console Parameters — page 1

```
change console-parameters                                        Page   1 of x
                              CONSOLE PARAMETERS
         Attendant Group Name: OPERATOR
                         COS: 0                                  COR: 0
       Calls in Queue Warning: 5                    Attendant Lockout? y
        Ext Alert Port (TAAS):
                         CAS: none
                                            Night Service Act. Ext.:
                 IAS (Branch)? n            IAS Tie Trunk Group No.:
         IAS Att. Access Code:                Alternate FRL Station:
              Backup Alerting? n       DID-LDN Only to LDN Night Ext? n
      Attendant Vectoring VDN:
```

AAR/ARS Access Code

Appears if the **CAS** field is **QSIG-branch**. An optional field that contains an AAR/ARS access code to route to the main PBX, if needed.

Valid entries	Usage
0 to **9**, *****, **#** blank	Enter up to 4 digits.

Alternate FRL Station

This is a display-only field indicating the extension of the alternate facility restriction level (FRL) activation station.

Attendant Group Name

Valid entries	Usage
1 to **27** alphanumeric characters	Enter a name for the attendant group.

Attendant Lockout

Attendant Lockout prevents an attendant from re-entering a multiple-party connection held on the console unless recalled by a telephone user.

Attendant Lockout provides privacy for parties on a multiple-party call held on the console. The held parties can hold a private conversation without interruption by the attendant.

Valid entries	Usage
y/n	Enter **y** to activate Privacy — Attendant Lockout. If **y** is entered, the attendant is prohibited from reentering a conference call that has been placed on hold unless recalled by a telephone user on the call.

Attendant Vectoring VDN

This field appears only if, on the System Parameters Customer-Options (Optional Features) screen, the **Attendant Vectoring** field is **y** and the **Tenant Partitioning** field is **n**.

Valid entries	Usage
Assigned VDN extension or blank	Enter an assigned Attendant VDN extension or blank.

Backup Alerting

Indicates whether or not system users can pick up alerting calls if the attendant queue has reached its warning state.

Calls In Queue Warning

Enter the number of incoming calls that can be in the attendant queue before the console's second Call Waiting lamp lights. The console's first Call Waiting lamp lights when any incoming calls are waiting to be answered. The second lamp lights when the number of calls waiting equals the value you entered in the **Calls in Queue Warning** field.

Valid entries	Usage
1 to attendant queue maximum	Enter the number of incoming calls that can be in the attendant queue before the console's second Call Waiting lamp lights. For queue maximum, see *Avaya Aura™ Communication Manager System Capacities Table*, 03-300511.

CAS

Valid entries	Usage
main	This is the main Communication Manager sever on which the attendant group is located. Uses non-ISDN signaling. You must enable the **CAS Main** field on the System Parameters Customer-Options (Optional Features) screen to select this option.
branch	This is a branch Communication Manager server: there are no local attendants, so attendant-seeking calls route to the main Communication Manager server. Uses non-ISDN signaling. You must enable the **CAS Branch** field on the System Parameters Customer-Options (Optional Features) screen to select this option.
none	Centralized Attendant Service is disabled.
QSIG-main	Same as main, but with QSIG signaling among the Communication Manager servers. You must set the **Centralized Attendant** field to **y** on the QSIG Optional Features screen to select this option.
QSIG-branch	Same as branch, but with QSIG signaling among the Communication Manager servers. You must set the **Centralized Attendant** field to **y** on the QSIG Optional Features screen to select this option.

CAS Back-Up Ext.

This field handles attendant-seeking calls if the RLT trunk group to the CAS Main server is out of service or if CAS Back-Up is activated. This field must be explicitly defined as an extension in the dial plan. Neither a prefixed extension nor a VDN extension is allowed. Appears only when **branch** is entered in the **CAS** field.

Valid entries	Usage
An extension number for a station	Enter an extension in the dial plan to use for CAS backup.
individual attendant console	
hunt group	
TEG	

COR

For more information about Class of Restriction (COR), see *Avaya Aura™ Communication Manager Feature Description and Implementation*, 555-245-205.

Valid entries	Usage
0 to **995**	Enter the class of restriction (COR) number that reflects the desired features for the attendant. You can override this COR, by assigning a different COR on the individual Attendant Console screen.

COS

For more information about Class of Service (COS), see *Avaya Aura™ Communication Manager Feature Description and Implementation*, 555-245-205.

Valid entries	Usage
1 to **15**	Enter a class of service (COS) number that reflects the desired features for all your attendant consoles. You can override this COS, by assigning a different COS on the individual Attendant Console screen.

DID-LDN Only to LDN Night Ext.

Valid entries	Usage
y	Enter **y** to allow only listed directory number (LDN) calls to go to the listed directory night service extension.
n	Enter **n** if you want all attendant seeking calls to route to the LDN night service extension.

Ext Alert Port (TAAS)

Enter the port number assigned to the external alerting device. This supports the Night Service — Trunk Answer From Any Station feature.

Note:
Type an **x** in this field to indicate that there is no hardware associated with this port assignment. If an **x** is used here, you must also fill in the **Ext Alert (TAAS) Extension** field.

Ext Alert (TAAS) Extension

Appears only when an **x** is entered in the **Ext Alert Port (TAAS)** field. This extension is used by the Terminal Translation Feature (TTI) to assign a port to the Ext Alert Port from a station on the Ext Alert port during system installation or provisioning. Once a port is assigned (either via TTI or by changing the **Ext Alert Port** field from the G3-MA or other manager terminal) the extension is automatically removed and treated as unassigned.

IAS Att. Access Code

Enter the extension number of the attendant group at the main server running Communication Manager. This entry is required when IAS Branch is **y**. Does not appear if, on the **System Parameters Customer-Options (Optional Features)** screen, the **Centralized Attendant** field is **y**.

IAS (Branch)

Enable or disable Inter-PBX Attendant Service (IAS) Branch feature. Does not appear if, on the System Parameters Customer-Options (Optional Features) screen, the **Centralized Attendant** field is **y**.

Note:
CAS and IAS cannot both be active at the same time.

IAS Tie Trunk Group No.

Note:
Enter the number of the tie trunk group to the main for the IAS (Branch). This entry is required when IAS Branch is **y**. Does not appear if, on the System Parameters Customer-Options (Optional Features) screen, the **Centralized Attendant** field is **y**.

Valid entries	Usage
1 to 666 **1** to **2000**	For DEFINITY CSI. For S8300/S87XX Servers

Night Service Act. Ext.

This is a display-only field containing the extension of the current night service activation station, if any. Such a station is administered by assigning it a **night-serv** button.

QSIG CAS Number

Appears if the **CAS** field is **QSIG-branch**. Contains the complete number of the attendant group at the main server running Communication Manager, or a vector directory number (VDN) local to the branch server. This field cannot be left blank

Valid entries	Usage
0 to **9**	Enter up to 20 digits.

RLT Trunk Group No.

Appears only when **branch** is entered in the **CAS** field. Enter the trunk group number corresponding to the Release Link Trunk (RLT) trunk group to the main location when supporting CAS Branch service.

Field descriptions for page 2

Figure 49: Console Parameters — page 2

```
change console-parameters                                        Page   2 of x
                              CONSOLE PARAMETERS

TIMING
  Time Reminder on Hold (sec): 10              Return Call Timeout (sec): 10
  Time in Queue Warning (sec):       Overflow timer to Group Queue (sec): 1024

  INCOMING CALL REMINDERS
      No Answer Timeout (sec): 20                      Alerting (sec): 40
                               Secondary Alert on Held Reminder Calls? y

ABBREVIATED DIALING
     List1: group    1        List2:                   List3:
            SAC Notification? n
                           COMMON SHARED EXTENSIONS
            Starting Extension:                  Count:
            Busy Indicator for Call Parked on Analog Station Without Hardware?
```

TIMING

Return Call Timeout (sec)

Valid entries	Usage
10 to **1024** or blank	Enter the time in seconds before a split away call (call extended and ringing a station or otherwise split away from the console) returns to the console. Be sure to allow five seconds for each ring at all points in a coverage path to ensure the entire path is completed before the call returns to the console.

Time In Queue Warning (sec)

Valid entries	Usage
9 to **999** or blank	Enter the number of seconds a call can remain in the attendant queue before activating an alert.

Time Reminder on Hold (sec)

Valid entries	Usage
10 to **1024**	Enter the number of seconds a call can remain on Hold.

Overflow timer to Group Queue (sec)

Valid entries	Usage
10 to **1024** Default: **Blank**	Enter the number of seconds a returning call will queue to the individual attendant before overflowing to the group. Allowed values are between 10 and 1024, and blank. The value applies if the attendant who previously handled the call is busy or unavailable. Blank indicates the call immediately goes to the group.

INCOMING CALL REMINDERS

Alerting (sec)

Enter the number of seconds after which a held or unanswered call is disconnected from an attendant loop and routed to another attendant or night service

No Answer Timeout (sec)

Enter the number of seconds a call to the attendant can remain unanswered without invoking a more insistent sounding tone. Be sure to allow five seconds for each ring at all points in a coverage path to ensure the entire path is completed before the call returns to the console.

Valid entries	Usage
10 to **1024** or blank	

Secondary Alert on Held Reminder Calls?

Valid entries	Usage
y	Enter **y** to begin attendant alerting for Held Reminder Calls with secondary alerting.
n	Enter **n** to have held reminder calls alert the attendant the same as normal calls. Normal calls start with primary alerting and then switch to secondary alerting when the No Answer Timeout expires.

ABBREVIATED DIALING

List1, List2, List3

You can assign up to 3 abbreviated dialing lists to each attendant. However, you cannot assign a personal list to an attendant

.

Valid entries	Usage
enhanced	Allows the attendant to access the enhanced system abbreviated dialing list.
group	Allows the attendant to access the specified group abbreviated dialing list. You also must enter a group number.
system	Allows the attendant to access the system abbreviated dialing list.

SAC Notification

Valid entries	Usage
y/n	Enables or disables Enhanced Attendant Notification for Send All Calls.

COMMON SHARED EXTENSIONS

Busy Indicator for Call Parked on Analog Station Without Hardware?

Valid entries	Usage
y/n	Enter **y** to indicate that the Busy Indicator lamp lights for incoming calls parked on AWOH stations. Default is **n**.

Count

Enter a number to indicate the number of consecutive extensions, beginning with the Start Extension to be used as common, shared extensions. For example, if you enter a starting extension of 4300 and a count of 3, the system provides three consecutive extension numbers (4300, 4301, and 4302) for parking calls.

The extensions should be assigned to the optional Attendant Selector Console in the 00 through 09 block (bottom row) in any hundreds group for easy identification by the attendant. The lamp associated with the number identifies “call parked” or “no call parked,” instead of busy or idle status.

Valid entries	Usage
1 to **1182** or blank	Enter a number to indicate the number of consecutive extensions, beginning with the Start Extension to be used as common, shared extensions.

Starting Extension

These extension numbers can be used by the attendant to park calls.

Field descriptions for page 3

Figure 50: Console Parameters — page 3

```
change console-parameters                                          Page 3 of x
                                CONSOLE PARAMETERS
QUEUE PRIORITIES
            Emergency Access:1_
             Assistance Call:2_
                     CO Call:2_
            DID to Attendant:2_
                    Tie Call:2_
         Redirected DID Call:2_
             Redirected Call:2_
                 Return Call:2_
                 Serial Call:2_
 Individual Attendant Access:2_
             Interpositional:2_
    VIP Wakeup Reminder Call:2_
          Miscellaneous Call:2_

 Call-Type Ordering Within Priority Levels? n
```

Call-Type Ordering Within Priority Levels?

If you use call-type ordering, calls to the attendant are first grouped by the queue priority level, then by call type, and, finally, in the order received.

Valid entries	Usage
y	Enter **y** if you want to present calls by call type. You can assign a **type-disp** button on the **Attendant Console** screen so that the attendant can review the call type for the active call.
n	Enter **n** if you wish the calls to be queued in chronological order by queue priority level.

QUEUE PRIORITIES

Attendant Priority Queue allows attendants to answer calls by call category (for example, by trunk type). The Attendant Priority Queue handles incoming calls to an attendant when the call cannot be immediately terminated to an attendant. The calling party hears ringback until an attendant answers the call.

You can assign the same priority level to more than one call. Priority 1 is the highest priority and is the default for Emergency Access. Assign a priority level from **1** through **13** to each of the call types.

The attendant call categories are:

- Emergency Access — A call from a telephone user who dials the emergency access code (default is highest-priority level)
- Assistance Call— A call from a telephone user who dials the attendant-group access code, or from a telephone that has the Manual Originating Line Service feature activated
- CO Call — An incoming trunk call (CO/FX/WATS trunk) to an attendant group. This does not include trunk calls that return to the attendant group after a timeout or deferred attendant recall.
- DID to Attendant — An incoming DID trunk call to an attendant group. This does not include trunk calls that return to the attendant group after a timeout or deferred attendant recall.
- Tie Call — An incoming TIE trunk call (dial-repeating or direct types) to an attendant group. This does not include trunk calls that return to the attendant group after a timeout or deferred attendant recall.
- Redirected DID Call — A DID or ACD call that times out due to ring/no-answer, busy condition (if applicable), or Number Unobtainable and reroutes to the attendant group.
- Redirected Call — A call assigned to one attendant, but redirected to the attendant group because the attendant is now busy

- Return Call — A call returned to the attendant after it times out. If the attendant is now busy, the call redirects to the attendant group.
- Serial Call — A call from the Attendant Serial Call feature when an outside trunk call (designated as a serial call by an attendant) is extended to and completed at a telephone, and then the telephone user goes on-hook. If the attendant who extended the call is busy, the call redirects to the attendant group.
- Individual Attendant Access — A call from a telephone user, incoming trunk call, or a system feature to the Individual Attendant Access (IAA) extension of a specific attendant. If the attendant is busy, the call queues until the attendant is available.
- Interposition — A call from one attendant to the Individual Attendant Access (IAA) extension of another attendant
- VIP Wakeup Reminder Call — A VIP Wakeup reminder call.
- Miscellaneous Call — All other calls.

The call types, in descending order of priority, are:

- Type 1 call: outgoing public-network calls receive answer supervision when the Answer Supervision Timer of the trunk group expires, even if the trunk is actually still ringing. Also, incoming calls when answered by the attendant.
- Type 2 call: incoming external public-network calls before they receive answer supervision or before the Answer Supervision Timer of the trunk group expires
- Type 3 call: all other calls (internal calls, conference calls, and tie-trunk calls of any type)

Note that external public-network calls have priority over all other calls including conference calls. And, answered public-network calls have priority over those calls not yet answered.

Field descriptions for page 4

Figure 51: Console Parameters — Queue Priorities screen

```
change console-parameters                                      Page 4 of x
                              CONSOLE PARAMETERS

QUEUE PRIORITIES

      MLPP PRECEDENCE CALL
       Flash Override: 2
                Flash: 3
            Immediate: 4
             Priority: 5
```

QUEUE PRIORITIES

Flash Override

Valid entries	Usage
1 to **17**	Enter the queue priority for Flash Override precedence level calls.

Flash

Valid entries	Usage
1 to **17**	Enter the queue priority for Flash precedence level calls.

Immediate

Valid entries	Usage
1 to **17**	Enter the queue priority for Immediate precedence level calls.

Priority

Valid entries	Usage
1 to **17**	Enter the queue priority for Priority precedence level calls.

Field descriptions for page 5

Note:
If MLPP is not enabled, the MLPP Queues page does not appear, and the following page appears as page 4.

Figure 52: Console Parameters — page 5

```
change console-parameters                                       Page 5 of x
                              CONSOLE PARAMETERS

     Use the 'list attendant' command to see all administered attendants.
```

Coverage Answer Group

This screen establishes Call Coverage Answer Groups.

An answer group contains up to eight members who act as a coverage point for another user. For example, if several secretaries are responsible for answering a department's redirected calls, all the secretaries could be assigned to an answer group. The answer group is assigned a group number, and that group number appears in the department's coverage path. All telephones in an answer group ring (alert) simultaneously. Any member of the group can answer the call.

Each coverage answer group is identified by a number from 1 through the maximum number allowed by your system configuration (see *Avaya Aura™ Communication Manager Hardware Description and Reference*, 555-245-207). The members of the group are identified by their extension number. Any telephone, including those administered without hardware (X-ported (but not attendants) can be assigned to a coverage answer group. Note that members whose extensions are X-ported cannot be alerted).

Field descriptions for page 1

Figure 53: Coverage Answer Group screen

```
change coverage answer-group n                                   Page 1 of x
                        COVERAGE ANSWER GROUP

                      Group Number: 3___
                      Group Name: COVERAGE_GROUP_

GROUP MEMBER ASSIGNMENTS
   Ext      Name (first 26 characters)      Ext      Name (first 26 characters)
1: ________ ___________________________ 5: ________ __________________________
2: ________ ___________________________ 6: ________ __________________________
3: ________ ___________________________ 7: ________ __________________________
4: ________ ___________________________ 8: ________ __________________________
```

Ext

Valid entries	Usage
An assigned extension for a station.	Enter the extension number (cannot be a Vector Directory Number extension) for each member of this coverage answer group.

Group Name

Enter the group name you want to use to identify this group.

Tip:
Enter the extension numbers that are group members. This allows a **`list coverage answer group`** command to be used to list the telephones that are alerted. The **`list`** command can be used in conjunction with the **`list station`**, **`list coverage path`**, and **`list hunt group`** commands to determine stations involved in call coverage. This makes it possible to follow call coverage for any extension, allowing the administrator to easily track call coverage paths.

Valid entries	Usage
Up to 27 characters	For example, typing pool, room 12, secy, and so on.

Group Number

A display-only field when the screen is accessed using an administration command such as **add** or **change**.

Name

This display-only field indicates the name assigned when the member's telephone is administered.

Note:
Avaya BRI stations support ASCII characters only. They do not support non-ASCII characters, such as Eurofont or Kanafont. Therefore, if you use non-ASCII characters in any Communication Manager **Name** field, such characters do not display correctly on a BRI station.

Coverage Path

This screen implements Call Coverage Paths. The screen provides the means to specify the call coverage criteria, the points in the coverage path used to redirect calls, and the number of times a principal's telephone rings before the call redirects to coverage.

Field descriptions for page 1

Figure 54: Coverage Path screen

```
change coverage path n                                         Page   1 of   x
                                COVERAGE PATH

            Coverage Path Number: n
                                            Hunt After Coverage: n
                Next Path Number: ___       Linkage: ___  ___

COVERAGE CRITERIA

  Station/Group Status   Inside Call   Outside Call
            Active?           n             n
              Busy?           y             y
      Don't Answer?           y             y Number of Rings:2
               All?           n             n
DND/SAC/Goto Cover?           y             y
  Holiday Coverage?           n             y  Holiday Table: 1

COVERAGE POINTS
Terminate to Coverage Pts. with Bridged Appearance? n

     Point1: _________ Rng: Point2: _________Rng:
     Point3: _________ Rng: Point4: _________Rng:
     Point5: _________ Rng: Point6: _________Rng:
```

Coverage Path Number

A display-only field indicating the coverage path being administered.

Holiday Coverage

This field determines when to redirect call to coverage for an inside or outside call.

Valid entries	Usage
y	Type **y** to send the call to an announcement.
n	Type **n** to send the call to the next point in the coverage path.

Holiday Table

This field determines when to redirect call to coverage for an inside or outside call. Available only when **Holiday Table** is set to **y** for an inside or outside call.

Valid entries	Usage
y/n	If the **Holiday Table** field is set to **y** for either inside or outside calls, the system uses a holiday table to route the call. Type the number of the holiday table to use.

Hunt After Coverage

Valid entries	Usage
y	Coverage treatment continues by searching for an available station in a hunt chain that begins with the hunt-to-station assigned on the Station screen of the last coverage point.
n	Coverage treatment is terminated; the call is left at the last available location (principal or coverage point).

Linkage

Display-only fields that show the (up to) two additional coverage paths in the coverage path chain.

Next Path Number

Enter the next coverage path in a coverage path chain. If the coverage criteria of the current coverage path is not satisfied, the system steps down this chain until it finds a coverage path with redirection criteria that matches the call status. If the chain is exhausted before the system finds a match, the call does not redirect to coverage. No path number here indicates that this path is the only path for the principal. See Call Coverage in *Avaya Aura™ Communication Manager Feature Description and Implementation,* 555-245-205, for more information.

Valid entries	Usage
1 to **9999** or blank	Enter the next coverage path in a coverage path chain.

COVERAGE CRITERIA

COVERAGE CRITERIA are the conditions that, when met, cause the call to redirect to coverage. Assign one of the following:

Valid entries	Usage
Active	Calls redirect if at least one call appearance is busy.
Busy	Calls redirect if all call appearances that accept incoming calls are busy.
Don't Answer	Calls redirect when the specified number of rings has been exceeded.
All	Calls redirect immediately to coverage and overrides any other criteria with a **y** in this column.
DND/SAC/ Goto Cover	Must be assigned before a user can activate Do Not Disturb (Hospitality Services), Send All Calls (SAC), or Go to Cover features. Allows a calling user, when calling to another internal extension, to redirect a call immediately to coverage by pressing a **Go to Cover** button. Allows a principal temporarily to direct all incoming calls to coverage, regardless of the other assigned coverage criteria by pressing the **Send All Calls** (or **Do Not Disturb**) button. **Send All Calls** also allows covering users to temporarily remove their telephones from the coverage path.

Number of Rings

Enter the number of rings.

Valid entries	Usage
1 to **99**	This is the number of rings a user's telephone rings before the system redirects the call to the first point in the coverage path.

COVERAGE POINTS

Point1, Point2, Point3, Point4, Point5, Point6

The alternate destinations that comprise a coverage path. Coverage points must be assigned sequentially beginning with Point 1 (do not leave gaps). Each path can have up to six coverage points.

Valid entries	Usage
extension	Redirects the call to an internal extension or announcement. **Note:** If entering a Multi-Location Dial Plan shortened extension, note the following: When entering a Multi-Location Dial Plan shortened extension in a field designed for announcement extensions, certain administration end validations that are normally performed on announcement extensions are not done, and resultant warnings or submittal denials do not occur. The shortened extensions also do not appear in any display or list that shows announcement extensions. Extra care should be taken to administer the correct type of announcement for the application when assigning shortened extensions.
attd	Redirects the call to the attendant or attendant group. If the system has Centralized Attendant Service (CAS), the call goes to the CAS attendant.
h1 to **h999**	Redirects the call to the corresponding hunt-group. For example, enter **h32** if you want a coverage point routed to hunt group 32. (See Hunt Group on page 404 for more information.)
c1 to **c750** **c1** to **c1000** (S8300/S87XX Servers)	Redirects the call to the corresponding coverage answer group. For example, enter **c20** if you want a coverage point routed to call coverage answer group 20. (See Coverage Answer Group on page 158 for more information.)
r1 to **r999** **r1** to **r1000** S8300/S87XX Servers	Redirects the call to the corresponding remote coverage point number. For example, enter **r27** if you want a coverage point routed to remote coverage point 27. (See Remote Call Coverage Table on page 735 for more information.)
v + extension	Redirects the call to the corresponding VDN extension. For example, enter **v12345** if you want the last administered coverage point to be the VDN associated with extension 12345. Note that a Vector Directory Number can be used only as the last administered point in a coverage path.
y + extension	Redirects the call to an internal extension, announcement, or the corresponding Vector Directory Number (VDN) extension as per the current date and time set in the **Holiday Table.**

If calls redirect to an AUDIX in a DCS network, administer a unique Hunt Group screen. Assign the AUDIX extension in the **Group Extension** field. If the AUDIX is connected to the local node, set the **Message Center** field to **audix**; if the AUDIX is connected to another node, set the **Message Center** field to **rem-audix**.

If calls redirect to Message Center (a special Uniform Call Distribution hunt group), AUDIX, or to the attendant, do not list any subsequent coverage points. These calls normally queue and never redirect to another coverage point. Calls to any hunt group queue if possible. Calls redirect from a hunt group only if all hunt group members are busy and either the queue is full or there is no queue.

If the Coverage of Calls Redirected Off-Net feature is not enabled, a remote coverage point functions as the last point in the coverage path, because the system is no longer have control of the call once it has redirected off-net. However, if the Coverage of Calls Redirected Off-Net feature is enabled, a call redirected off-net can be monitored by the system and brought back for further call coverage processing.

Rng

Valid entries	Usage
1 to **99** or blank	Enter the number of rings at this coverage point before the system redirects the call to the next point in the coverage path.

Terminate to Coverage Pts. with Bridged Appearances

Valid entries	Usage
y	Allows a call to alert as both a bridged call and a redirected call.
n	The call skips the coverage point if it has already alerted as a bridged call.

Crisis Alert System Parameters

This screen allows you to define the system parameters associated with sending crisis alert messages.

Field descriptions

Figure 55: Crisis Alert System Parameters screen

```
change system-parameters crisis-alert                        page 1 of x
                          CRISIS ALERT SYSTEM PARAMETERS

ALERT STATION
    Every User Responds? n

ALERT PAGER
            Alert Pager? y
  Originating Extension: 7768
      Crisis Alert Code: 911
                Retries: 5
   Retry Interval (sec): 30
            Main Number: 303-555-0800

                      Pager Number          Pin Number
                      1: 3035559001         1: 7614567890
                      2: 123456789012345    2: ppp1234567890pp
                      3: 123456789012345    3: ppp1234567890pp

                  DTMF Duration - Tone (msec): 100   Pause (msec): 100
```

ALERT STATION

Every User Responds

Controls who needs to respond to a crisis alert.

Valid entries	Usage
y	If set to **y**, all users who have a crisis alert button are notified and must clear the alert for every emergency alert. Assign crisis alert buttons only to attendant consoles and stations that must be notified of an emergency call.
n	If set to **n**, all users are notified, but only one user needs to acknowledge an alert. This user might be the attendant or any other digital telephone with a crisis alert button. When the alert is acknowledged by one user, the alert is cleared at all stations except the one that acknowledged the alert.

ALERT PAGER

Alert Pager

Valid entries	Usage
y/n	Enter **y** to use Crisis Alert to a Digital Pager.

Crisis Alert Code

Displays when the **Alert Pager** field is **y**. This field requires an entry before submitting the screen.

Valid entries	Usage
1 through 3 digits	The numbers in this field are the first 3 digits in the crisis alert pager message. Avaya recommends you enter the numbers used to call the local emergency service or any digits used for an emergency situation (for example, 911).

DTMF Duration - Tone (msec)

The length of time the Dual-Tone Multi-Frequency (DTMF) tone is heard for each digit. Displays when the **Alert Pager** field is **y**.

Valid entries	Usage
20 to **2550**	Enter a number in increments of 10.

Main Number

The main telephone number to the location or a location code. This field is optional and does not require an entry. Displays when the **Alert Pager** field is **y**.

Valid entries	Usage
digits **0** to **9** **-** (dash)	Enter a number up to 15 digits to identify the location where the crisis alert call originated. It can be the main number to the location or a numerical identification. Any dashes are for display purposes only and not included in the message sent to the pager. This entry is the last group of digits displayed in the pager message.

Originating Extension

Used as the extension originating the call to send a crisis alert message to a pager. Displays when the **Alert Pager** field is **y**. This field requires an entry before submitting the screen.

Valid entries	Usage
0 to **9**	Requires a valid unassigned extension according to the dial plan.

Pager Number

Displays when the **Alert Pager** field is **y**. One of these fields must have a number or the screen cannot be submitted.

Valid entries	Usage
1 to 15 digits **-** (dash)	Any dashes are for display purposes only and not included in the message sent to the pager One of the pager number fields must have a number or the screen cannot be submitted.

Pause (msec)

The length of time between DTMF tones for each digit. Displays when the **Alert Pager** field is **y**.

Valid entries	Usage
20 to **2550**	Enter a number in increments of 10.

Pin Number

This field can be used for any combination of the pager pin number and pauses or left blank. Displays when the **Alert Pager** field is **y**.

Valid entries	Usage
digits **0** to **9** **p**(ause) **#**(pound) *****(star)	Enter a number up to 15 digits. A pause (about 2 seconds) is for timing of the message. For instance, after the pin number you might want to have a pause to allow time for the pager service to set up the correct pager message box. If the pager service requires you to submit a PIN number, enter it here.

Retries

Displays when the **Alert Pager** field is **y**.

Valid entries	Usage
0 to **10**	The number of times the system tries to send out the alert message in case of an unsuccessful attempt. This increases the chances that the pager receives a crisis alert message.

Retry Interval (sec)

Displays when the **Alert Pager** field is **y**. This field is not used unless the **Retries** field is **1** to **10**.

Valid entries	Usage
30 to **60**	The administrable time period (in seconds) between retries. If an attempt to call the pager fails, the retry call attempts after the retry interval period.

CTI Link

The `cti-link` commands are available only if, on the System Parameters Customer-Options (Optional Features) screen, either the **ASAI Link Core Capabilities and/or Computer Telephony Adjunct Links** field is **y**.

Field descriptions for page 1

Figure 56: CTI Link screen when Type field is ASAI or ADJLK

```
add cti-link next                                                Page 1 of x
                                 CTI LINK
 CTI Link: 1
Extension: 40001
     Type: ASAI
     Port: 1C0501                                                   COR: 1
     Name: ASAI CTI Link 1









BRI OPTIONS
               XID? y     Fixed TEI? n
       MIM Support? n
                          CRV Length: 2
```

Figure 57: CTI Link screen when Type field is ASAI-IP or ADJ-IP

```
add cti-link next                                                Page 1 of x
                                 CTI LINK
 CTI Link: 1
Extension: 40001
     Type: ASAI-IP
                                                                    COR: 1
     Name: ASAI CTI Link 1
```

CTI Link

A display-only field indicating the CTI link number.

Valid entries	Usage
1 to system max	Communication Manager on a DEFINITY Server CSI, DEFINITY G3i, S8300 Server, S87XX Fiber-PNC Servers.

Extension

This field displays the extension for this link.

Type

For each link that you want to add to your system, you must specify the CTI link type.

Valid entries	Usage
ADJLK	Enter the CTI link type.
ADJ-IP	
ASAI	
ASAI-IP	

Port

Appears when the **Type** field is **ASAI** or **ADJLK**. Enter 7 characters to specify a port, or an x.

Valid entries	Usage
01 to **64**	First and second numbers are the cabinet number
A to **E**	Third character is the carrier
01 to **20**	Fourth and fifth characters are the slot number
01 to **32**	Sixth and seventh characters are the circuit number
x	Indicates that there is no hardware associated with the port assignment. Use for AWOH.

Name

Enter a name associated with this CTI link.

COR

Enter a Class of Restriction (COR) number to select the desired restriction.

BRI Options

XID

Appears when the **Type** field is **ASAI** or **ADJLK**. Used to identify Layer 2 XID testing capability.

MIM Support

Management Information Message Support. A display-only field that appears when the **Type** field is **ASAI** or **ADJLK**.

Fixed TEI

Appears when the **Type** field is **ASAI** or **ADJLK**. It indicates that the endpoint has a fixed Terminal Endpoint Identifier (TEI).

The TEI identifies a unique access point within a service. You must administer TEIs for fixed TEI terminals. However, for terminals with the automatic TEI capability, the system dynamically assigns the TEI.

Valid entries	Usage
y/n	Entering **y** displays the **TEI** field. For ASAI, enter **y**.

CRV Length

Appears when the **Type** field is **ASAI** or **ADJLK**. Enter **1** or **2** to indicate the length of CRV for each interface.

Field descriptions for page 2

Figure 58: CTI Link screen when Type field is ASAI-IP or ADJ-IP

```
add cti-link next                                          Page 2 of x
                                 CTI LINK
FEATURE OPTIONS

    Event Minimization?        Special Character for Restricted Number?
                         Send Disconnect Event for Bridged Apppearance?
                                       Two-Digit Aux Work Reason Codes?
                                           Block CMS Move Agent Events?
```

Block CMS Move Agent Events

Valid entries	Usage
y/n	When this option is set to **y**, if CMS sends an agent-move-while-staffed message (MVAGSFD8), ASAI does not send the associated agent Logout Event Report (C_Logout), Login Event Report (C_login) and Agent Work Mode Change event report messages to report the changes involved with the move of agents while staffed. Default is **n**.

Event Minimization

This option can be used when event reports normally would be sent on multiple associations, but the adjunct does not need to see more than one. Typically, these event reports are identical except for the association they are sent over (for example, call control, domain control, or active notification). Some applications discard duplicate events, so in this case, there is no point in sending them across the ASAI CTI link. When enabled, this option allows only a single such event to be sent. The selection of the association on which the event will be sent is based on association precedence as follows: active notification (if enabled), call control (if enabled), or domain control (if enabled). Use the **Station** screen to change this option. The new option settings take effect the next time the ASAI link is activated.

Valid entries	Usage
y/n	Enter **y** to control the behavior for that particular link.

Send Disconnect Event for Bridged Appearance

Valid entries	Usage
y/n	Enter **y** to indicate that an event report is sent when a bridged appearance disconnects.

Special Character for Restricted Number

Enables an ASAI CTI link to indicate the calling number restricted presentation within an event report. For further information, see *Avaya Communication Manager ASAI Technical Reference*, 555-230-220.

Valid entries	Usage
y/n	When set to **y** and a calling number received in a SETUP message has the presentation indicator set (octet 3a in the calling number), then "*" is appended to the calling party number in the ASAI message.

Two-Digit Aux Work Reason Codes

Valid entries	Usage
y/n	Enter **y** to enable sending two digit Reason Codes over the ASAI link. All messages that include Aux Work Reason Codes allow a codes of 1 to 99. This field can only be set to **y** when **Two-Digit Aux Work Reason Codes?** on the Feature-Related System Parameters screen is set to **y**. Default is **n**.

Customer Options

See **System Parameters Customer-Options (Optional Features)**.

Data Module

The following section provides descriptions of standard fields on Data Module screens. Some of the fields are used for specific data module types; others are used for all data modules. Unique fields and fields requiring special consideration are listed with the appropriate data module descriptions in this book.

Field descriptions for page 1

Figure 59: Data Module screen

```
change data-module nn                                          Page   1 of x
                                DATA MODULE

   Data Extension: 30            Name: 27                             BCC:
             Type: data-line___   COS: 1        Remote Loop-Around Test?
             Port: _______        COR: 1          Secondary data module?
              ITC: restricted__    TN: 1                     Connected to: dte


  ABBREVIATED DIALING
    List1:

  SPECIAL DIALING OPTION:



  ASSIGNED MEMBER (Station with a data extension button for this data module)

            Ext     Name
        1: 1002    27 character   station name
```

BCC

(Bearer Capability Class) A display-only field used with Data Line, Netcon, Processor Interface, Point-to-Point Protocol, Processor/Trunk (**pdm** selection), and System Port Data Modules. Appears when the **ISDN-PRI** or **ISDN-BRI Trunks** field is **y** on the System Parameters Customer-Options (Optional Features) screen. The value in this field corresponds to the speed setting of the data module. This field can be compared with the BCC value in an associated routing pattern when attempted calls utilizing the data module fail to complete. The BCC values must be the same.

See Generalized Route Selection in *Avaya Aura™ Communication Manager Feature Description and Implementation,* 555-245-205, for a detailed description of Bearer Capability Classes (BCC) and their ability to provide specialized routing for various types of voice and data calls. The BCC value is used to determine compatibility when non-ISDN-PRI facilities are connected to ISDN facilities (ISDN-PRI Interworking).

Valid entries	Usage
1	Relates to 56-bkps
2, 3, 4	Relates to 64 kbps

Board

Used with Announcement Data Modules. Enter the five character announcement circuit pack number that identifies the physical circuit pack to which the announcement module is connected. You can enter **x** in this field to indicate that there is no hardware associated with this port assignment.

The five character announcement board number is comprised of:

Characters	Meaning	Value
1 to 2	Cabinet Number	**1** to **64** (S87XX Series IP-PNC)
3	Carrier	**A** to **E**
4 to 5	Slot Number or X	**0** to **20**

Broadcast Address

Used with Ethernet data modules. See *Administering Network Connectivity on Avaya Aura™ Communication Manager*, 555-233-504, for more information. Does not appear for S87XX Series IP-PNC.

Connected Data Module

Used with Processor Interface (used with DEFINITY CSI only) data modules.

Connected to

Used with Data Line and Processor/Trunk (**pdm** selection) Data Module. This field shows what the Asynchronous Data Unit (ADU) is connected to.

Valid entries	Usage
dte	Data Terminal Equipment. Used with Data Line and Processor/Trunk Data Modules.
isn	Information Systems Network. Used with Data Line and Processor/Trunk Data Modules.

COS

Does not appear for **ethernet**. Enter the desired class of service.

Valid entries	Usage
0 to **15**	Select the allowed features.

COR

Does not appear for **ethernet**. Enter the desired class of restriction.

Valid entries	Usage
0 to **995**	Select the allowed restriction.

Data Extension

A display-only field indicating the extension assigned to the data module.

Enable Link

Used with Point-to-Point, and Processor Interface data modules.

Establish Connection

Used with Point-to-Point, and Processor Interface (used with DEFINITY CSI only) data modules.

IP Address Negotiation

Used with Point-to-Point data modules. Does not appear for S87XX Series IP-PNC.

ITC

(Information Transfer Capability) Used with 7500, Announcement, data-line, Netcon, Processor/ Trunk (**pdm** selection), Processor Interface, and System Port Data Modules. Appears only when, on the Trunk Group screen, the **Comm Type** field is **56k-data** or **64k-data**. Indicates type of transmission facilities to be used for ISDN calls originated from this endpoint. Does not display for voice-only or BRI stations.

Valid entries	Usage
restricted	Either restricted or unrestricted transmission facilities are used to complete the call. A restricted facility is a transmission facility that enforces 1's density digital transmission (that is, a sequence of 8 digital zeros are converted to a sequence of 7 zeros and a digital 1).
unrestricted	Only unrestricted transmission facilities are used to complete the call. An unrestricted facility is a transmission facility that does not enforce 1's density digital transmission (that is, digital information is sent exactly as is).

Link

Used with Ethernet, Point-to-Point, and Processor Interface (used with DEFINITY CSI only) data modules. See *Administering Network Connectivity on Avaya Aura™ Communication Manager*, 555-233-504, for more information. This field is in different locations on the screen for different data module types.

Valid entries	Usage
1 to **99**	Enter a communication interface link number.

Maintenance Extension

Used with Netcon and Processor Interface Data Modules.

Valid entries	Usage
Enter the extension number required to perform maintenance functions on the standby netcon physical channel in a duplicated system.	The standby remote loop around tests fails if this field is not administered.

MM Complex Voice Ext

Used with 7500 and World Class BRI Data Modules. Does not appear on S87XX Series IP-PNC. This field contains the number of the associated telephone in the multimedia complex. This field appears only after you set the **Multimedia** field to **y**. This field is left blank until you enter the data module extension in **MM Complex Data Ext** on the Station screen.

Valid entries	Usage
Valid values conform to your dial plan	Once you complete the field on the Station screen, these two extensions are associated as two parts of a one-number complex, which is the extension of the telephone.

Multimedia

Used with the 7500 and World Class BRI Data Modules. Appears only if, on the System Parameters Customer-Options (Optional Features) screen, the **MM** field is **y**.

Valid entries	Usage
y/n	Enter **y** to make this data module part of a multimedia complex.

Name

Valid entries	Usage
Up to 27 alphanumeric characters	Enter the name of the user associated with the data module. The name is optional and can be blank. **NOTE:** Avaya BRI stations support ASCII characters only. They do not support non-ASCII characters, such as Eurofont or Kanafont. Therefore, if you use non-ASCII characters in any Communication Manager **Name** field, such characters do not display correctly on a BRI station.

Network uses 1's for Broadcast Addresses

Used with Ethernet data modules. See *Administering Network Connectivity on Avaya Aura™ Communication Manager*, 555-233-504, for more information.

Node Name

Used with Ethernet (not on S87XX Series IP-PNC) and Point-to-Point data modules. See *Administering Network Connectivity on Avaya Aura™ Communication Manager*, 555-233-504, for more information.

PDATA Port

Used with System Port Data Modules. Enter a seven-digit alphanumeric port location to which the data module is connected. Used to relate the physical PDATA port to which the mode 3 portion of the system port is connected. This entry must be assigned to a port on a PDATA Line Board.

Valid entries	Usage
01 to **22**	First and second characters are the cabinet number
01 to **64**	First and second characters are the cabinet number (S87XX Series IP-PNC)
A to **E**	Third character is the carrier
01 to **20**	Fourth and fifth characters are the slot number in the carrier
01 to **12**	Sixth and seventh characters are the circuit number

Physical Channel

Used with Netcon and Processor Interface Data Modules. The Physical Channel number is referred to on associated system forms as the Interface Link number.

Valid entries	Usage
01 to **08**	For Processor Interface Data Modules, enter the 2-digit circuit number of the Processor Interface port. A multi-carrier cabinet system supports the use of two Processor Interface circuit packs, the first circuit pack (mounted in Control Carrier A) supports physical channels or links 01 through 04; the second (mounted in Control Carrier A) supports physical channels or links 05 through 08. A single-carrier cabinet system supports one Processor Interface circuit pack and physical channels or links 01 through 04 only.
01 to **04**	For DEFINITY CSI configurations. For Netcon Data Modules, enter a netcon data channel.

Port

Used with 7500, Data Line, Ethernet, Processor/Trunk, PPP, System Port, and World Class BRI Data Modules. Specifies a port location to which the data module is connected.

Characters	Meaning	Value
1-2	Cabinet Number	**01** to **64** (S87XX Series IP-PNC)
3	Carrier	**A** to **E**
4-5	Slot Number	**0** to **20**
6-7	Circuit Number	**01** to **31** (S87XX Series IP-PNC (tdm, pdm) configurations) **01** to **16** (ppp for S87XX Series IP-PNC) **01** to **08** (system-port for S87XX Series IP-PNC) **17/33** (Ethernet on S87XX Series IP-PNC)

Note:
You can enter **x** in the **Port** field to indicate that there is no hardware associated with the port assignment (also known as Administration Without Hardware (AWOH). These stations are referred to as "phantom stations." If this data module is designated as a secondary data module (Secondary data module set to **y**) An **x** cannot be entered into this field. The port of a primary data module cannot be changed to **x** if a secondary data module is administered.

Remote Loop-Around Test

Used with Processor/Trunk Data Modules. Appears when the **Type** field is **pdm**, or **tdm**.

Valid entries	Usage
y/n	For Processor/Trunk Data Modules, enter **y** if the data module supports a loop-back test at the EIA interface. In general, Avaya equipment supports this test but it is not required by Level 2 Digital Communications Protocol. Enter **n** to abort a request for this test.

Secondary data module

Used with Processor/Trunk Data Modules. Appears only when the **Type** field is **pdm**. The primary data module must be administered before the secondary data module can be added. If the **Port** field is **x**, the **Secondary Data Module** field cannot be **y**.

Valid entries	Usage
y	This PDM is the secondary data module used for Dual I-channel AUDIX networking.
n	This is the primary PDM, or if this data module is not used for AUDIX networking.

Subnet Mask

Used with Point-to-Point data modules (for S87XX Series IP-PNC). See *Administering Network Connectivity on Avaya Aura™ Communication Manager*, 555-233-504, for more information.

TN

Valid entries	Usage
1 through **100**	Enter the Tenant Partition number.

Type

Enter the type of data module.

Valid entries	Usage
7500	Assigns a 7500 Data Module. The 7500 data module supports automatic TEI, B-channel, maintenance and management messaging, and SPID initialization capabilities. BRI endpoints, both voice and/or data, are assigned to either the ISDN-BRI - 4-wire S/T-NT Interface circuit pack or the ISDN-BRI - 2-wire U circuit pack. Each can support up to 12 ports. Since BRI provides multipoint capability, more than one ISDN endpoint (voice or data) can be administered on one port. For BRI, multipoint administration allows for telephones having SPID initialization capabilities, and can only be allowed if no endpoint administered on the same port is a fixed tie endpoint and no station on the same port has B-channel data capability. Currently, multipoint is restricted to 2 endpoints per port.
announcement	Assigns an announcement data module. The announcement data module is built-in to the integrated announcement circuit pack and is administered using the Announcement Data Module screen. This data module allows the system to save and restore the recorded announcements file between the announcement circuit pack and the system memory.
data-line	Assigns a Data Line Data Module. The Data Line Data Module (DLDM) screen assigns ports on the Data Line circuit pack (DLC) that allows EIA 232C devices to connect to the system. The DLC, with a companion Asynchronous Data Unit (ADU), provides a less expensive data interface to the system than other asynchronous DCP data modules. The DLC supports asynchronous transmissions at speeds of Low and 300, 1200, 2400, 4800, 9600, and 19200 bps over 2-pair (full-duplex) lines. These lines can have different lengths, depending on the transmission speed and wire gauge. The DLC has 8 ports. The connection from the port to the EIA device is *direct*, meaning that no multiplexing is involved. A single port of the DLC is equivalent in functionality to a data module and a digital line port. The DLC appears as a data module to the Digital Terminal Equipment (DTE) and as a digital line port to the server running Communication Manager. The DLC connects the following EIA 232C equipment to the system: ● Printers ● Non-Intelligent Data Terminals ● Intelligent Terminals, Personal Computers (PCs) ● Host Computers ● Information Systems Network (ISN), RS-232C Local Area Networks (LANs), or other data switches.
	1 of 2

Valid entries	Usage
ethernet	Assigns an Ethernet data module. The Ethernet Data Module screen assigns the 10BaseT port on the Control-LAN (C-Lan) circuit pack. This port provides a TCP/IP connection to network hub or LAN. See *Administering Network Connectivity on Avaya Aura™ Communication Manager*, 555-233-504, for more information on Ethernet data modules.
ni-bri	Assigns an NI-BRI Data Module.
pdm	Assigns a DCE interface for Processor/Trunk Data Modules. These screens assign Modular Processor Data Modules (MPDMs) and Modular Trunk Data Modules (MTDMs). One screen is required for assigning MPDMs (700D), 7400B, 7400D or 8400B Data Module, and another screen for MTDMs (700B, 700C, 700E, 7400A). One screen must be completed for each MPDM, 7400B, 7400D, 8400B or MTDM. The MPDM, 7400B, or 8400B Data Module provides a Data Communications Equipment (DCE) interface for connection to equipment such as data terminals, CDR output devices, on-premises administration terminal, Message Server, Property Management System (PMS), AUDIX, and host computers. It also provides a Digital Communications Protocol (DCP) interface to the digital switch. (DCE is the equipment on the network side of a communications link that provides all the functions required to make the binary serial data from the source or transmitter compatible with the communications channel.) The MTDM provides an Electronic Industries Association (EIA) Data Terminal Equipment (DTE) interface for connection to off-premises private line trunk facilities or a switched telecommunications network and a DCP interface for connection to the digital switch. (DTE is the equipment comprising the endpoints in a connection over a data circuit. For example, in a connection between a data terminal and a host computer, the terminal, the host, and their associated modems or data modules make up the DTE.) The MTDM or 7400A Data Module also can serve as part of a conversion resource for Combined Modem Pooling.
ppp	Assigns a Point-to-Point Protocol data module. The PPP Data Module screen assigns a synchronous TCP/IP port on the Control Lan (C-Lan) circuit pack. These ports are tailored to provide TCP/IP connections for use over telephone lines. See *Administering Network Connectivity on Avaya Aura™ Communication Manager*, 555-233-504, for more information on Point-to-Point data modules.
system-port	Assigns a System Port Data Module.
tdm	Assigns a DTE interface for Processor/Trunk Data Modules. See the **pdm** entry above.
wcbri	Assigns a World Class BRI Data Module.
	2 of 2

DESTINATION

CHAP

Appears when the **Type** field is **ppp**. Used with Point-to-Point data modules.

CHAP Secret

Appears when the **CHAP** field is **y**. Used with Point-to-Point data modules.

Valid entries	Usage
character string	Enter 1 to 30 characters; first character cannot be @.

Digits

This field appears when the **Type** field is **ppp**. Used with Point-to-Point data modules.

Valid entries	Usage
An extension, or Trunk Access Code (TAC) and extension of destination connection, or blank	Enter the number that the local data module dials to establish a connection to a far-end data module in a private network. .

Node Name

Appears when the **Type** field is **ppp**. Used with Point-to-Point data modules. See *Administering Network Connectivity on Avaya Aura™ Communication Manager*, 555-233-504 for more information.

ABBREVIATED DIALING

List1

Used with 7500, Data Line, Netcon, Processor/Trunk, Processor Interface, and World Class BRI Data Modules. Supports Data Hot Line. This field can be left blank.

Valid entries	Usage
e	Enhanced
g	Group. You also must enter a group list number.
p	Personal. You also must enter a personal list number.
s	System.

SPECIAL DIALING OPTION

Used with 7500, Data Line, Netcon, Processor/Trunk, Processor Interface, and World Class BRI Data Modules. Identifies the type of dialing for calls when this data module originates calls.

Valid entries	Usage
hot-line	Enter the destination number in the **HOT LINE DESTINATION** field. When user goes off-hook on the data module, the hot line destination number gets dialed.
default	Enter the destination number in the **DEFAULT DIALING LINE DESTINATION** field. When user goes off-hook on the data module and presses **Enter** at the **DIAL** prompt, the default dialing destination number gets dialed.
blank	For regular (normal) keyboard dialing.

HOT LINE DESTINATION

Abbreviated Dialing Dial Code

Used with 7500, Data Line, Netcon, Processor/Trunk, Processor Interface, and World Class BRI Data Modules. Appears only when the **Special Dialing Option** field is **hot-line**. Entry in this field supports Data Hot Line

Valid entries	Usage
0 to **999**	This number is associated with the AD List. When the user goes off-hook on a Data Hot Line call, the system dials the AD number.

DEFAULT DIALING

Abbreviated Dialing Dial Code

Used with 7500, Data Line, Netcon, Processor/Trunk, Processor Interface, and World Class BRI Data Modules. Only appears when the **Special Dialing Option** field is **default**. When the user goes off-hook and enters a carriage return following the DIAL prompt, the system dials the AD number. The data call originator can also perform data-terminal dialing by specifying a dial string that might or might not contain alphanumeric names.

Valid entries	Usage
0 to **999**	Enter a list number associated with the abbreviated dialing list.

CIRCUIT SWITCHED DATA ATTRIBUTES

Used with 7500 and World Class BRI Data Modules.

Note:
These fields represent defaults needed for modem pooling conversion resource insertion when the endpoint does not support data query capability and administered connections. These fields have no significance for data modules providing data query [all Avaya -supported ISDN-BRI data modules (7500 and ADM)]. For Avaya ISDN-BRI or World Class ISDN-BRI data modules, use the default settings.

Default Duplex

Used with 7500 and World Class BRI Data Modules. Used to identify the duplex mode.

Valid entries	Usage
full	Allows simultaneous two-way transmission.
half	Allows only one transmission direction at a time.

Default Mode

Used with 7500 and World Class BRI Data Modules. Used to identify the data mode.

Valid entries	Usage
sync	Synchronous
async	Asynchronous

Default Speed

Used with 7500 and World Class BRI Data Modules. Used to identify the data rate.

Valid entries	Usage
1200 **2400** **4800** **19200**	
56000 **64000**	Can be entered when the **Default Mode** field is **sync**.

ASSIGNED MEMBER

Ext and Name

Used with Data Line, Announcement, Netcon, Processor/Trunk, Processor Interface, and System Port Data Modules. Displays the extension number and name of the user (previously administered) with associated **Data Extension** buttons who shares the module.

DATA MODULE CAPABILITIES

Default Data Applications

Used with 7500, National ISDN, and World Class BRI Data Modules. Indicates the mode used for this data module when an administered connection has one of these types of data modules as the originator. See Generalized Route Selection in *Avaya Aura™ Communication Manager Feature Description and Implementation,* 555-245-205, for additional information.

Valid entries	Usage
M0	Mode 0. Use this setting for a WCBRI endpoint used as an administered connection.
M1	Mode 1
M2_A	Mode 2 asynchronous
M2_S	Mode 2 synchronous
M3/2	Mode 3/2 adaptable

Default ITC

Used with 7500, National ISDN, and World Class BRI Data Modules. Indicates the type of transmission facility used for this data module when an administered connection has one of these types of data modules as the originator.

Valid entries	Usage
restricted	Either restricted or unrestricted transmission facilities are used to complete the call. A restricted facility is a transmission facility that enforces 1's density digital transmission (that is, a sequence of 8 digital zeros are converted to a sequence of 7 zeros and a digital 1).
unrestricted	Only unrestricted transmission facilities are used to complete the call. An unrestricted facility is a transmission facility that does not enforce 1's density digital transmission (that is, digital information is sent exactly as is).

MM Complex Voice Ext:

Used with 7500 and World Class BRI Data Modules. This display-only field contains the number of the associated telephone in the multimedia complex. It only appears when the **Multimedia** field is **y**. This field is left blank until you enter the data module extension in **MM Complex Data Ext** field on the Station screen. Once you complete the field on the Station screen, these two extensions are associated as two parts of a one-number complex, which is the extension of the telephone. Valid values conform to your dial plan.

Field descriptions for page 2 - Type data-line

This version of page 2 appears when the **Type** field is **data-line**.

Figure 60: Data Line Data Module screen— if KYBD Dialing is y

```
change data-module nn                                                 Page 2 of x
                              DATA MODULE
        CAPABILITIES
                      KYBD Dialing? y                         Configuration? n
                          Busy Out? n
        SPEEDS
              Low? y           1200? y            4800? y             19200? y
              300? y           2400? y            9600? y        Autoadjust? n
        OPTIONS
                   Permit Mismatch? n                          Dial Echoing? y
               Disconnect Sequence: two-breaks                  Answer Text? y
                            Parity: even            Connected Indication? y
```

See DLC Option Settings on page 195 for additional information when assigning entries for the remaining fields on the screen.

CAPABILITIES

Busy Out

This option should be enabled for DTEs that are members of a hunt group and to allow "busy out" when DTE turns power off so that calls do not terminate on that DTE.

Valid entries	Usage
y/n	Enter **y** to place the DLC port in a busied-out state once the DTE control lead to the DLC is dropped.

Configuration

Appears when the **KYBD Dialing** field is **y**. This option normally is enabled for "originate/receive" DTE such as non-intelligent terminals and disabled for intelligent devices such as computers. The **KYBD Dialing** field must be **y** with this option.

Valid entries	Usage
y/n	Enter **y** to allow the viewing and changing of options from the DTE.

KYBD Dialing

This option must be enabled to allow data endpoints to originate calls via the EIA 232C interface and obtain ASCII feedback text. When enabled, the user gets the dial prompt. This option normally is enabled for "originate/receive" DTE that has a need to set up data calls. If this option is disabled, originations cannot be done at the DTE and text feedback does not occur at the DTE during call setup/take down. Data call answering is still allowed but without text feedback.

Note:
ADU-type hunt groups connecting the system to terminal servers on a host computer should have these hunt group extensions assigned as "no" keyboard dialing.

Valid entries	Usage
y/n	Enter **y** to allow keyboard dialing. This enables the data endpoint to receive and transmit text during call origination or termination. Low must be **n**.

SPEEDS

Enter **y** to select operating speeds as follows:

Valid entries	Usage
Low	Enter **y** to instruct the DLC to operate at a low speed from 0 to 1800 bits per second (bps). Enter **n** if the **KYBD Dialing** field is **y**.
300, 1200, 2400, 4800, 9600, or 19200	Enter **y** beside the desired operating speed. Enter **n** if the speed is not desired. The DLC can be any one of these speeds. The speed is matched for the duration of the call, from call setup to call takedown. When multiple speeds are selected (select three or more, do not select just two speeds) and autoadjust is disabled, the DTE’s speed must be the highest selected speed. This is required because all feedback text is delivered to the DTE at the highest selected speed.
Autoadjust	Appears when the **KYBD Dialing** field is **y**. Enter **y** which tells the DLC port to automatically adjust to the operating speed and parity of the DTE it is connected to. Enter **n** if this option is not desired. Autoadjust can be selected with any of the speeds selected in the previous step. Autoadjust allows the DLC port to determine the speed and parity of the DTE and then match itself to this speed. Autoadjust only applies to calls originated by the user through Keyboard Dialing.

OPTIONS

Answer Text

Appears when the **KYBD Dialing** field is **y**. This option enables text feedback that is normally delivered to the DTE when a call is answered or disconnected. The Answer Text option applies to DLC-generated text as well as text received from the system. If this option is disabled, the system still generates the text, but the DLC prevents it from being sent to the device.

This applies to the following messages:

- INCOMING CALL
- ANSWERED
- DISCONNECTED
- DISCONNECTED OTHER END

This option usually is disabled when the answering DTE is a computer or an intelligent device.

Valid entries	Usage
y/n	Enter **y** to allow text messages to be delivered to the DTE when a call is being answered.

Connected Indication

Appears when the **KYBD Dialing** field is **y**. This option generates a "CONNECTED" message to the DTE when the connection has been established. If the **KYBD Dialing** field is **n**, the connected indication is provided by the DLC activating its EIA 232C control lead.

Valid entries	Usage
y/n	Enter **y** to select this option.

Dial Echoing

Appears when the **KYBD Dialing** field is **y**.

Valid entries	Usage
y/n	Enter **y** to echo characters back to the DTE. Dial echoing should be disabled when keyboard dialing is done by an intelligent device.

Disconnect Sequence

Appears when the **KYBD Dialing** field is **y**. Selects the sequence for a disconnect.

Valid entries	Usage
long-break	A long-break is greater than 2 seconds.
two-breaks	Two-breaks is within 1 second.

Parity

Appears when the **KYBD Dialing** field is **y**. Select the desired type of parity. The DLC generates the parities when call setup text is sent to the DTE. The DLC does not check the parity when receiving dialing characters. Parity has nothing to do with the far end; it is used by the DLC to terminal communications during call setup. Set to match the connected DTE.

Valid entries	Usage
even	Set to match the connected DTE.
odd	
mark	
space	

Permit Mismatch

This option allows the EIA interface to operate at a rate different than that agreed to in the data module handshake. (The data module handshake is always the highest compatible rate as determined by the reported speed option of each data module.) Permit Mismatch eliminates the need to change the DTE/DLC speed every time a call is placed to/from an endpoint operating at a different speed. When this option is enabled, the DLC reports the highest optioned speed and all the lower speeds (or the previously selected autoadjust speed) during the handshake process.

Valid entries	Usage
y/n	Enter **y** to instruct the DLC to operate at the highest selected speed, which is a higher rate than the far-end data module.

CAUTION:
Caution must be used when using this option to send information from a DTE/DCE that is transmitting data at higher rates than that of the far end. Sustained usage of this type transmission results in loss of data. Whenever this option is enabled, the DTE must match the highest speed selected for the associated DLC port.

This option is intended to be used by a DTE device operating locally at a higher baud rate than that of its far-end connection but transmitting relatively low amounts of data (for example, a user typing at a terminal). Also, this option can be selected whether or not Keyboard Dialing is selected.

Note:
The Low speed setting is not reported as an available speed when the **Permit Mismatch** field is **y**.

DLC Option Settings

The following provides additional information on the option settings for DLCs when used with the following types of devices:

- Printers
- Non-intelligent terminals
- Data terminals and personal computers
- Host computers
- Information Systems Network (ISN)

Printers

A DLC port with a companion ADU, when attached to a printer, usually terminates a data call. Therefore, in this connection, the printer is the endpoint device. The originating device might be attached to a DCP mode 2 data module (such as the MPDM) or the DLC. A Z3A ADU extends the range of the EIA 232C connection.

When a receive-only printer (or any printer that does not generate the Transmit Data and DTR leads) is used, the ADU must be powered from a small plug-mounted transformer (2012D, or equivalent) connected to pins 7 and 8 of the modular jack. (See **ADU User Manual** for details.)

An ADU cannot be used if the printer has hardware flow control using the Clear To Send (CTS) lead. An ADU can be used, however, if the printer is using software flow control.

A printer connected to a DLC is usually assigned as a line. Table 1 lists the option settings for printer connections.

Table 1: DLDM screen settings for printer connection

Field on screen	Option	Comments
Speed	Highest speed at which the Printer operates	Subject to distance limitations; Autoadjust not used
KYBD Dialing	no	
Busy Out	yes	If printer is member of Hunt Group
Permit Mismatch	yes	No, if printer is low speed
Parity	-	Don't care
Dial Echoing	-	Don't care
Disconnect Sequence	-	Don't care
		1 of 2

Table 1: DLDM screen settings for printer connection (continued)

Field on screen	Option	Comments
Answer Text	-	Don't care
Connected Indication	-	Don't care
Configuration	no	
		2 of 2

Non-intelligent terminals

A non-intelligent terminal connected to the DLC usually is assigned as a line. Table 2 lists the option settings for non-intelligent terminals.

Table 2: DLDM screen settings for connection to non-intelligent terminals

Field On screen	Option	Comments
Speed	All speeds at which the terminal can operate; autoadjust	Subject to distance limitations; Autoadjust when the **KYBD Dialing** field is **y** and the Terminal can generate an ASCII "return"
KYBD Dialing	yes	
Busy Out	no	Yes, if terminal is member of a hunt group
Permit Mismatch	yes	-
Parity	Same as DTE	
Dial Echoing	yes	Only if the **KYBD Dialing** field is **y**
Disconnect Sequence	2	Depends on terminal
Answer Text	yes	
Connected Indication	-	Don't care
Configuration	yes	

Data terminals and personal computers

An intelligent data terminal or a personal computer (PC) attached to a DLC can either originate or terminate a data call. A single ADU at the site of the originating device extends the distance signals can travel to the switch (the model ADU depends on the terminal connector). An analog telephone can be attached to this arrangement whenever an ADU uses the standard building wiring. Table 3 lists the option settings used for data terminal and personal computer connections.

Table 3: DLDM screen settings for connection to data terminal or personal computer

Field on screen	Option	Comments
Speed	All speeds at which the Data Terminal or PC can operate	Subject to distance limitations; Autoadjust not used
KYBD Dialing	yes	
Busy Out	no	Yes, if device is accessed through a hunt group
Permit Mismatch	yes	No, if device does not support XON/XOFF flow control
Parity	Same as DTE	
Dial Echoing	no	These devices can dial in the ASCII stream without human intervention
Disconnect Sequence	Long <BREAK>	-
Answer Text	no	These devices might not want to see any text
Connected Indication	-	Don't care
Configuration	yes	

Host computers

A host computer can originate and terminate a data call. For this application, the number of DLCs required depends on the number of ports needed. An MADU can be used (instead of 8 ADUs) to complete the connection. Table 4 lists option settings for a port that has a terminating connection to a host computer or an originating connection from a host computer.

Note:
If the **KYBD Dialing** field is **n**, the rest of the option settings are irrelevant.

Table 4: DLDM screen settings for terminating connection to host computer

Field on screen	Option	Comments
Speed	All speeds at which the computer can operate	Subject to distance limitations; Autoadjust not used
KYBD Dialing	no	
Busy Out	-	Don't care
Permit Mismatch	-	Don't care
Parity	-	Don't care
Dial Echoing	-	Don't care
Disconnect Sequence	-	Don't care
Answer Text	-	Don't care
Connected Indication	-	Don't care
Configuration	-	Don't care

Field Descriptions for page 2 - Type 7500, WC-BRI, NI-BRI

This version of page 2 appears when **Type** is 7500, WC-BRI, and NI_BRI

Figure 61: 7500, World Class BRI, and NI-BRI Data Module screen

```
change data-module nn                                          Page 2 of x
                                 DATA MODULE
  BRI LINK/MAINTENANCE PARAMETERS
                   XID? y      Fixed TEI? n      TEI: ___
          MIM Support? y     Endpt Init? y     SPID: 300_____    MIM Mtce/Mgt? y
```

BRI LINK/MAINTENANCE PARAMETERS

Country Protocol

Used with World Class BRI data modules. Enter the protocol that corresponds to your supported initialization and codesets. The Country Protocol must match any previously-administered endpoint on the same port. The following table lists the valid protocol entries. For a list of country codes, see the Country code table on page 886.

Country/Area	Protocol
Australia	2
ETSI (Europe)	etsi
Japan	3
Singapore	6
United States (Bellcore National ISDN)	1

Endpt ID

Used with World Class BRI and NI-BRI data modules. Appears only if the **Endpt Init** field is **y**. This field provides for multipoint configuration conformance to the Bellcore Terminal Initialization procedures. In these procedures, a multipoint configuration requires that the last 2 digits of the Service Profile Identifier (SPID) be between **00** and **63** and be binary unique for each endpoint. This field, combined with the SPID, gives the effective SPID administered into the terminal. Bellcore ISDN-1 requires that the SPID programmed into the endpoint contain at least 9 digits. (For example, if the **SPID** field is **1234**, and the **Endpt ID** field is set to **01**, then the SPID administered on the terminal is 000123401. The three leading zeros are necessary to create a 9-digit SPID.)

Valid entries	Usage
00 to **62**	Enter a 2-digit number. Each **Endpt ID** field must have a unique value for each endpoint on the same port.

Endpt Init

Used with 7500, World Class BRI, and NI-BRI Data Modules. Endpoint initialization is a procedure, required for multipoint operation, by which User Service Order Profile (USOP) is associated with an endpoint on the ISDN-BRI. This association is made via the Service Profile Identifier (SPID), administered into the system and entered into the ISDN-BRI terminal. For a ISDN-BRI terminal to become operational in a multipoint configuration, both the administered SPID and the SPID programmed into the ISDN-BRI terminal must be the same. This means that the SPID of the new or re-used terminals must be programmed to match the administered SPID value.

Valid entries	Usage
y/n	Indicates the terminal’s endpoint initialization capability.

Fixed TEI

Used with 7500, World Class BRI, and NI-BRI Data Modules. Used to indicate whether the endpoint has Fixed Terminal Equipment Identifier (TEI) capability. TEI identifies a unique access point within a service. For Fixed TEI stations, the TEI must be administered. Terminals with automatic TEI capability, the associated TEI is assigned by the system.

Valid entries	Usage
y/n	Enter **y** to indicate the endpoint has Fixed Terminal Equipment Identifier (TEI) capability.

MIM Mtce/Mgt

Used with 7500 Data Modules.

Valid entries	Usage
y/n	Management Information Message (MIM) Support. Entering **y** indicates the terminal supports MIM Maintenance and Management capabilities, other than endpoint initialization.

MIM Support

Used with 7500 Data Modules.

Valid entries	Usage
y/n	Used to support two types of capabilities: MIM endpoint initialization capability (SPID support), and other Maintenance/Management capability.

SPID

Used with 7500, World Class BRI, and NI-BRI Data Modules. Appears only if the **Endpt Init** field is **y**. The Service Profile Identifier (SPID) is a variable parameter of up to 10 digits. The SPID must be different for all terminals on the ISDN-BRI and from the Service SPID. The SPID should always be assigned. If the SPID is not assigned for the first ISDN-BRI on a port, any other ISDN-BRI assignment to that port is blocked.

Valid entries	Usage
0 to **9999999999**	Assign a Service Profile Identifier (SPID) for this data module.

TEI

Used with 7500, World Class BRI, and NI-BRI Data Modules. Appears only if the **Fixed TEI** field is **y**.

Valid entries	Usage
0 to **63**	Enter a 1 to 2-digit number.

XID

(Exchange identification) Used with 7500, World Class BRI, and NI-BRI Data Modules. Used to identify layer 2 XID testing capability.

Valid entries	Usage
y/n	Avaya recommends setting to **n**.

Date and Time

Use this screen to set the system date and time, to select the daylight savings plan number, if any, and to show whether the current time is standard time or daylight savings. Settings on this screen affect the internal clock and timestamp of the server running Communication Manager. You should update the date and time for a leap year or a system restart after a power failure. The correct date and time assure that CDR records are correct. CDR does not work until the date and time have been entered.

For additional information, see *Avaya Aura™ Call Center 5.2 Automatic Call Distribution (ACD) Reference*, 07-602568.

Field descriptions for page 1

Figure 62: Date and Time screen

```
set time                                                                 Page 1
                                DATE AND TIME
DATE
   Day of the Week: _______Month: ________
  Day of the Month: __ Year: ____

TIME
  Hour: __ Minute: __Second: __                Type: ________________

        Daylight Savings Rule: _

WARNING: Changing the date or time may impact BCMS, CDR, SCHEDULED EVENTS,
      and MEASUREMENTS
```

Day of the Month

Valid entries	Usage
1 to **31**	Enter the current day of the month. The system clock uses this as the current date.

Day of the Week

Valid entries	Usage
Sunday through **Saturday**	Enter the current day of the week. The system clock uses this as the current day.

Daylight Savings Rule

This field displays which daylight savings rule is in use for your system.

Valid entries	Usage
0 to **15**	Enter the appropriate rule number. The system clock uses this as the current daylight savings rule. These rules are defined on the Daylight Savings Rules screen.

Hour

The system uses a 24-hour clock. For example, 14:00 is the same as 2:00 p.m.

Valid entries	Usage
0 to **23**	Enter the current hour to be used by the system clock.

Minute

Valid entries	Usage
0 to **59**	Enter the current minute. The system clock uses this as the current minute.

Month

Valid entries	Usage
January through **December**	Enter the current month. The system clock uses this as the current month.

Second

This display-only field shows the seconds and cannot be modified. It resets to zero when you save the information on this screen.

Type

Valid entries	Usage
daylight-savings	Enter **daylight-savings** to indicate daylight savings time is in effect.
standard	Enter **standard** to indicate standard time is in effect.

Year

Valid entries	Usage
1990 to **2099**	Enter the current year. The system clock uses this as the current year.

Related topics

To update the date and time for the change to or from daylight savings time, use the Daylight Saving Rule screen.

Daylight Savings Rules

Use this screen to enter up to 15 customized daylight savings rules. You can specify the day, month, date, time, and increment each daylight savings rule goes into effect and the day, month, date, and time it stops. Rule 0 makes no adjustment to the system clock for daylight savings and cannot be modified. Rule 1 applies to all time zones in the U.S. and begins on the first Sunday on or after March 8 at 2:00 a.m. with a 01:00 increment. Daylight Savings Time stops on the first Sunday on or after November 1 at 2:00 a.m., also with a 01:00 increment (used as a decrement when switching back to Standard time. Telephone displays are affected by these settings.

Field descriptions for page 1

Figure 63: Daylight Savings Rules screen

```
change daylight-savings-rulesPage 1 of 2
DAYLIGHT SAVINGS RULES
Rule   Change DayMonth  Date   Time    Increment

0:No Daylight Savings

1:Start: first __Sunday_ on or after _March___ 8_ at 2:00_     ___01:00__
  Stop:  first __Sunday_ on or after November 1_ at 2:00_
2:Start: first _________ on or after ________ __ at __:__
  Stop:  first _________ on or after ________ __ at __:__
3:Start: first _________ on or after ________ __ at __:__
  Stop:  first _________ on or after ________ __ at __:__
4:Start: first _________ on or after ________ __ at __:__
  Stop:  first _________ on or after ________ __ at __:__
5:Start: first _________ on or after ________ __ at __:__
  Stop:  first _________ on or after ________ __ at __:__
6:Start: first _________ on or after ________ __ at __:__
  Stop:  first _________ on or after ________ __ at __:__
7:Start: first _________ on or after ________ __ at __:__
  Stop:  first _________ on or after ________ __ at __:__
```

Change day (Start)

Valid entries	Usage
Sunday through **Saturday** or **Day**	Enter the day of the week you want the clock to move ahead to begin daylight savings. If you enter **Day** in this field, the clock changes on the exact date entered in the next two fields.

Change day (Stop)

Valid entries	Usage
Sunday through **Saturday** or **Day**	Enter the day of the week you want the clock to move back to return to standard time. If you enter **Day** in this field, the clock changes on the exact date entered in the next two fields.

Date (Start)

Valid entries	Usage
0 to **31**	Enter the day of the month you want the clock to move ahead to begin daylight savings.

Date (Stop)

Valid entries	Usage
0 to **31**	Enter the date you want the clock to move back to return to standard time.

Increment (Start)

Valid entries	Usage
0 to **23**	Enter the number of hours you want the clock to move ahead for daylight savings and to move back to return to standard time.
0 to **9**	Enter the number of minutes you want the clock to move ahead for daylight savings and to move back to return to standard time.

Month (Start)

Valid entries	Usage
January through **December**	Enter the number of hours you want the clock to move ahead for daylight savings and to move back to return to standard time.

Month (Stop)

Valid entries	Usage
January through **December**	Enter the number of hours you want the clock to move ahead for daylight savings and to move back to return to standard time.

Rule

This display-only field indicates the daylight savings rule number.

Time (Start)

The system uses a 24-hour clock. For example, 14:00 is the same as 2:00 p.m.

Valid entries	Usage
0 to **23**	Enter the hour you want the clock to move ahead to begin daylight savings.
0 to 59	Enter the minute you want the clock to move ahead to begin daylight savings.

Time (Stop)

The system uses a 24-hour clock. For example, 14:00 is the same as 2:00 p.m.

Valid entries	Usage
0 to **23**	Enter the hour you want the clock to move back to return to standard time.
0 to **59**	Enter the minute you want the clock to move back to return to standard time.

DCS to QSIG TSC Gateway

Use the DCS to QSIG TSC Gateway screen to determine when and how to convert messages from an administered AUDIX NCA-TSC to a QSIG NCA-TSC. This screen maps the AUDIX NCA-TSC to the appropriate machine ID index to find the QSIG subscriber entry in the QSIG MWI-Prefix screen. It also assigns the voice mail number to be used when a DCS served-user node interrogates a QSIG message center.

This screen only appears if the Interworking with **DCS** field is enabled on the System Parameters Customer-Options (Optional Features) screen.

Field descriptions for page 1

Figure 64: DCS to QSIG TSC Gateway screen

```
change isdn dcs-qsig-tsc-gateway                                  Page 1 of x

                                DCS TO QSIG TSC GATEWAY

                                  AAR/                                          AAR/
                                  ARS                                          ARS
Mach  Sig   TSC    Voice Mail    Access    Mach  Sig   TSC     VoiceMail    Access
 ID   Grp  Index    Number        Code      ID   Grp  Index     Number      Code
 __   __   ___   ____________    _____     ___   ___  ____   _________     ____
 __   __   ___   ____________    _____     ___   ___  ____   _________     ____
 __   __   ___   ____________    _____     ___   ___  ____   _________     ____
 __   __   ___   ____________    _____     ___   ___  ____   _________     ____
 __   __   ___   ____________    _____     ___   ___  ____   _________     ____
 __   __   ___   ____________    _____     ___   ___  ____   _________     ____
 __   __   ___   ____________    _____     ___   ___  ____   _________     ____
 __   __   ___   ____________    _____     ___   ___  ____   _________     ____
 __   __   ___   ____________    _____     ___   ___  ____   _________     ____
 __   __   ___   ____________    _____     ___   ___  ____   _________     ____
 __   __   ___   ____________    _____     ___   ___  ____   _________     ____
```

AAR/ARS Access Code

This field can be left blank.

Valid entries	Usage
0 to **9**, *****, **#**	Enter up to 4-digit access code.

Mach ID

You can enter up to 20 machine IDs.

Valid entries	Usage
1 to **20**	Enter a unique machine ID. The system does not allow you to specify an ID that you already entered on the Processor Channel screen.

Sig Grp

You must complete the **Signaling Group** field for each machine ID.

Valid entries	Usage
1 to **110**	Enter the assigned signaling group number between 1 and 110 for DEFINITY CSI.
1 to **650**	Enter the assigned signaling group number between 1 and 650 for S8300/S87XX Servers.

TSC Index

You must complete the **TSC Index** field for each machine ID.

Valid entries	Usage
1 to **64**	Enter the assigned signaling group number for **qsig-mwi** application type on the Signaling Group screen.

Voice Mail Number

This field can be left blank.

Valid entries	Usage
0 to **9**	Enter the complete Voice Mail Dial Up number up to 15 digits.

Dial Plan Analysis Table

The Dial Plan Analysis Table is the system's guide to translating the digits dialed by users. This screen enables you to determine the beginning digits and total length for each type of call that Communication Manager needs to interpret. The Dial Plan Analysis Table and the Dial Plan Parameters screen work together to define your system's dial plan.

Note:

In Communication Manager 5.0 and later, you can administer dial plans per-location. Typing the command **`change dialplan analysis n`** displays the all-locations Dial Plan Analysis screen. The ***n*** specifies that dialed strings beginning with the value **`n`** are displayed first. To access a per-location screen, type **`change dialplan analysis location n`**, where ***n*** represents the number of a specific location. For details on command options, see online help, or *Maintenance Commands for Avaya Aura™ Communication Manager, Media Gateways and Servers*, 03-300431.

Field descriptions for page 1

Figure 65: Dial Plan Analysis Table screen

```
change dialplan analysis                                          Page 1 of x
                              DIAL PLAN ANALYSIS TABLE
                                    Location: All         Percent Full: 7

       Dialed  Total  Call         Dialed  Total  Call        Dialed  Total  Call
       String  Length Type         String  Length Type        String  Length Type
         00      2    attd
         1       3    dac
         2       4    ext
         3       4    ext
         3       1    aar
         4       1    ars
         4       5    ext
         5       5    ext
         5       7    ext
         6       5    ext
         7210    7    ext
         8       7    ext
         9       1    fac
         *       3    fac
         #       3    fac
```

Call Type

Valid entries	Usage
aar	Automatic Alternate Routing — Used to route calls within your company over your own private network. In order to use this code in your dial plan, the ARS/AAR Dialing without FAC feature must be enabled on the System Parameters Customer-Options **(Optional Features)** screen. (Contact your Avaya technical support representative to discuss the ARS/AAR Dialing Without FAC feature before enabling it.) When dialing digits of Call Type aar, as soon as the dialed digits have reached the administered length, the digits are treated as if an AAR feature access code (FAC) was dialed. Control is transferred and the digits are routed according to the AAR Analysis and Digit Conversion forms. In the example shown on the Dial Plan Analysis Table on page 209, extensions of 3xxx cannot be dialed directly. Whenever a user dials the first digit of 3, the system immediately interprets the dialed string as an AAR string and transfers control to AAR. Extensions of 3xxx can only be accessed using AAR Digit Conversion. That is, you must dial a longer AAR number from which AAR Digit Conversion deletes leading digits to form a number of the form 3xxx.
ars	Automatic Route Selection — Used to route calls that go outside your company over public networks. ARS is also used to route calls to remote company locations if you do not have a private network. In order to use this code in your dial plan, the ARS/AAR Dialing without FAC feature must be enabled on the System Parameters Customer-Options **(Optional Features)** screen. (Contact your Avaya technical support representative to discuss the ARS/AAR Dialing Without FAC feature before enabling it.) When dialing digits of Call Type ars, as soon as the dialed digits have reached the administered length, the digits are treated as if an ARS feature access code (FAC) was dialed. Control is transferred and the digits are routed according to the ARS Analysis and Digit Conversion forms. In the example shown on the Dial Plan Analysis Table on page 209, extensions of 4xxxx cannot be dialed directly. Whenever a user dials the first digit of 4, the system immediately interprets the dialed string as an ARS string and transfers control to ARS. Extensions of 4xxxx can only be accessed using ARS Digit Conversion. That is, you must dial a longer ARS number from which ARS Digit Conversion deletes leading digits to form a number of the form 4xxxx.
	1 of 3

Valid entries	Usage
attd	Attendant — Defines how users call an attendant. Attendant access numbers can start with any number from 0 to 9 and contain 1 or 2 digits. If a telephone's COR restricts the user from originating calls, this user cannot access the attendant using this code. Beginning with the November 2003 release of Communication Manager (2.0), you can also administer the attendant access code by entering an appropriate **fac** or **dac** entry on the Dial Plan Analysis screen, and then entering the actual access code on the Feature Access Code (FAC) screen. Location-specific attendant access codes can be administered on the Locations screen.
dac	Dial access code — Allows you to use trunk access codes (TAC) and feature access codes (FAC) in the same range. Dial access codes can start with any number from 0 to 9, * or # and can contain up to 4 digits. If an extension entry and a DAC entry have the same Dialed String, the extension entry can be longer than the DAC entry only if all of the trunk groups covered by that DAC entry have **Dial Access** on the Trunk Group screen set to **n**. You can use the DAC to activate or deactivate a Communication Manager feature or to seize a trunk from a trunk group, or both. In the first case, the DAC functions as a FAC, in the second as a TAC. For example, you can define the group 300 to 399 for dial access codes, and allow both FAC and TAC in that range. You can use 4-digit DACs for ordinary trunk access, but they do not work for attendant control of trunk groups, trunk-ID buttons, or DCS, and only the last 3 digits of the codes can be recorded in CDR records. See also the description below for **fac**.
ext	Primary extension — Defines extension ranges that can be used on your system. Extension can have a first digit of 0 through 9 and can be 1 to 7 digits in length. Extension cannot have the same first digit as a 1-digit ARS or AAR feature access code (FAC). When a dial plan has mixed station numbering, extensions of various lengths (all with the same first digit) are mapped on the Dial Plan Analysis table. The system then employs an inter-digit time-out to ensure that all dialed digits are collected.
fac	Feature access code only — A FAC can be any number from 1 to 9 and contain up to 4 digits. You can use * or #, but only as a first digit. Avaya recommends that a FAC have the longest total length for a given dialed string when using mixed numbering. Otherwise, problems might occur when, for example, 3-digit FACs and 4-digit extensions begin with the same first digit and the FAC is an abbreviated dialing list access code. However, if the entry in the dial plan that defines the FAC is used to define the AAR or ARS access code, then it *must* have the longest total length in the dial plan.
	2 of 3

Valid entries	Usage
pext	Prefixed extension — Is made up of a prefix (first digit) that can be a **0** to **9** (* and # not allowed) and an extension number of up to 5 digits in length. The maximum length of a prefix and extension combination is 6 digits. You cannot administer a dial access code with the same first digit as a prefixed extension. The purpose of the prefix is to identify the call type as an extension. After digit collection, the prefix digit is removed from the string of dialed digits. The remaining digits (extension number) are then processed. A prefixed extension allows the use of extensions numbers with any dialed string (the extension length must be specified on the table). The "prefixed extension" cannot have the same dialed string as the ARS or AAR facility access code (FAC).
udp	Works identically to **ext**, with this exception: ● If dialed digits match the **Call Type udp**, Communication Manager automatically checks the UDP Table first to see if there is a match, regardless of the value in the **UDP Extension Search Order** field on the **Dial Plan Parameters** screen. If there is no match, Communication Manager then checks the local server. ● If dialed digits match the **Call Type** of **ext**, Communication Manager checks the value in the **UDP Extension Search Order** field on the **Dial Plan Parameters** screen. – If the value in the **UDP Extension Search Order** field on the **Dial Plan Parameters** screen is **udp-table-first**, Communication Manager checks the UDP Table first to see if there is a match. If there is no match, Communication Manager then checks the local server. – If the value in the **UDP Extension Search Order** field on the **Dial Plan Parameters** screen is **local-extensions-first**, Communication Manager checks the local server first to see if there is a match. If there is no match, Communication Manager then checks the UDP Table. **Note**: The **udp Call Type** allows Communication Manager to recognize strings of 14 and 15 digits, which are longer than the maximum extension length of 13 digits. However, **udp** can be used with any length.
	3 of 3

Dialed String

The dialed string contains the digits that Communication Manager analyzes to determine how to process the call. This field allows you to enter up to four digits, so you can allocate blocks of 1000 numbers even when using a 7-digit dial plan

.

Valid entries	Usage
0 to **9**, ***** and #	Enter any combination of 1 to 4 digits. the following restrictions apply: ● The digits * and # can only be used as first digits, and only for the Call Types **fac** and **dac**. ● For Call Type **attd**, if the Total Length is 2, the Dialed String must be 2 digits long. ● Two Dial Plan entries can use the same Dialed String only if the Dialed String is 1 digit long. Longer Dialed Strings must all be unique. ● A new entry cannot be administered if it causes an existing extension, feature access code, or trunk access code to become inaccessible.

Location

This is a display-only field. Typing the command **`change dialplan analysis`** displays the all-locations screen, and populates this field with **all**. The ***n*** specifies that dialed strings beginning with the value **`n`** are displayed first. To access a per-location screen, type **`change dialplan analysis location`** ***n***, where ***n*** represents the number of a specific location. This field then displays the number of the specified location. For details on command options, see online help, or *Maintenance Commands for Avaya Aura™ Communication Manager, Media Gateways and Servers*, 03-300431.

Valid entries	Usage
1 to **64**	Defines the location of the server running Communication Manager that uses this Dial Plan Analysis Table. On the System Parameters Customer-Options (Optional Features) screen, the **Multiple Locations** field must be set to **y** for values other than **all** to appear.
all	Indicates that this Dial Plan Analysis Table is the default for all port network (cabinet) locations. Appears only if the **Multiple Locations** field is **n** on the System Parameters Customer-Options (Optional Features) screen.

Percent Full

Displays the percentage (0 to 100) of the system's memory resources that have been allocated for the dial plan that are currently being used.

Total Length

Valid entries	Usage
1 to **2** for **attd** **1** to **4** for **dac** **1** to **4** for **fac** **1** to **7** for **ext** **2** to **6** for **pext**	Enter the number of digits for this call type. The allowed length varies by call type. This must be greater than or equal to the number of digits in the Dialed String.

Dial Plan Parameters

The Dial Plan Parameters screen works with the Dial Plan Analysis Table to define your system's dial plan.

It also controls the appearance of digit extensions on station displays. These multi-digit extensions can be hard to read when displayed as a block. Communication Manager allows you to select the display format for 6-13 digit extensions.

Field descriptions for page 1

Figure 66: Dial Plan Parameters screen

```
change dialplan parameters                                        Page 1 of x
                              DIAL PLAN PARAMETERS

                   Local Node Number: 2
                     ETA Node Number:
                 ETA Routing Pattern:
          UDP Extension Search Order: local-extensions-first
        AAR/ARS Internal Call Prefix:
  AAR/ARS Internal Call Total Length:
Retry ARS Analysis if All-Location Entry Inaccessible? n

EXTENSION DISPLAY FORMATS

                           Inter-Location/SAT         Intra-Location
 6-Digit Extension:        xx.xx.xx                   xx.xx.xx
 7-Digit Extension:        xxx-xxxx                   xxx-xxxx
 8-Digit Extension:        xx.xx.xx.xx                xx.xx.xx.xx
 9-Digit Extension:        xxx-xxx-xxx                xxx-xxx-xxx_
10-Digit Extension:        xxx-xxx-xxxx               xxx-xxx-xxxx_
11-Digit Extension:        xxxx-xxx-xxxx              xxxx-xxx-xxxx
12-Digit Extension:        xxxx-xxx-xxxx              xxxxxxxx-xxxx
13-Digit Extension:        xxxxxxxxxxxxx              xxxxxxxxxxxxx
```

AAR/ARS Internal Call Prefix

The digits entered in this field are concatenated with the calling or called extension. Appears only if, on the System Parameters Customer-Options (Optional Features) screen, the **ARS/AAR Dialing Without FAC** field is **y**. (Contact your Avaya technical support representative to discuss the ARS/AAR Dialing Without FAC feature before enabling it.)

Valid entries	Usage
0 to **9** up to 8 digits**,** or blank	Enter a string from 1 to 8 digits long (not including * or #).

AAR/ARS Internal Call Total Length

The total length of the internal call digit string. Appears only if, on the System Parameters Customer-Options (Optional Features) screen, the **ARS/AAR Dialing Without FAC** field is **y**. (Contact your Avaya technical support representative to discuss the ARS/AAR Dialing Without FAC feature before enabling it.)

Valid entries	Usage
6 to **10** or blank	Enter the total length of the internal call digit string, which includes the Internal Call Prefix and the calling or called extension.

Note:
If either the **AAR/ARS Internal Call Prefix** or the **AAR/ARS Internal Call Total Length** field is non-blank and valid, the other must also be non-blank and valid. In addition, the longest extension length on the Dial Plan Analysis screen, plus the length of the **ARS/AAR Internal Call Prefix**, must equal or be greater than, the **ARS/AAR Internal Call Total Length** value.

ETA Node Number

Enter the number of the destination server for Extended Trunk Access (ETA) calls. ETA calls are unrecognized numbers you can send to another switch for analysis and routing. Such numbers can be Facility Access Codes, Trunk Access Codes, or extensions that are not in the UDP table.

Valid entries	Usage
1 to **999** or blank	Enter the number of a destination server.

ETA Routing Pattern

Enter the number of the routing pattern to reach the destination server.

Valid entries	Usage
1 to **640** or blank	Avaya DEFINITY ServerCSI.
1 to **999** or blank	Avaya S8300 and S87XX Servers.

Local Node Number

Enter a number to identify a specific node in a server network. This entry must match the DCS switch node number and the CDR node number if they are specified.

Valid entries	Usage
1 to **63**	Enter the number of a specific node in a network.
blank	Can be blank if automatic restoration, DCS, and CDR are not used.

Retry ARS Analysis if All-Location Entry Inaccessible

This field appears on the Dial Plan Parameters screen in Communication Manager Release 4.0.x or later.

Valid entries	Usage
y	The system finds and uses the best possible entry in the per-location ARS table, if the all-location table points to a trunk group that cannot be accessed because the network has fragmented.
n	The system does not retry ARS analysis when a trunk group cannot be accessed because the network has fragmented.

UDP Extension Search Order

Appears if, on the System Parameters Customer-Options (Optional Features) screen, the **Uniform Dialing Plan** field is **y**. Specifies the first table to search to match a dialed extension. If the dialed extension is not found in the specified place, it then is searched for in the "other" place.

Valid entries	Usage
local-extensions -first	Searches the local server first to match a dialed extension; if not found, then uses the UDP tables to route the call.
udp-table-first	Searches the UDP tables for an off-switch conversion; if not found, then searches the local server for the dialed extension.

EXTENSION DISPLAY FORMATS

Extension Display Format (6-13 digits)

Use these fields to specify how the system punctuates extensions for display. The punctuation field is divided into two columns, one for **Inter-Location/SAT** displays, and one for **Intra-Location** displays. Blank spaces are sometimes used in telephone extensions, especially outside of the U.S. Dots (**.**) are used on SAT screens in place of blanks. The following table gives the maximum number of punctuation marks permitted for each extension length:

Note:
The number of punctuation marks that the system allows is determined by the number of “x”s in the format:

- If the format contains fewer than 6 x's, no punctuation marks can be entered.
- For 6 or more x's, the maximum number of punctuation marks is determined by the following table.

Extension Length	Max Punctuation Marks	Max Total Length
6	2	8
7	1	8
8	3	11
9	3	12
10	3	13
11	2	13
12	1	13
13	0	13

Valid entries	Usage
xx.xx.xx	This field can contain all ‘x’ characters (no punctuation) or you can use a combination of ‘x’ characters and 0 to 2 hyphens (-), spaces, or periods (.) to depict how extensions display. If the format contains fewer than 6 x's, no punctuation marks can be entered. You must specify a format or accept the default. You cannot leave this field blank. The default values for the 8-, 9-, 10-, 11-, 12-, and 13-digit fields are those shown in Figure 66.

Digit Absorption

This screen implements up to 5 digit absorption lists. The screen might be required for each CO and FX trunk group connected to a step-by-step CO. Each outgoing digit string from the server running Communication Manager to the step-by-step CO is treated according to entries in the "Absorption Treatment Assignment" section of the screen.

Note:
If the **Digits** field on the Trunk Group screen is blank, you cannot administer Digit Absorption.

Field descriptions for page 1

Figure 67: Digit Absorption screen

```
change digit absorption                                              Page 1 of x
                              DIGIT ABSORPTION

                           List Number: __
  ABSORPTION TREATMENT INFORMATION (All selections must be from same group)
                Choice      Meaning
    Group I.      A       Digit not absorbed.
                  B       Digit absorbed repeatedly.
                  C       Digit absorbed once with no further absorption.
    Group II.     A       Digit not absorbed.
                  D       Digit absorbed only if it is the first digit.
                  E       Digit absorbed only if it is the second digit and
                          the first digit was already absorbed.
                  F       Digit asorbed only if it is the first or second digit.

  ABSORPTION TREATMENT ASSIGNMENT (select treatment (A-F) for each digit below)
          0: A          2: A         4: A         6: A         8: A
          1: A          3: A         5: A         7: A         9: A
```

Absorption Treatment Assignment

Valid entries	Usage
A to **F**	Enter a desired treatment letter. All choices for the digits 0 through 9 must be taken from the same group (Group I or Group II).

Absorption Treatment Information

This is a display-only section. It shows how Digit Absorption treats each digit, 0 through 9, depending on the assignment of A through C for Group I, and A, D, E, and F for Group II. Enter the assignment on the next section on the screen.

List Number

A display-only field indicating the Digit Absorption List number (**0** to **4**). The list number is referenced from a field entry on the associated trunk group.

Display Parameters

Use this screen to establish how extensions of 6 to 13 digits are punctuated. There are 26 possible Display Parameters screens, numbered from 1-25. This screen is linked to the value that is entered in the **Display Parameters** (**Disp Parm**) field on the **Locations** screen.

Field descriptions for page 1

Figure 68: Display Parameters screen

```
 change display-parameters 5                                      Page 1 of 1
                              DISPLAY PARAMETERS

EXTENSION DISPLAY FORMATS

   Note: If a format is blank, the corresponding format administered
         on the Dial Plan Parameters form will be used

                              Inter-Location      Intra-Location
        6-Digit Extension:    ________            ________
        7-Digit Extension:    ________            ________
        8-Digit Extension:    ___________         ___________
        9-Digit Extension:    ____________        ____________
       10-Digit Extension:    _____________       _____________
       11-Digit Extension:    _____________       _____________
       12-Digit Extension:    _____________       _____________
       13-Digit Extension:    _____________       _____________

       Default Call Appearance Display Format: inter-location
```

EXTENSION DISPLAY FORMATS

The fields in this section of the Display Parameters screen override similar fields on the **Dial Plan Parameters** screen. If you leave these fields on the Display Parameters screen blank, the values on the Dial Plan Parameters screen apply.

Extension Display Format (6-13 digits)

Use these fields to specify how the system punctuates extensions for display. The punctuation field is divided into two columns, one for **Inter-Location** displays, and one for **Intra-Location** displays. Blank spaces are sometimes used in telephone extensions, especially outside of the U.S. Dots (**.**) are used on SAT screens in place of blanks.

Note that the number of punctuation marks that the system allows is determined by the number of “x”s in the format:

- If the format contains fewer than 6 x's, no punctuation marks can be entered.
- For 6 or more x's, the maximum number of punctuation marks is determined by the following table.

The following table gives the maximum number of punctuation marks permitted for each extension length:

Extension Length	Max Punctuation Marks	Max Total Length
6	2	8
7	1	8
8	3	11
9	3	12
10	3	13
11	2	13
12	1	13
13	0	13

Valid entries	Usage
xx.xx.xx or blank	This field can contain all ‘x’ characters (no punctuation) or you can use a combination of ‘x’ characters and 0 to 2 hyphens (-), spaces, or periods (.) to depict how extensions display. If the format contains fewer than 6 x's, no punctuation marks can be entered. The default is blank.

Default Call Appearance Display Format

This field only affects call appearances on telephones that support downloadable call appearance buttons, such as the 2420 and 4620 telephones. Bridged call appearances are not affected by this field.

Valid entries	Usage
inter-location	The system displays the complete extension on downloadable call appearance buttons. This is the default.
intra-location	The system displays a shortened version of the extension on downloadable call appearance buttons.

Inter-Location

Use this field to specify punctuation for calls between locations. This is the default.

Intra-Location

Use this field to specify punctuation for calls within a location.

Valid entries	Usage
y	Enter **y** when the **Signaling Mode** field is **CAS** and the DS1 link is providing E-1 service.
n	Enter **n** for all other applications.

DS1 Circuit Pack

Use this screen to administer all DS1 circuit packs.

Field descriptions for page 1

Figure 69: DS1 Circuit Pack screen

```
 add ds1 nnnn                                                  Page 1 of x
                                DS1 CIRCUIT PACK

             Location: _____                       Name: _______________
             Bit Rate: _____                Line Coding: ____
    Line Compensation: _                   Framing Mode: ___
       Signaling Mode: ___________            D-Channel: __
              Connect: _________              Interface: ___________
         Interconnect: ___                Peer Protocol: ________
                                       Country Protocol: ____
                                       Protocol Version: _
 Interface Companding: _____                       CRC? _
            Idle Code: ________
                            DCP/Analog Bearer Capability: ________
                                       T303  Timer(sec):
                                      Disable Restarts?:

    MMI Cabling Board: _____  MMI Interface: ESM

 MAINTENANCE PARAMETERS

       Slip Detection? _                  Near-end CSU Type: ________
                                   Block Progress Indicator? n
```

Figure 70: DS1 Circuit Pack screen for Croatia and South Africa

```
add ds1 nnnn                                                  Page 1 of x
                                DS1 CIRCUIT PACK

             Location: ______                         Name: ____________
             Bit Rate: ______                  Line Coding: ____

       Signaling Mode: __________

         Interconnect: ____                Country Protocol: ____

 Interface Companding: ____
            Idle Code: ____

               Received Digital Metering Pulse Minimum (ms):
               Received Digital Metering Pulse Maximum (ms):
                      Received Digital Metering Pulse Value:
       Slip Detection: __                 Near-end CSU Type: ______
                                        Block Progress Indicator? n
```

The following screen is valid *only* for the TN2242.

Figure 71: DS1 Circuit Pack screen for Channel Associated Signaling

```
add ds1 nnnn                                                  Page 1 of x
                                DS1 CIRCUIT PACK

              Location: 01A13                          Name: _______________
              Bit Rate: 2.048                   Line Coding: cmi

        Signaling Mode: CAS
          Interconnect: pbx
                                            Country Protocol: 3

  Interface Companding: mulaw
             Idle Code: 11111111



 MAINTENANCE PARAMETERS

        Slip Detection? n
```

Bit Rate

Use this field to select the maximum transmission rate for DS1 circuit packs that support either T-1 or E-1 service. For circuit packs that only support one of these services, the field is a display-only field.

Note:
Once an `add ds1` operation is complete (that is, the DS1 screen has been submitted) you can't change the **Bit Rate** field with a `change ds1` command. Instead, execute a `remove ds1` command. Then use the `add ds1` command to administer the circuit pack again. You'll have to re-enter all the information for the circuit pack.

TN464C (and later release) circuit packs have an option switch that must be set to match the entry in the **Bit Rate** field.

Valid entries	Usage
1.544	Use for T-1 service.
2.048	Use for E-1 service.

Channel Numbering

The ETSI and ISO QSIG specifications require that B-channels on an E1 be encoded as 1 to 30 in the Channel ID IE. Prior to the existence of this field, Communication Manager only used this scheme for Country Protocols 2a (Australia) and 13a (Germany 1TR6). This field appears when the **Signaling Mode** field is **isdn-pri**, the **Bit Rate** field is **2.048**, the **Connect** field is **pbx**, and the **Interface** field is **peer-master** or **peer-slave**.

Valid entries	Usage
timeslot	
sequential	If Communication Manager is connected via QSIG trunks to a switch/ server supporting the ETSI QSIG or ISO QSIG specifications, this field must be **sequential**. When the **Signaling Mode** field is **isdn-pri** and the **Bit Rate** field is **2.048**, but the **Channel Numbering** field does not display because of the setting of other fields, it is set internally to **sequential** for 2a (Australia) and 13a (Germany).

Connect

In order to control communications at layers 2 and 3 of the ISDN-PRI protocol, use this field to specify what is on the far end of this DS1 link. This field only appears when the **Signaling Mode** field is **isdn-pri**.

Valid entries	Usage
pbx	Enter **pbx** if this DS1 link is connected to another switch in a private network. If **pbx** is entered, the **Interface** field appears.
line-side	Enter **line-side** when Communication Manager is acting as the network side of an ISDN-PRI interface. Use **line-side** to connect to Roll About Video equipment.
network	Enter **network** when the DS1 link connects Communication Manager to a central office or any other public network switch.
host	Enter **host** when the DS1 link connects Communication Manager to a computer.

Country Protocol

The entry in this field must match the country protocol used by the far-end server. For connections to a public network, your network service provider can tell you which country protocol they are using.

This field appears if the **Signaling Mode** field is **CAS** or **isdn-pri**. For the Japanese 2Mbit trunk circuit pack, this is a display-only field if the **Signaling Mode** field is **CAS**.

Note:
For a list of country codes, see the Country code table on page 886.

Valid entries	Usage
1 to **25**	Enter the country protocol used by the central office at which this link terminates.
etsi	Enter **etsi** if your network service provider uses the protocol of the European Telecommunications Standards Institute (ETSI). Enter **etsi** only if the **Signaling Mode** field is **isdn-pri**.

CRC

This field indicates whether a cyclic redundancy check (CRC) is performed on transmissions that the DS1 circuit pack receives. This field does not display for all circuit packs.

Valid entries	Usage
y	Enter **y** when the **Signaling Mode** field is **CAS** and the DS1 link is providing E-1 service.
n	Enter **n** for all other applications.

D-Channel

The Japanese 2Mbit trunk circuit pack, when administered to support ISDN-PRI signaling, allows you to assign the D-channel to any channel from 1 to 31 in an E-1 facility. You cannot submit the screen if this field is blank. Using the `change ds1` command, you can change this field if the D-channel is not used in a signaling group. This field appears only when the **Location** field indicates the circuit pack is a Japanese 2Mbit trunk circuit pack and the **Signaling Mode** field is **isdn-pri**.

Valid entries	Usage
1 to **31**	Enter the number of the channel that is used as the D-channel.

DCP/ANALOG Bearer Capability

This field appears when the **Signaling Mode** field is **isdn-pri**. This field sets the information transfer capability in a bearer capability IE of a setup message to **speech** or **3.1kHz**.

Valid entries	Usage
3.1kHz	Provides 3.1kHz audio encoding in the information transfer capability.
speech	Provides speech encoding in the information transfer capability.

Disable Restarts

Use this field to control whether outgoing RESTART messages are sent. This field appears when one of the following is true:

- **Country Protocol** is **3 (Japan)**
- **Country Protocol** is **ETSI**

- **Peer Protocol** is **QSIG**

This field and the **Protocol Version** field are mutually exclusive. Only one of the fields can be displayed. You can also use this field to disable QSIG restarts.

Valid entries	Usage
y	Outgoing restarts are disabled, that is, RESTART messages are not sent.
n	Outgoing RESTART messages are sent. This is the default.

DMI-BOS

The DMI/BOS protocol is used for high-speed digital communications between a host computer and Communication Manager. With this 24-channel protocol, channels 1 to 23 of the DS1 link carry data and channel 24 carries control signaling. DMI/BOS has greater capacity than a robbed-bit 24-channel facility. This field appears only when the **Signaling Mode** field is **common-chan**.

Valid entries	Usage
y	Enter **y** to activate the Digital Multiplexed Interface-Bit Oriented Signaling (DMI-BOS) format.
n	Enter **n** to use an Avaya proprietary format.

Framing Mode

Use this field to select either superframe (sf or d4) or extended superframe (esf) for T1 service on the DS1 link. The framing mode you use must match the mode used on the other end of the link, so work with your network services provider to determine the appropriate entry for this field.

This field only appears if the **Bit Rate** field is **1.544** (that is, if you're using T-1 service). If you're using E-1 service, Communication Manager automatically selects CEPT1 framing.

Tip:
Avaya recommends using ESF when your service provider supports it, especially if you might someday upgrade the facility to ISDN. The ESF format provides enhanced performance measurements and uses a sophisticated error-checking method to ensure data integrity.

Valid entries	Usage
d4	Enter **d4** to use the basic DS1 superframe (sf). Avaya recommends this mode only for voice traffic.
esf	Enter **esf** to use the Extended Superframe format. Avaya recommends this mode for digital data traffic. If you enter **esf** for a TN464F, TN767E, or a later suffix DS1 circuit pack, a second page of the DS1 Circuit Pack screen becomes available to administer ESF Data Link options.

Idle Code

CAUTION:
Customers: The entry in the **Country Protocol** field sets the default idle code. Do not change the default without assistance from Avaya or your network services provider.

For some circuit packs, this is a display-only field.

Valid entries	Usage
any 8-digit string of 0’s and 1’s	This entry sets the signal sent out over idle DS0 channels. The string must be compatible with the protocol used by the far-end switch/server.

Interconnect

For E-1 service using channel-associated signaling, the entry in this field tells Communication Manager whether the DS1 circuit pack is using a public or private network protocol. The entry in this field must agree with the entry in the **Group Type** field on the Trunk Group screen. This field appears only when the **Signaling Mode** field is **CAS**.

Valid entries	Usage
pbx	If **pbx** is selected, the board operates as a tie trunk circuit pack.
CO	If **CO** is selected, the board operates as a CO or DID circuit pack. Use **CO** for Enterprise Mobility User (EMU)/EC500 administration.

Interface

This field only appears when the **Connect** field is **pbx**; that is, when this DS1 link is providing an ISDN-PRI connection in a private network. The **Interface** field controls how your server negotiates glare with the far-end switch. The servers at either end of the DS1 link must have complementary settings in this field: if not, the D-channel won't even come up. For example, if the Avaya S8XXX Server at one end of the link is administered as **network**, the other end must be administered as **user**.

Valid entries	Usage
Use the following 2 values for private network applications in the U.S.	
network	Enter **network** if your server overrides the other end when glare occurs. If you are connecting your server to a host computer, set this field to **network**.
user	Enter **user** if your server releases the contested circuit and looks for another when glare occurs. If you are connecting your server to a public network, set this field to **user**.
Use the following values for private networks (including QSIG networks) outside the U.S. Entering either of these values causes the **Peer Protocol and Side** fields to appear.	
peer-master	Enter **peer-master** if your switch overrides the other end when glare occurs.
peer-slave	Enter **peer-slave** if your switch releases the contested circuit and looks for another when glare occurs.

Interface Companding

The entry in this field must match the companding method used by the far-end switch. This field does not appear for all DS1 circuit packs.

Valid entries	Usage
alaw	Enter **alaw** for E-1 service.
mulaw	Enter **mulaw** for T-1 service.

Interworking Message

This field determines what message the server sends when an incoming ISDN trunk call interworks (is routed over a non-ISDN trunk group).

Valid entries	Usage
PROGress	Normally select this value. PROGress asks the public network to cut through the B-channel and let the caller hear tones such as ringback or busy tone provided over the non-ISDN trunk.
ALERTing	ALERTing causes the public network in many countries to play ringback tone to the caller. Select this value only if the DS1 is connected to the public network, and it is determined that callers hear silence (rather than ringback or busy tone) when a call incoming over the DS1 interworks to a non-ISDN trunk.

ITN-C7 Long Timers

This field controls the T302 and T303 timers. It only appears if the **Signaling Mode** field is **isdn-pri**.

Valid entries	Usage
y	Use if you want to increase the length of the long timers.
n	Leave n if you want to use the default long timers.

Line Coding

This field selects the type of line coding used on this facility. The setting in this field must match the setting on the far-end of the link, or you must have an intervening CSU to convert the line coding protocols. Voice calls work even if line coding does not match, but a single data call brings down the DS1 facility. For the TTC 2Mb CMI Trunk circuit pack, this is a display-only field showing **cmi** (coded mark inversion).

The following information is for reference. Talk with your network service provider or your Avaya technical support representative to find the appropriate protocol for your application.

CAUTION:
If you change this field, you must busy out the DS1 circuit pack. You must also change the following screens: Route Pattern*,* Access Endpoint*,* PRI Endpoint*,* Signaling Group*, and* Trunk Group*.*

Note:
When the DS1 circuit pack is used for ISDN service, the ISDN D-channel data is inverted when **ami-basic** or **ami-zcs** is entered and not inverted when **b8zs** or **hdb3** is entered.

Valid entries	Usage
b8zs (bipolar eight zero substitution)	Enter **b8zs** for T-1 facilities that support voice and/or data traffic. Enter **b8zs** if you need a 64K clear channel.
ami-zcs (alternate mark inversion - zero code suppression)	Enter **ami-zcs** only for T-1 facilities that carry voice traffic: Avaya does not recommend this for digital-data applications. If you anticipate upgrading this facility to ISDN, use b8zs line coding if possible.
ami-basic (alternate mark inversion-basic)	Enter **ami-basic** for unrestricted E-1 facilities.
hdb3 (high density bipolar 3)	Enter **hdb3** for restricted E-1 facilities.
cmi (coded mark inversion)	Used in Japan, **cmi** is the only type of line coding you can use with the Japanese 2Mbit trunk circuit pack. This field becomes a display-only field when you are administering the Japanese 2Mbit trunk circuit pack.

Line Compensation

The appropriate entry in this field varies with the type of cable used, so work with your network service provider to determine the correct setting in your situation. The following table shows the appropriate entries for different lengths of 22-gauge ABAM cable terminated on a DSX-1 cross-connect.

Valid entries	Usage
1	Length: 000 to 133 (ft), 000 to 40.5 (m)
2	Length: 133 to 266 (ft), 40.5 to 81.0 (m)
3	Length: 266 to 399 (ft), 81.0 to 122 (m)

Valid entries	Usage
4	Length: 399 to 533 (ft), 122 to 163 (m)
5	Length: 533 to 655 (ft), 163 to 200 (m)

The following table shows the appropriate entries for different lengths of 22-gauge ABAM cable directly connecting to DS1 interfaces.

Valid entries	Usage
1	Length: 0000 to 0266(ft), 000 to 081(m)
2	Length: 0266 to 0532(ft), 081 to 162(m)
3	Length: 0532 to 0798(ft), 162 to 243(m)
4	Length: 0798 to 1066(ft), 243 to 325(m)
5	Length: 1066 to 1310(ft), 325 to 400(m)

Location

This display-only field shows the port address specified in the `add` command when the circuit pack was first administered.

MMI Cabling Board

This field appears only if the **MMCH** field is **y** on the System Parameters Customer-Options (Optional Features) screen.

Valid entries	Usage
slot address (cabinet, carrier, slot)	Enter the slot location (cabinet, carrier, slot) of the multimedia interface circuit pack that is connected to the Expansion Services Module (ESM).

MMI Interface

This display-only field appears if the **MMCH** field is **y** on the System Parameters Customer-Options (Optional Features) screen and there is a value in the **MMI Cabling Board** field.

Name

Use this field to assign a significant, descriptive name to the DS1 link. Avaya recommends putting the vendor's circuit ID for the link in this field, because that information helps you troubleshoot problems with the link, but you could also use this field to indicate the function or the destination of this DS1 facility. In that case, put the DS1 link circuit ID in the **Name** field of the trunk group associated with this link.

Valid entries	Usage
1 to 15 characters	Enter a name for the DS1 link. **NOTE:** Avaya BRI stations support ASCII characters only. They do not support non-ASCII characters, such as Eurofont or Kanafont. Therefore, if you use non-ASCII characters in any Communication Manager **Name** field, such characters do not display correctly on a BRI station.

Peer Protocol

This allows you to administer the peer level protocol that operates in a private network. This field appears if the **Interface** field is **peer-master** or **peer-slave**. To enter **Q-SIG**, the **Basic Call Setup** field on the System Parameters Customer-Options (Optional Features) screen must be **y**

.

Valid entries	Usage
Q-SIG	This implements QSIG Network Basic Call.
TTC	For private networking. Requires a Digital Trunk (Japan 2 MB TTC) (TN2242) circuit pack.

Protocol Version

In countries whose public networks allow multiple layer-3 signaling protocols for ISDN-PRI service, this field selects the protocol that matches your network service provider's protocol. See Public network signaling administration for ISDN-PRI Layer 3 on page 236 to see which countries support which protocols.

This field appears only when:

- The **Signaling Mode** field is **isdn-pri** and the **Connect** field is **network.**

- The **Signaling Mode** field is **isdn-pri,** the **Connect** field is **pbx**, and the **Interface** field is **user** or **network**.

Valid entries	Usage
a, b, c, d	The entry in this field must match the protocol used by your network service provider, so work with your vendor to determine the appropriate entry.

WARNING:
The AT&T Switched Network Protocol does not support restricted displays of connected numbers. Therefore, if you administer the 1a country-protocol/ protocol-version combination on the DS1 screen, you cannot set the **Send Connected Number** field to r (restricted) on the ISDN-PRI Trunk Group screen, as this causes display problems.

Public network signaling administration for ISDN-PRI Layer 3

The table below shows Communication Manager public-network access connections for ISDN-PRI Layer 3.

Admin value	Country	Protocol supported	B-channel mtce msg
1-a	United States, Canada	AT&T TR 41449/ 41459 (tested with AT&T network, Canadian network, and MCI network)	Service
1-b	United States	Bellcore TR 1268; NIUF.302; ANSI T1.607	Restart
1-c	United States	NORTEL DMS-250 BCS36/IEC01	Service
1-d	United States	Telecordia SR-4287	Service
2-a	Australia	AUSTEL TS014.1; Telecom Australia TPH 1856 National ISDN protocol	Restart
2-b	Australia	ETSI ISDN protocol	Restart
3	Japan	NTT INS-NET	Restart
4	Italy	ETS 300 102	Restart
5	Netherlands	ETS 300 102	Restart
6	Singapore	ETS 300 102	Restart
7	Mexico	ETS 300 102	Restart
			1 of 2

Admin value	Country	Protocol supported	B-channel mtce msg
8	Belgium	ETS 300 102	Restart
9	Saudi Arabia	ETS 300 102	Restart
10-a	United Kingdom	ETS 300 102 (for connection to DASS II/ DPNSS through external converter)	Restart
10-b	United Kingdom, Ireland	ETS 300 102 (Mercury); British Telecom ISDN 30; Telecom Eireann SWD 109	None
11	Spain	Telefonica ISDN Specification	Restart
12-a	France	VN4 (French National PRI)	None
12-b	France	ETS 300 102 modified according to P10-20, called Euronumeris	None
13-a	Germany	FTZ 1 TR 6 (German National PRI)	None
13-b	Germany	ETS 300 102	Restart
14	Czech Republic, Slovakia	ETS 300 102	Restart
15	Russia (CIS)	ETS 300 102	Restart
16	Argentina	ETS 300 102	Restart
17	Greece	ETS 300 102	Restart
18	China	ETS 300 102	Restart
19	Hong Kong	ETS 300 102	Restart
20	Thailand	ETS 300 102	Restart
21	Macedonia	ETS 300 102	Restart
22	Poland	ETS 300 102	Restart
23	Brazil	ETS 300 102	Restart
24	Nordic	ETS 300 102	Restart
25	South Africa	ETS 300 102	Restart
ETSI-a	Europe, New Zealand, etc.	ETS 300 102	Restart
ETSI-b		ETS 300 102	None
			2 of 2

Received Digital Metering Pulse Maximum (ms)

This field appears only when the **Signaling Mode** field is **cas** (Channel Associated Signaling), the **Interconnect** field is **co** or **pbx**, and the **Country Protocol** field is administered for a protocol that uses periodic pulse metering (PPM) as defined in Incoming digital PPM signaling default per Country Protocol code on page 239. The default value depends on the **Country Protocol f**ield's entry. For a list of country codes, see the Country code table on page 886.

Valid entries	Usage
20 to **1000** ms in increments of 10ms.	Work with your network services provider to determine the appropriate entry. The entry must be greater than the **Received Digital Metering Pulse Minimum** field.

Received Digital Metering Pulse Minimum (ms)

This field appears only when the **Signaling Mode** field is **cas** (Channel Associated Signaling), the **Interconnect** field is **co** or **pbx**, and the **Country Protocol** field is administered for a protocol that uses periodic pulse metering (PPM) as defined in Incoming digital PPM signaling default per Country Protocol code on page 239. The default value depends on the **Country Protocol** field's entry. For a list of country codes, see the Country code table on page 886.

Valid entries	Usage
20 to **1000** ms in increments of 10ms.	Work with your network services provider to determine the appropriate entry. The entry must be less than the **Received Digital Metering Pulse Maximum** field.

Received Digital Metering Pulse Value

This field appears when the **Signaling Mode** field is **cas** (Channel Associated Signaling), the **Country Protocol** field is **21**, and the **Interconnect** field is **co** or **pbx**.

Valid entries	Usage
0, **1**	Work with your network services provider to determine the appropriate entry.

Table 5: Incoming digital PPM signaling default per Country Protocol code

Code	Country	PPM Min (ms)	PPM Max (ms)	PPM Value
0	null	NA	NA	NA
1	U.S.	NA	NA	NA
2	Australia	80	180	0
3	Japan	NA	NA	NA
4	Italy	120	150	1
5	Netherlands	90	160	0
6	Singapore	NA	NA	NA
7	Mexico	20	180	1
8	Belgium	20	180	1
9	Saudi Arabia	NA	NA	NA
10	UK	NA	NA	NA
11	Spain	20	220	0
12	France	NA	NA	NA
13	Germany	NA	NA	NA
14	Czech Republic	20	420	1
15	Russia CIS	NA	NA	NA
16	Argentina	10	180	1
17	Greece	100	180	1
18	China	NA	NA	NA
				1 of 2

Table 5: Incoming digital PPM signaling default per Country Protocol code (continued)

Code	Country	PPM Min (ms)	PPM Max (ms)	PPM Value
19	Hong Kong	NA	NA	NA
20	Thailand	20	180	1
21	Macedonia Croatia	120 20	180 80	1 1
22	Poland	100	150	0
23	Brazil	NA	NA	NA
24	Nordic	NA	NA	NA
25	South Africa	160	240	0, 1
				2 of 2

Side

This field controls how your server running Communication Manager resolves glare at layer 3 over an ISDN-PRI link in QSIG private networks. It appears if the **Interface** field is **peer-master** or **peer-slave**.

The default value of the field changes depending upon which value the **Interface** field contains.

CAUTION:
It is critical that administration on this server correctly pairs with the administration of the far-end switch/server. If the far-end is administered as the b side, this field should be set to a regardless of whether the layer 2 designation is peer-master or peer-slave, and vice versa.

Valid entries	Usage
a	Enter **a** if the **Interface** field is **peer-master** (this server overrides the far-end when glare occurs).
b	Enter **b** if the **Interface** field is **peer-slave** (this server releases the contested circuit and looks for another when glare occurs).

Signaling Mode

This field selects the signaling method used for the DS1 link. This mode must match the method used on the other end of the link, so work with your network services provider to determine the appropriate entry for this field.

Valid entries	Usage
CAS (Channel Associated Signaling)	Enter **CAS** for out-of band signaling with E-1 service. This setting yields 30 64-kbps B-channels for voice or data transmission. Channel 0 is used for framing while channel 16 carries signaling. Enter CAS for Enterprise Mobility User (EMU)/EC500 administration.
robbed-bit	Enter **robbed-bit** for in-band signaling with T-1 service. This setting yields 24 56-kbps B-channels for voice transmission.
isdn-pri	Enter **isdn-pri** for either T-1 or E-1 ISDN service. This setting supports both Facility Associated Signaling and Non-Facility Associated Signaling.
isdn-ext	Enter **isdn-ext** for either T-1 or E-1 ISDN service. This setting supports only Non-Facility Associated Signaling. **Note**: NFAS is primarily a feature for ISDN-T1 connections offered by service providers in North America and Hong Kong. However, it can also be used on private-network connections, and in that context it is possible to set up NFAS using ISDN-E1 interfaces.
common-chan	Enter **common-chan**, for out-of-band signaling with T-1 service. This setting yields 23 64-kbps B-channels for voice or data transmission. Channel 24 is used for signaling.

T303 Timer (sec)

Use this field to enter the number of seconds the system waits for a response from the far end before invoking Look Ahead Routing. Appears when the **Group Type** field is **isdn-pri**.

Valid entries	Usage
2 to **10**	Enter **a** if the **Interface** field is **peer-master** (this server overrides the far-end when glare occurs).

MAINTENANCE PARAMETERS

Alarm When PRI Endpoint Detached

Use this field for DS1 circuit packs connected to Roll-About Video equipment. This field appears only when the **Connect** field is **line-side**.

Valid entries	Usage
y/n	Enter **y** if you want the server running Communication Manager to generate an alarm when the DS1 board detects a loss of signal (for example, if the video equipment is disconnected).

EC Configuration

Appears when **Echo Cancellation** is **y** on the DS1 Circuit Pack screen.

Valid entries	Usage
1 to **15**	Enter **a** if the **Interface** field is **peer-master** (this server overrides the far-end when glare occurs).

EC Direction

Direction of echo cancellation. Appears when **Echo Cancellation** is **y** on the DS1 Circuit Pack screen.

Valid entries	Usage
inward/outward	Enter **a** if the **Interface** field is **peer-master** (this server overrides the far-end when glare occurs).

Echo Cancellation

Appears when **DS1 Echo Cancellation** is **y** on the System Parameters Customer-Options (Optional Features) screen and circuit packs support echo cancellation.

Valid entries	Usage
y/n	Enter **y** to allow echo cancellation.

Near-end CSU Type

This field appears only when the DS1 circuit pack is a TN767D or TN464E or later suffix model, the **Bit Rate** field is **1.544**, and the **Country Protocol** field is **1** (U.S.) or **3** (Japan). This field does not display for all circuit packs.

Valid entries	Usage
other	Enter **other** if no channel service unit is attached to the DS1 facility or if the CSU is an external unit. No options are available on page 2 for administering an external CSU.
integrated	Enter **integrated** if a 120A CSU module is attached to the DS1 board. This integrated channel service unit (ICSU) can accept software-administrable option downlinks (that is, it can respond to test codes from technician's equipment and report its status). When you enter **integrated**, fields for administering options on the ICSU appear on page 2 of the DS1 Circuit Pack screen.

Slip Detection

Slips — synchronization errors — slow digital transmissions and can result in data loss. The server maintains a slip-count record for each DS1 interface to detect errors and evaluate their severity (the type of alarm). If as many as 50 percent of those spans administered for slip detection are experiencing slips (with respect to the primary), then a decision is made to switch to the secondary.

CAUTION:
Always enter **y** for DS1 circuit packs that serve as primary or secondary synchronization references.

Valid entries	Usage
y	Enter **y** to allow maintenance software to measure the slip-rate of this circuit pack and determine whether it's excessive. Typically, enter **y** for DS1 spans used for data applications and for spans used as synchronization references. This excludes all T1-spans connecting channel banks, unless the channel bank is externally timed. This entry enables switching between the primary and secondary synchronization references and an internal high-accuracy clock.
n	Enter **n** for DMI-BOS links or when testing is not required. Normally, enter **n** for DS1 spans that are used exclusively for voice and that do not serve as the primary or secondary synchronization source.

Block Progress Indicator

This field allows you to block sending of the progress indicator in the SETUP message. It appears when the **Country Protocol** field is set to 1 and the **Protocol Version** field is set to b.

Valid entries	Usage
y	Enter **y** to prevent the progress indicator from being sent in the SETUP message.
n	Enter **n** to allow the progress indicator to be sent.

Field descriptions for page 2

Figure 72: DS1 Circuit Pack screen

```
add ds1 nnnn                                                     Page 2 of x
                               DS1 CIRCUIT PACK

ESF DATA LINK OPTIONS

                      Network Management Protocol:
  Send ANSI-T1.403 One-Second Performance Reports?
                              Far-end CSU Address:


INTEGRATED CSU OPTIONS
                                     Transmit LBO:
                                     Receive ALBO:
                                     Upon DTE LOS:

CPE LOOPBACK JACK OPTIONS
                   Supply CPE Loopback Jack Power?
```

CAUTION:
Customers: Do not change fields on this page without assistance from Avaya or your network service provider.

Page 2 does not appear for all DS1 circuit packs. For those circuit packs that support it, this page appears only when the **Framing Mode** field is **esf** or the **Near-end CSU Type** field is **integrated**.

ESF DATA LINK OPTIONS

Far-end CSU Address

This field, which, appears only if the **Framing Mode** field is **esf**.

Valid entries	Usage
a b	Enter **b**. This field administers the transmit direction address used for the `ESF data link` command with both integrated and external channel service units (CSU).

Network Management Protocol

This field appears only if the **Framing Mode** field is **esf**.

Valid entries	Usage
tabs	The entry in this field, used only with circuit packs that have an integrated channel service unit (CSU), allows the data link to be remotely monitored.

Send ANSI-T1.403 One-Second Performance Reports

This field selects whether your DS1 circuit pack sends error reports to the far-end server/switch. These reports are useful for network management, and are sent at 1-second intervals when enabled. This field appears only if the **Framing Mode** field is **esf**. It is used only with circuit packs that have an integrated channel service unit (CSU).

Valid entries	Usage
y/n	Enter **n**. Consult your Avaya technical support representative if you think you might want to use these reports.

INTEGRATED CSU OPTIONS

Receive ALBO (Receive Automatic Line Build-Out)

This field increases the strength of incoming signals by a fixed amount to compensate for line losses.

Valid entries	Usage
26db	To set this field correctly, measure the signal loss on this specific facility. However, you can enter **26db** for most applications. **36db** is occasionally appropriate, mainly on campus networks that don't conform to public telephone network standards.
36db	

Transmit LBO (Transmit Line Build-Out)

This field reduces the outgoing signal strength by a fixed amount. The appropriate level of loss depends on the distance between your Communication Manager server (measured by cable length from the smart jack) and the nearest repeater. Where another server/switch is at the end of the circuit, as in campus environments, use the cable length between the 2 switches to select the appropriate setting from the table below. This field appears if the **Near-end CSU Type** field is **integrated**.

Valid entries	Usage
0db	For distances of 2,001 to 3,000 feet
-7.5db	For distances of 1,001 to 2,000
-15db	For distances of 0 to 1,000 feet
-22.5db	For mid-span repeaters

Upon DTE LOS

DTE stands for "Data Terminal Equipment." This field tells Communication Manager what to do if the outgoing signal from the DS1 circuit pack (the data terminal equipment) to the network is lost.

Valid entries	Usage
loopback	Enter **loopback** to return the network signal to the network. This prevents any alarms from being generated at the far-end
ais	Enter **ais** (Alarm Indicator Signal) to send an unframed all-ones signal (the AIS or Blue Alarm) to the far-end server/switch. This option alerts your network service provider to the problem immediately and aids in troubleshooting.

CPE LOOPBACK JACK OPTIONS

Supply CPE Loopback Jack Power

If a CPE (Customer Premise Equipment) Loopback Jack is installed, the DS1 board should supply power to the equipment during loopback testing.

Valid entries	Usage
y/n	Enter **y** if a CPE Loopback Jack is installed. If not, you must enter **n**.

Related topics

See DS1 Trunk Service in *Avaya Aura™ Communication Manager Feature Description and Implementation,* 555-245-205, for more information.

Duplicate Station

Use this screen to add telephones is to copy the information from an existing telephone and modify it for each new telephone. For example, you can configure one telephone as a template for an entire work group. Then, you merely duplicate the template Station screen to add all the other extensions in the group. Note that only telephones of the same model can be duplicated. The duplicate command copies all the feature settings from the template telephone to the new telephones.

Note:
For field descriptions of specific fields, see the **Station** screen.

Figure 73: Duplicate Station screen - page 1

```
duplicate station 1234567890123                                      Page 1 of 2
                                     STATION

                                                    Security            Endpt
Extension        Port    Name                       Code                ID
1234567890123    1234567 123456789012345678901234567 12345678            12
```

Figure 74: Duplicate Station screen - page 2

```
duplicate station 1234567890123                                      Page 2 of 2
                                     STATION

Extension        Room        Jack   Cable
1234567890123    1234567890 12345 12345
```

Duplicate Vector

Use this screen to define the vector numbers and names for the vectors to be duplicated from the master vector and to display one VDN extension that the vector is assign to. An asterisk (*) appears if the vector is assigned to two or more VDNs.

Upon submission of this screen, copies of the master vector are created, numbered and named as specified on the screen, with all steps populated exactly the same as the original. After the vector duplicates are created, you can use the **`change vector`** command to add to or otherwise edit the vector(s), including changing the type vector fields as required. *Goto* references, particularly, should be reviewed for appropriateness in the copies.

Figure 75: Duplicate Vector screen - page 1

```
duplicate vector 1                                                  Page 1 of x
                             DUPLICATE VECTOR

        Vector    Name                Assigned to VDN          More VDN's
Master    1      Number 9
          3
```

Vector

The first row displays the existing master vector showing the vector number and name (if assigned) for the master vector specified in the duplicate vector command line. In the next row, enter the number of an unassigned vector between 1 and 2000 (1 to 256 for S8300/S8400 platforms).

The lines following the master vector are for defining the vector numbers and names for the duplicates to be created. The screen shows 16 lines numbered 1 to 16 for specifying the vector numbers and (optionally) names for the vectors that are copies/duplicates of the master vector. If a **`start`** ***`nnnn`*** entry is included on the command line, the specified ***`nnnn`*** number is to be used as the first vector number to be used for creating the duplicates. If a start number is entered on the command line without including a count entry, then only one vector number is pre-entered as the vector number for the duplicate. If the start vector number specified is populated (has one or more steps administered), the first unused vector after the specified start vector number is pre-entered.

If a **`count`** ***`xx`*** entry is included in the command line, that count (***`xx`***) is to be used to define how many vector numbers (up to 16) are to be pre-entered on the Duplicate Vector screen to be used when creating the duplicates of the master vector. The pre-entered vector numbers are numbered sequentially beginning with the first unused vector found either starting with vector number 1 (if a **`start`** ***`nnnn`*** entry is not included) or starting at the specified start number (***`nnnn`***). If any of the vectors in that sequence are already defined with one or more steps assigned, then those numbers (defined vectors) are to be skipped when listing the numbers for the vector duplicates. If the vectors chosen for the pre-entered listing have a name assigned (without any steps populated), the vector names are shown on the Duplicate Vector screen along with the pre-assigned vector numbers. You can change the listing of one or more pre-entered vector numbers to replace the vector numbers chosen by the system, or to add additional vector numbers for duplicates. You can use any unassigned vector number in this field.

Name

Enter a name for the new vector. Entry of a vector name is optional so that duplicates can be created without a vector name entered. Any pre-assigned vector names can also be replaced with a different name which is to be used when creating the duplicates.

Assigned VDN

This field displays an extension number (up to 13 digits) of the first VDN (in numerical extension order) to which the vector is assigned to, if any. The **Assigned VDN** and **More VDN's** columns are populated for the master vector and any of the duplicate vectors which may already be assigned to one or more VDNs. Pre-entered vector numbers have these columns populated when the screen first appears. User-entered vector numbers appear in these columns when tabbing to the next vector number field.

More VDN's

This field displays an asterisk (*) if there is at least one more VDN with that vector assigned.

Enable File Transfer

Use this screen to enable SFTP on TN799BP Control Lan (C-LAN) and VAL circuit packs.

Field descriptions for page 1

Figure 76: Enable File Transfer screen

```
enable filexfer a03                                                  Page 1
                             ENABLE FILE TRANSFER
             Login:______
          Password:___________
          Password:___________
            Secure?
```

Login

Valid entries	Usage
3 to 6 alphanumeric characters	Enter your login.

Password

Valid entries	Usage
7 to 11 characters, with at least one number	Enter your password.

Password

Valid entries	Usage
7 to 11 characters, with at least one number	Repeat your password for verification. Entry must be identical in both **Password** fields.

Secure

Valid entries	Usage
y/n	Enter **y** to enable SFTP instead of FTP or TFTP. If the circuit pack does not support a secure session, no session is enabled. Default is **y**.

Enable Session

Use this screen to enable SSH instead of Telnet.

Field descriptions for page 1

Figure 77: Enable Session screen

```
enable session                                                    Page 1
                             ENABLE SESSION
             Login:______
          Password:___________
          Password:___________
            Secure?
     Time to Login:
```

Login

Valid entries	Usage
3 to 6 alphanumeric characters	Enter your login.

Password

Valid entries	Usage
7 to 11 characters, with at least one number	Enter your password.

Password

Valid entries	Usage
7 to 11 characters, with at least one number	Repeat your password for verification. Entry must be identical in both **Password** fields.

Secure

Valid entries	Usage
y/n	Enter **y** to indicate that SSH is used instead of Telnet. Default is **y**.

Time to Login

This field appears only if the board in question is a TN2302.

Valid entries	Usage
0 to **255**	Enter the number of minutes allowed for login before the session times out. Default is blank.

Extended Pickup Group

This screen allows grouping of pickup groups into extended pickup groups. This allows users to answer calls outside their immediate group. The maximum number of groups that can be added to an extended pickup group is 25.

Field descriptions for page 1

Figure 78: Extended Pickup Group screen

```
change extended-pickup-group n                              Page   1 of  x
                              EXTENDED PICKUP GROUP

                            Extended Group Number: 56

Pickup   Pickup Group                                 Pickup   Pickup Group
Number     Number                                     Number     Number

 0:      _____                                         13:      _____
 1:      _____                                         14:      _____
 2:      _____                                         15:      _____
 3:      _____                                         16:      _____
 4:      _____                                         17:      _____
 5:      _____                                         18:      _____
 6:      _____                                         19:      _____
 7:      _____                                         20:      _____
 8:      _____                                         21:      _____
 9:      _____                                         22:      _____
10:      _____                                         23:      _____
11:      _____                                         24:      _____
12:      _____                                         25:      _____
```

Extended Group Number

This display-only field shows the number associated with a collection of pickup groups. The extended group is a collection of pickup groups that can answer calls from other pickup groups in the same extended group.

Pickup Group Number

This field determines which call pickup groups can answer calls in the extended pickup group.

Valid entries	Usage
1 to **800** or blank (DEFINITY CSI) **1** to **5000** or blank (S8300 and S87XX Servers)	Enter the pickup group numbers for each of the pickup groups that you want to belong to this extended group.

Pickup Number

This display-only field shows the pickup number assigned to the pickup group number. This is the number users dial after the feature access code (FAC) to pick up calls in their extended pickup group.

Extensions Administered to have an MCT-Control Button

This screen lists the extensions that can take control of a Malicious Call Trace (MCT) request. In order to give a user the ability to take control of such requests, you need to add their extension to this list and assign them a **mct-control** feature button.

Field descriptions for page 1

Figure 79: Extensions Administered to have an MCT-Control Button screen

```
 display mct-group-extensions                                          Page 1 of 2

Extensions Administered to have an MCT-Control Button:
    1234567890123       1234567890123       1234567890123       1234567890123
 1: 41917           19:                 37:                 55:
 2:                 20:                 38:                 56:
 3: 41963           21:                 39:                 57:
 4:                 22:                 40:                 58:
 5: 41801           23:                 41:                 59:
 6: 41973           24:                 42:                 60:
 7: 43911           25:                 43:                 61:
 8:                 26:                 44:                 62:
 9: 41908           27:                 45:                 63:
10:                 28:                 46:                 64:
11:                 29:                 47:                 65:
12:                 30:                 48:                 66:
13:                 31:                 49:                 67:
14:                 32:                 50:                 68:
15:                 33:                 51:                 69:
16:                 34:                 52:                 70:
17:                 35:                 53:                 71:
18:                 36:                 54:                 72:
```

1 to 100

Enter the extension for a telephone or attendant console that you want to have an **MCT-Control** button. Note that you must also assign the **mct-control** button on the extension's Station or Attendant Console screen.

Note:
Page 2 contains elements 73 to 100.

Extensions to Call Which Activate Features by Name

With this screen, you can assign a dialed extension to a feature within Communication Manager. This extension is called a feature name extension (FNE). You must set up the FNE mapping. All extensions must fit your dial plan and because they are implemented system-wide. These extensions are paired with feature access codes (FACs). When a user calls the extension, the feature access code activates the feature. Administer the FACs on the Feature Access Code (FAC) screen. For more information about individual features, see *Avaya Aura™ Communication Manager Feature Description and Implementation*, 555-245-205.

To administer this screen, type:

`change off-pbx-telephone feature-name-extensions [set <1-99>]` or blank.

The set number you designate in the command displays. If you do not enter a set number with the command, set 1 automatically displays.

Note:
The set number is not the same as the location number on the Communication Manager **Locations** screen.

Figure 80: Extensions to Call which Activate Features By Name screen - page 1

```
change off-pbx-telephone feature-name-extensions                Page   1 of   x
     EXTENSIONS TO CALL WHICH ACTIVATE FEATURES BY NAME
                     Set Name:

        Active Appearance Select:
             Automatic Call Back:
      Automatic Call-Back Cancel:
                Call Forward All:
     Call Forward Busy/No Answer:
             Call Forward Cancel:
                       Call Park:
           Call Park Answer Back:
                    Call Pick-Up:
            Calling Number Block:
          Calling Number Unblock:
   Conditional Call Extend Enable:
  Conditional Call Extend Disable:
             Conference Complete:
            Conference on Answer:
           Directed Call Pick-Up:
           Drop Last Added Party:
```

Figure 81: Extensions to Call which Activate Features By Name screen - page 2

```
change off-pbx-telephone feature-name-extensions                Page   2 of   x
     EXTENSIONS TO CALL WHICH ACTIVATE FEATURES BY NAME

       Exclusion (Toggle On/Off):
      Extended Group Call Pickup:
          Held Appearance Select:
          Idle Appearance Select:
              Last Number Dialed:
            Malicious Call Trace:
     Malicious Call Trace Cancel:
             Off-Pbx Call Enable:
            Off-Pbx Call Disable:
                   Priority Call:
                          Recall:
                  Send All Calls: 27090
           Send All Calls Cancel: 27091
               Transfer Complete:
             Transfer On Hang-Up:
          Transfer to Voice Mail:
         Whisper Page Activation:
```

Field descriptions

Extension

Each **Extension** field is an extension that matches your dial plan. A user dials the extension from their Extension to Cellular telephone to activate an FAC administered for that feature.

Valid entries	Usage
0 to **9** or blank	Type any valid and assigned extension number for the Communication Manager feature you want users to access from their Extension to Cellular telephones. Default is blank.

Note:
The **Transfer to Voice Mail** FNE is used when a user is active on a call and wants to transfer the other party to voice mail, or to the principal's voice mail, if this is a covered call. This FNE can also be used when a user goes off-hook for the first time and dials the **Transfer to Voice Mail** FNE to be connected to the voice mail administered in his coverage path. This is identical to dialing a Transfer to Voice Mail feature access code (FAC).

Feature Access Code (FAC)

This screen assigns feature access codes (FACs) that, when dialed, activate or cancel the system features. Each field on this screen has the same valid values, which must conform to feature access codes or dial access codes as defined by your dial plan.

Field descriptions for page 1

Figure 82: Feature Access Code (FAC) screen

```
change feature-access-codes                                       Page   1 of   x
                               FEATURE ACCESS CODE (FAC)
         Abbreviated Dialing List1 Access Code:
         Abbreviated Dialing List2 Access Code:
         Abbreviated Dialing List3 Access Code:
Abbreviated Dial - Prgm Group List Access Code:
                      Announcement Access Code: *11
                       Answer Back Access Code:
                         Attendant Access Code:
      Auto Alternate Routing (AAR) Access Code:
    Auto Route Selection (ARS) - Access Code 1: 9     Access Code 2:
                 Automatic Callback Activation:        Deactivation:
Call Forwarding Activation Busy/DA:        All:        Deactivation:
   Call Forwarding Enhanced Status:        Act:        Deactivation:
                         Call Park Access Code:
                       Call Pickup Access Code:
CAS Remote Hold/Answer Hold-Unhold Access Code:
                  CDR Account Code Access Code:
                        Change COR Access Code:
                   Change Coverage Access Code:
            Conditional Call Extend Activation:        Deactivation:
                   Contact Closure   Open Code:          Close Code:
```

Abbreviated Dialing List1 Access Code

Used to access AD list 1.

Abbreviated Dialing List2 Access Code

Used to access AD list 2.

Abbreviated Dialing List3 Access Code

Used to access AD list 3.

Abbreviated Dial - Prgm Group List Access Code

Used to enter a group list from a telephone. The user's extension must be entered on the Abbreviated Dial Group List screen in order to program the group list.

Announcement Access Code

Used to record announcements.

Answer Back Access Code

Used to retrieve parked calls.

Attendant Access Code

This field only appears and is valid if an **attd** entry does not exist on the Dial Plan Analysis screen. You cannot have an entry in both the Dial Plan Analysis screen and the Feature Access Code (FAC) screen. While the Dial Plan Analysis screen allows administration of only one attd code that connects to one attendant, this field on the Feature Access Code (FAC) screen allows you to administer more than one attendant access code in a single distributed network. Attendant access numbers can start with any number from 0 to 9 and contain 1 or 2 digits.

Auto Alternate Routing (AAR) Access Code

Used to access AAR.

Auto Route Selection (ARS) Access Code1

Used to access ARS. You can have one ARS access code for local and one for long distance, and route accordingly.

(ARS) Access Code 2

Also used to access ARS.

Automatic Callback Activation/Deactivation

Used to activate/cancel Automatic Callback.

Call Forwarding Activation Busy/DA

Used to forward calls to an administered number if the user is busy or does not answer.

Call Forwarding Enhanced Activation/Deactivation

Enter feature access code numbers to allow users to activate and deactivate Enhanced Call Forwarding. The FACs for activation and deactivation must be administered together. One can't exist without the other. In contrast, the FAC for **Call Forwarding Enhanced Status** can exist by itself and without the others.

Call Forwarding Enhanced Status

Used to display the status of Enhanced Call Forwarding.

Call Park Access Code

Used to park an active call, which can then be retrieved from a different station using the answer back access code. Do not administer to have the same first digit as another feature access code that is longer in length.

Call Pickup Access Code

Used to answer a call directed to a pickup group.

CAS Remote Hold/Answer Hold-Unhold Access Code

Used by a Centralized Attendant Service (CAS) attendant to place calls on hold and answer calls held at a remote server running Communication Manager. This FAC can also be used by an analog station. Flashing the switch-hook for the proper interval (between 200 and 1000 ms) while talking on an existing call causes the existing call to be placed on soft hold, allowing the analog user to dial the Answer Hold-Unhold FAC to Hard hold the call.

CDR Account Code Access Code

Used prior to entering an account code for CDR purposes.

Change COR Access Code

Used to allow users to change their class of restriction (COR) from a telephone. This field can only be used if the **Change COR by FAC** field is enabled on the System Parameters Customer-Options (Optional Features) screen.

Change Coverage Access Code

Used to change a coverage path from a telephone or remote station.

Contact Closure Close Code

Used to close a contact closure relay. Must be administered if the **Contact Closure Open Code** field is administered.

Contact Closure Open Code

Used to open a contact closure relay. Must be administered if the **Contact Closure Close Code** field is administered.

Field descriptions for page 2

Figure 83: Feature Access Code (FAC) screen

```
change feature-access-codes                                    Page   2 of  x
                               FEATURE ACCESS CODE (FAC)
                   Contact Closure  Pulse Code:

                  Data Origination Access Code:
                      Data Privacy Access Code:
              Directed Call Pickup Access Code:
        Directed Group Call Pickup Access Code:
     Emergency Access to Attendant Access Code:
        EC500 Self-Administration Access Codes:
                     Enhanced EC500 Activation:        Deactivation:
            Enterprise Mobility User Activation:        Deactivation:
  Extended Call Fwd Activate Busy D/A      All:        Deactivation:
        Extended Group Call Pickup Access Code:
               Facility Test Calls Access Code:
                             Flash Access Code:
             Group Control Restrict Activation:        Deactivation:
                    Hunt Group Busy Activation:        Deactivation:
                              ISDN Access Code:
                Last Number Dialed Access Code:
     Leave Word Calling Message Retrieval Lock:
   Leave Word Calling Message Retrieval Unlock:
```

Contact Closure Pulse Code

Used to pulse a contact closure relay.

Data Origination Access Code

Used to originate a data call from a voice station.

Data Privacy Access Code

Used to isolate a data call from call waiting or other interruptions.

Directed Call Pickup Access Code

Used to establish directed call pickup.

Directed Group Call Pickup Access Code

The feature access code (FAC) can be used to pick up a call from any pickup group, if the user belongs to a pickup group.

EC500 Self Administration Access Code

The Self Administration Feature (SAFE) Access Code allows users to self-administer their cell phone number for the Extension to Cellular feature. Users can add or change their cell phone number through this feature access code. An administrator can still enter or change cell phone numbers. The user calls the SAFE access code and enters their cell phone number. The administration sequence differs based on what telephone is used to access SAFE. For more information, see *Avaya Aura™ Communication Manager Feature Description and Implementation*, 555-245-205 and *Avaya Extension to Cellular User's Guide*, 210-100-700.

Emergency Access To Attendant Access Code

Used to gain access to the attendant in an emergency situation. Such calls alert as emergency calls. This field cannot be used if the **Emergency Access to Attendant** field is not enabled on the System Parameters Customer-Options (Optional Features) screen.

Enhanced EC500 Activation

Type a feature access code number to allow users to activate Extension to Cellular remotely.

Enhanced EC500 Deactivation

Type a feature access code number to allow users to deactivate Extension to Cellular remotely.

Enterprise Mobility User Activation

Type a feature access code number to activate the Enterprise Mobility User feature for a particular user, associating the features and permissions of their primary telephone to a telephone of the same type anywhere within the customer's enterprise. For more information about Enterprise Mobility User, see *Administering Avaya Aura™ Communication Manager*, 03-300509.

Enterprise Mobility User Deactivation

Type a feature access code number to deactivate the Enterprise Mobility User feature. For more information about Enterprise Mobility User, see *Administering Avaya Aura™ Communication Manager*, 03-300509.

Extended Call Fwd Activate Busy D/A

Used to activate call forwarding from a telephone or remote location.

Extended Call Fwd Activate All

Used to activate call forwarding from a telephone or remote location.

Extended Call Fwd Deactivation

Used to deactivate call forwarding from a telephone or remote location.

Note:
An extension must have Station Security Codes administered to use the following FACs:

- Extended Call Forward All Activate
- Extended Call Forward Busy/Don't Answer Activate
- Extended Call Forward Deactivate
- Change Coverage

Extended Group Call Pickup Access Code

The feature access code (FAC) users enter when a call directed to another pickup group is to be answered. Users must enter a valid "Pickup Number" following the Extended Group Call Pickup Access Code to complete the operation.

Facility Test Calls Access Code

Used to place activate a facility test call.

SECURITY ALERT:
To ensure the security of your system, leave Facility Test Calls Access Code blank except when actually testing trunks.

Flash Access Code

Used to generate trunk flash. This code ensures that the flash signal is interpreted by the central office switch, rather than by Communication Manager.

Group Control Restrict Activation / Deactivation

Used to change the restriction level for all users with a given class of restriction. Requires console permissions.

Hunt Group Busy Activation/Deactivation

Hunt group members use the **Hunt Group Busy Activation** FAC to make the extension unavailable and the **Hunt Group Busy Deactivation** FAC to make the extension available.

The hunt group member must dial:

- a two-digit hunt group number for a hunt group supporting up to 99 extensions (small hunt group) or
- a four-digit hunt group number for a hunt group supporting up to 8000 extensions (large hunt group)

If needed, add zeros before the hunt group number to ensure that the small hunt group has a two-digit number and the large hunt group has a four-digit number.

ISDN Access Code

Used to place an ISDN call without using ARS, AAR, or UDP.

Last Number Dialed Access Code

Used to redial the last number dialed from this station.

Leave Word Calling Message Retrieval Lock

Used to lock the display module on telephones. The lock function activates at a telephone by dialing this system-wide lock access code. This prevents unauthorized users from displaying, canceling, or deleting messages associated with the telephone. The **Lock Messages** field on the Station screen also must be enabled.

Leave Word Calling Message Retrieval Unlock

Used to unlock a telephone's display module. The lock function is canceled at the telephone by dialing this unlock FAC followed by the SCC.

Field descriptions for page 3

Figure 84: Feature Access Code (FAC) screen

```
change feature-access-codes                                   Page 3 of x
                             FEATURE ACCESS CODE (FAC)
             Leave Word Calling Send A Message: *60
           Leave Word Calling Cancel A Message: #60
   Limit Number of Concurrent Calls Activation:         Deactivation:
               Malicious Call Trace Activation:         Deactivation:
         Meet-me Conference Access Code Change:

PASTE  (Display PBX data on Phone) Access Code:
  Personal Station Access (PSA) Associate Code:        Dissociate Code:
        Per Call CPN Blocking Code Access Code:
      Per Call CPN Unblocking Code Access Code:
                              Posted Messages:
          PIN Checking for Private Calls Access Code:*11
PIN Checking for Private Calls Using AAR Access Code:*12
PIN Checking for Private Calls Using ARS Access Code:*13

                  Priority Calling Access Code:
                           Program Access Code:

       Refresh Terminal Parameters Access Code:
              Remote Send All Calls Activation:       Deactivation:
              Self Station Display Access Code:
                     Send All Calls Activation:       Deactivation:
         Station Firmware Download Access Code:
```

Leave Word Calling Send A Message

Used to send a leave word calling message.

Leave Word Calling Cancel A Message

Used to cancel a leave word calling message.

Limit Number of Concurrent Calls Activation/Deactivation

Used to limit concurrent calls on a station even when additional call appearances normally would be available.

Malicious Call Trace Activation

Used to activate a trace request on a malicious call.

Meet-me Conference Access Code Change

Allows the controlling user of a Meet-me Conference VDN to change the access code.

PASTE (Display PBX data on telephone) Access Code

Allows users to view call center data on display telephones. PASTE is used in conjunction with Avaya IP Agent.

Personal Station Access (PSA) Associate Code

Used to associate a telephone with the telephone features assigned to a user’s extension.

Personal Station Access (PSA) Dissociate Code

Used to remove the association between a physical telephone and an extension number. You cannot provide the code until Personal Station Access (PSA) on the System Parameters Customer-Options **(Optional Features)** screen is **y**.

Per Call CPN Blocking Code Access Code

If CPN blocking is off for a trunk group, users can turn it on for a call by using this code. When they dial this code, the calling party number is not sent to the public network.

Per Call CPN Unblocking Code Access Code

If CPN blocking is on for a trunk group, users can turn it off for a call by using this code. When they dial this code, the calling party number is sent to the public network.

Posted Messages

Only appears if the **Posted Messages** field is set to **y** on the **System Parameters Customer-Options (Optional Features)** screen. Used to access the Posted Messages feature. See Posted Messages in *Avaya Aura™ Communication Manager Feature Description and Implementation,* 555-245-205, for more information

PIN Checking for Private Calls Access Code

When the feature is enabled via system-parameters features in Figure 84: Feature Access Code (FAC) screen on page 266, then it displays the three new fields:

- **PIN Checking for Private Calls Access Code**
- **PIN Checking for Private Calls Using AAR Access Code**
- **PIN Checking for Private Calls Using ARS Access Code**

Note:
These 3 fields show up only when the feature is enabled via system-parameters features. If the feature is not enabled, the 3 fields remain hidden. PIN FACs *11/*12/*13 are visible only when the feature **PIN Checking for Private Calls** is activated in Figure 108: Feature-Related System Parameters screen on page 377.

Priority Calling Access Code

Used to enable priority calling, a special type of call alerting between internal telephone users, including the attendant. The called party hears a distinctive ringing when the calling party uses Priority Calling.

Program Access Code

Used to program abbreviated dial buttons on an individual telephone.

Refresh Terminal Parameters Access Code

Used to update terminal parameters on an individual telephone when system settings have changed.

Remote Send All Calls Activation/Deactivation

Used to activate or deactivate the Send All Calls feature. Requires console permissions.

Self Station Display Activation

The **Self Station** field is not active. If set to a valid FAC, a digital station displays its primary extension number when the FAC is entered.

Send All Calls Activation/Deactivation

Used to activate or deactivate sending all calls to coverage with minimal or no alerting at the station.

Station Firmware Download Access Code

This field specifies the feature access code used for 2420/2410 DCP station firmware downloads.

Field descriptions for page 4

Figure 85: Feature Access Code (FAC) screen

```
change feature-access-codes                                    Page 4 of x
                             FEATURE ACCESS CODE (FAC)
                       Station Lock Activation:       Deactivation:
      Station Security Code Change Access Code:
              Station User Admin of FBI Assign:                Remove:
  Station User Button Ring Control Access Code:
                          Telephone Activation: #*
             Terminal Dial-up Test Access Code:
Terminal Translation Initialization Merge Code:       Separation Code:
            Transfer to Voice Mail Access Code:
          Trunk Answer Any Station Access Code:
              User Control Restrict Activation:       Deactivation:
  Voice Coverage Message Retrieval Access Code:
 Voice Principal Message Retrieval Access Code: *80
           Whisper Page Activation Access Code:
```

Station Lock Activation/Deactivation

Used to activate or deactivate Station Lock.

Station Security Code Change Access Code

Enter the code the user must dial to change their Station Security Code. The SCC must be administered before the user can change it using this FAC. That is, a user cannot change a blank SCC.

Station User Admin of FBI Assign

Used to activate or deactivate Facility Busy Indicators.

Station User Button Ring Control Access Code

Used to control the ring behavior for each line appearance and bridged appearance from the station. Allows users to have their telephones ring either silently or audibly.

Terminal Dial-Up Test Access Code

Used to perform tests on digital telephones to make sure that the telephone and the buttons are communicating properly with the server running Communication Manager. The Terminal Dial-Up test ensures that the terminal and each of its buttons can communicate with the server. This test is initiated by a user entering this feature access code. This test is mostly for use by terminal service personnel, but may be used by any station user.

In order to use this feature, simply lift the receiver on a supported terminal and dial the feature access code. The terminal's button lights are extinguished, the display clears, and the message waiting lamp lights. This lamp remains lit throughout the test. When a button is depressed during the test, the server responds with the appropriate tone, light, or display. Depressing another button clears the lamp, the tone and ringer associated with the previous button and lights the lamp, sends new tone and ringer associated with the new button. If the same button is depressed, the lamp, tone and ringer are turned OFF, which makes the button work as an ON/OFF button. The terminal remains in this test mode until you hang up the receiver. To use a Terminal Dial-up Test FAC on a telephone with bridged appearances, add a bridged-appearance of the principal telephone.

Terminal Translation Initialization Merge Code

Enter the digits that must be dialed to install (merge) a station without losing any of its previous feature settings. The Terminal Translation Initialization Separation Code must have been used, or an **X** administered in the **Port** field of the Station screen, when the telephone was removed from its former location in order for the Terminal Translation Initialization Merge Code to be effective. (If you try to use this and it doesn't work, check the Station screen for this extension. If there is still a port assigned, type **X** in the **Port** field, then try the TTI merge again.)

Terminal Translation Initialization Separation Code

Enter the digits that must be dialed to remove (separate) a station from a location without losing any of its feature settings.

Transfer to Voice Mail Access Code

Enter the digits that must be dialed to allow coverage to transfer the caller to the original call recipient's AUDIX mail where the caller can leave a message. Do not administer this code to have the same first digit as another feature access code that is longer in length.

Trunk Answer Any Station Access Code

Enter the access code that station users must dial to answer calls alerting on night bells.

User Control Restrict Activation/Deactivation

Used to change the restriction level for a specific extension. Requires console permissions.

Voice Coverage Message Retrieval Access Code

Allows users to retrieve voice messages for another user (for whom they are a coverage point) via a digital display module.

Voice Principal Message Retrieval Access Code

Allows users to retrieve their own voice messages for another user via a digital display module.

Whisper Page Activation Access Code

Allows users to place a page to another user's telephone, when active on a call. The paged user, and not the other parties on the call, hears the page.

Field descriptions for page 5

The feature access codes on this page pertain only to ACD call centers.

Figure 86: Feature Access Code (FAC) screen

```
change feature-access-codes                                   Page 5 of x
                        FEATURE ACCESS CODE (FAC)

                  Automatic Call Distribution Features

                   After Call Work Access Code: ___
                            Assist Access Code: ___
                           Auto-In Access Code: ___
                          Aux Work Access Code: ___
                             Login Access Code: ___
                            Logout Access Code: ___
                         Manual-In Access Code: ___
     Service Observing Listen Only Access Code: ___
     Service Observing Listen/Talk Access Code: ___
         Service Observing No Talk Access Code: ___
                   Add Agent Skill Access Code: ___
                Remove Agent Skill Access Code: ___
            Remote Logout of Agent Access Code: ___
```

Add Agent Skill Access Code

Enter the digits an agent must dial to be able to add a skill to their current skill set.

After Call Work Access Code

Enter the code the agent must dial when the agent performs work-related ACD activities.

Assist Access Code

Enter the digit the agent must dial to request assistance from the split supervisor.

Auto-In Access Code

Enter the code the agent must dial to become automatically available to receive another ACD call each time a call is released.

Aux Work Access Code

Enter the code the agent must dial when the agent performs non-ACD activities.

Login Access Code

Enter the code the agent must dial to gain access to the ACD functions. This is a system-wide code for all ACD agents.

Logout Access Code

Enter the logout code the agent must enter to exit ACD. This is a system-wide logout code for all ACD agents.

Manual-In Access Code

Enter the code the agent must dial to receive a single, new ACD call upon the completion of an ACD call.

Note:
The following two fields appear only if **Service Observing (Remote/By FAC)** on the System Parameters Customer-Options **(Optional Features)** screen is **y**.

Remove Agent Skill Access Code

This field appears only if **Service Observing (Remote/By FAC)** on the System Parameters Customer-Options **(Optional Features)** screen is **y**. Enter the digits an agent must dial to be able to remove a skill from their current skill set.

Remote Logout of Agent Access Code

This field appears only if **Service Observing (Remote/By FAC)**, **Vectoring (Basic)**, and **Vectoring (Prompting)** on the System Parameters Customer-Options (Optional Features) screen are set to **y.** Enter the digits you need to dial to remotely logout an idle ACD or EAS agent.

Service Observing Listen Only Access Code

Enter the code that must be dialed to allow a station with Service Observing permission (COR) to listen to other agent ACD calls without being heard on the ACD call.

Service Observing Listen/Talk Access Code

Enter the code that must be dialed to allow a station with Service Observing permission (COR) to both listen and be heard on an ACD call.

Service Observing No Talk Access Code

The following field appears only if **Expert Agent Selection (EAS) Enabled** is optioned on the Feature-Related System-Parameters screen. Enter the code that must be dialed to allow a station with Service Observing permission (COR) to listen only without reserving a 2nd timeslot for potential toggle to talk and listen mode. When this FAC is used for activation, the observing connection is listen only. Any attempt to toggle to talk via the Service Observing (SO) feature button is denied.

Field descriptions for page 6

The feature access codes on this page pertain only to Call Vectoring/ Prompting features.

Figure 87: Feature Access Code (FAC) screen

```
change feature-access-codes                                    Page 6 of x
                             FEATURE ACCESS CODE (FAC)

                        Call Vectoring/Prompting Features

     Converse Data Return Code:

   Vector Variable 1 (VV1) Code:
   Vector Variable 2 (VV2) Code:
   Vector Variable 3 (VV3) Code:
   Vector Variable 4 (VV4) Code:
   Vector Variable 5 (VV5) Code:
   Vector Variable 6 (VV6) Code:
   Vector Variable 7 (VV7) Code:
   Vector Variable 8 (VV8) Code:
   Vector Variable 9 (VV9) Code:
```

Converse Data Return Code

Enter a 1 to 4 digit number (# can be used as the first digit). If there is data to be returned to the switch, the Converse Data Return Code is outpulsed before the data to be passed is outpulsed.

Vector Variable x (VVx) Code (1-9)

Enter a 1 to 4 digit number (# can be used as the first digit). Entry of the Vector Variable Code FAC allows you to change the variable on the **Variables for Vectors** screen.

Field descriptions for page 7

The feature access codes on this page pertain only to Hospitality features.

Figure 88: Feature Access Code (FAC) screen

```
change feature-access-codes                                    Page 7 of x
                             FEATURE ACCESS CODE (FAC)

                               Hospitality Features

                          Automatic Wakeup Call Access Code: *11____
                 Housekeeping Status (Client Room) Access Code: _______
                 Housekeeping Status (Client Room) Access Code: _______
                 Housekeeping Status (Client Room) Access Code: _______
                 Housekeeping Status (Client Room) Access Code: _______
                 Housekeeping Status (Client Room) Access Code: _______
                 Housekeeping Status (Client Room) Access Code: _______
                     Housekeeping Status (Station) Access Code: _______
                     Housekeeping Status (Station) Access Code: _______
                     Housekeeping Status (Station) Access Code: _______
                     Housekeeping Status (Station) Access Code: _______
                        Verify Wakeup Announcement Access Code: _______
                              Voice Do Not Disturb Access Code: _______
```

Automatic Wakeup Call Access Code

Enter the access code the user must dial to schedule or cancel a wakeup call.

Housekeeping Status (Client Room) Access Code

Enter the access code the housekeeper dials from the client's room to provide room status. These codes are transmitted to the Property Management System (PMS) for processing. You can assign a definition to the six codes on the **Hospitality** screen.

Housekeeping Status (Station) Access Code

Enter the access code the housekeeper must dial to provide room status. This access code must be dialed from designated telephones. There are four codes.

Verify Wakeup Announcement Access Code

Enter the access code the user can dial to verify a wakeup announcement.

Voice Do Not Disturb Access Code

Enter the access code the user must dial to enter or cancel a do not disturb request without using a display - through the use of voice prompting.

Field descriptions for Multimedia Features page

The feature access codes on this page pertain only to Multimedia Call Handling (MMCH).

Figure 89: Feature Access Code (FAC) screen

```
change feature-access-codes                                       Page 8 of x
                               FEATURE ACCESS CODE (FAC)
                                  Multimedia Features

                       Basic Mode Activation:
                    Enhanced Mode Activation:
                 Multimedia Call Access Code:
       Multimedia Data Conference Activation:  Deactivation:
        Multimedia Multi-Address Access Code:
            Multimedia Parameter Access Code:
```

Basic Mode Activation

If you enter this FAC when your system is an Enhanced multimedia complex, it reverts to a Basic multimedia complex. If you enter this FAC when your system is a Basic mode station it does not perform any activities.

Enhanced Mode Activation

If you enter this FAC when your system is a Basic multimedia complex, it becomes an Enhanced multimedia complex. If you enter this FAC when your system is an Enhanced mode station it does not perform any activities.

Multimedia Call Access Code

If you enter this FAC from any voice station, it indicates to Communication Manager that you are making an Enhanced mode multimedia call. If you originate a multimedia call with the multimedia call access code, it originates a call according to the Default Multimedia Parameters selected on the Feature-Related System Parameters screen.

Multimedia Data Conference Activation

If you enter this FAC from any voice station that is participating in a multimedia call, it alerts Communication Manager that you want to enable data collaboration with the other parties on the call. If you enter this FAC a second time, it gives denial treatment (since a collaboration session is already active). This FAC only applies to voice stations on servers equipped with ESM adjuncts.

Multimedia Data Conference Deactivation

If you enter this FAC from the telephone that enabled data collaboration on a multimedia mode call, it deactivates the data session and revert to a voice and video call. If a user enters this FAC while participating in a data-collaboration multimedia call that the user did not initiate, the system responds with denial treatment.

Multimedia Multi-Address Access Code

The multimedia multi-address access code is similar to the multimedia call access code. It allows origination of a multimedia call from a voice station. It is used when the destination being dialed requires a different address for each of the 2 B-channels. For example, ISDN-BRI provided by a Central Office is provisioned with separate listed directory numbers for each B-channel. In order to make a 2B multimedia call to such a device, two sets of addresses must be entered.

Originating a multimedia call with the multimedia multi-address access code originates a call according to the Default Multimedia Parameters selected on the Feature-Related System-Parameters screen.

Multimedia Parameter Access Code

This FAC can be entered by any voice station to indicate to Communication Manager that you want to initiate a multimedia mode call with a specific bearer capability. This FAC would be followed by a 1 or 2 to indicate the following parameter selections respectively: 2x64 (unrestricted initial system default), 2x56 (restricted).

Field descriptions for MLPP features page

The feature access codes on this page pertain only to Multiple Precedence and Preemption (MLPP) calls.

Figure 90: Feature Access Code (FAC) screen

```
change feature-access-codes                                    Page 8 of x
                              FEATURE ACCESS CODE (FAC)
                                 MLPP Features

             Precedence Calling Access Code: 8


    WNDP PRECEDENCE ACCESS CODES:
                 Flash Override Access Code: 90
                          Flash Access Code: 91
                      Immediate Access Code: 92
                       Priority Access Code: 93
                        Routine Access Code: 94
```

Precedence Calling Access Code

Enter a feature access code that conforms to your dial plan, to be used to access the Multiple Level Precedence and Preemption feature.

WNDP PRECEDENCE ACCESS CODES

The system uses a feature access code to determine the precedence level for a call when the Worldwide Numbering and Dial Plan (WNDP) feature is active. Different feature access codes are assigned for each PRECEDENCE level. When the WNDP feature is not active, the user dials the PRECEDENCE CALLING feature access code followed by a digit indicating the precedence level of the call.

Flash Override Access Code

Enter a FAC to correspond with the Flash Override preemption level.

Flash Access Code

Enter a FAC to correspond with the Flash preemption level.

Immediate Access Code

.Enter a FAC to correspond with the Immediate preemption level.

Priority Access Code

.Enter a FAC to correspond with the Priority preemption level.

Routine Access Code

.Enter a FAC to correspond with the Routine preemption level.

Feature-Related System Parameters

This screen implements system parameters associated with various system features.

Note:
This screen used to contain Call Coverage and Call Forwarding parameters. These fields have been moved to a separate screen, which you can access with the command **`change system-parameters coverage-forwarding`**.

Field descriptions for page 1

Figure 91: Feature-Related System Parameters screen

```
change system-parameters features                               Page   1 of   x
                            FEATURE-RELATED SYSTEM PARAMETERS
                               Self Station Display Enabled? y
                                    Trunk-to-Trunk Transfer: none
               Automatic Callback with Called Party Queuing? n
    Automatic Callback - No Answer Timeout Interval (rings): 3
                       Call Park Timeout Interval (minutes): 10
        Off-Premises Tone Detect Timeout Interval (seconds): 20
                                 AAR/ARS Dial Tone Required? y
                             Music/Tone on Hold: none
              Music (or Silence) on Transferred Trunk Calls? no
                       DID/Tie/ISDN/SIP Intercept Treatment: attd
    Internal Auto-Answer of Attd-Extended/Transferred Calls: transferred
                  Automatic Circuit Assurance (ACA) Enabled? y
    ACA Referral Calls: local      ACA Referral Destination:
               ACA Short Holding Time Originating Extension:
               ACA Long  Holding Time Originating Extension:
Treat ISDN Call to Busy User as ACA Short Holding Time Call? n
             Abbreviated Dial Programming by Assigned Lists? n
       Auto Abbreviated/Delayed Transition Interval (rings): 2
                     Protocol for Caller ID Analog Terminals: Bellcore
    Display Calling Number for Room to Room Caller ID Calls? n
```

AAR/ARS Dial Tone Required

A second dial tone provides feedback to the user that additional dialing can occur.

Valid entries	Usage
y/n	Enter **y** to indicate a second dial tone is to be given to the calling party on a incoming tie or DID trunk call that is to be routed via AAR/ARS.

Abbreviated Dial Programming by Assigned Lists

Valid entries	Usage
y	Enter **y** to allow programming by station's assigned list.
n	Enter **n** if using Program Access code to indicate which personal list is to be programmed.

ACA Referral Calls

Indicates where ACA referral calls generate. This field only appears when the **Automatic Circuit Assurance (ACA) Enabled** field is **y**.)

Valid entries	Usage
local	Local referral calls generate on and for the local switch.
primary	Primary referral calls generate on the local switch for remote servers/ switches as well as the local switch.
remote	Remote referral calls generate at another server in a DCS network. In this case, the remote node number must also be entered. The remote node number is the same node number as defined on the Dial Plan screen. Also, **ACA** button status transmits to other servers/switches when in a DCS network.

ACA Referral Destination

The specified extension should be equipped with a display module. This field only appears if ACA Referral Calls is **local** or **primary**.

Valid entries	Usage
An extension	Enter the extension on a local server running Communication Manager that is to receive the ACA referral call.
attd	Enter **attd** for attendant.

ACA Remote PBX Identification

This field only appears if **ACA Referral Calls** is **remote**.

Valid entries	Usage
1 to **63**	Enter a number to identify the switch in a DCS network that makes the referral call. Do not define the remote server/switch identified in this field as **local** on the system's Dial Plan screen.

ACA Short Holding Time Originating Extension and ACA Long Holding Time Originating Extension

Valid entries	Usage
An unassigned extension	Do not use the same extension number for both fields. The specified extensions are assigned automatically by the system when the screen is submitted. These fields only display if **ACA Referral Calls** is **local** or **primary**.

Auto Abbreviated/Delayed Transition Interval (rings)

Valid entries	Usage
1 to **16**	Enter the number of rings before an automatic abbreviated/ delayed transition is triggered for a call.

Automatic Callback — No Answer Timeout Interval (rings)

Valid entries	Usage
2 to **9**	Enter the number of times the callback call rings at the calling station before the callback call is canceled.

Automatic Callback With Called Party Queuing

Valid entries	Usage
y	To enable the feature, select **y**.
n	To disable the feature, select **n**. The default value is **n**.

Automatic Circuit Assurance (ACA) Enabled

If **Automatic Circuit Assurance (ACA) Enabled** is **n**, associated ACA fields do not display.

Must have an **aca-halt** button administered on the user's station. If you enable this feature, complete the following ACA-related fields.

Valid entries	Usage
y	Enter **y** if ACA measurements are taken.
n	Otherwise, enter **n**.

Call Park Timeout Interval (minutes)

Valid entries	Usage
1 to **90**	Enter the number of minutes a call remains parked before it cancels.

DID/Tie/ISDN/SIP Intercept Treatment

Valid entries	Usage
A recorded announcement extension	Toll charges do not apply to DID and private network calls routed to an announcement. **Note:** If entering a Multi-Location Dial Plan shortened extension, note the following: When entering a Multi-Location Dial Plan shortened extension in a field designed for announcement extensions, certain administration end validations that are normally performed on announcement extensions are not done, and resultant warnings or submittal denials do not occur. The shortened extensions also do not appear in any display or list that shows announcement extensions. Extra care should be taken to administer the correct type of announcement for the application when assigning shortened extensions.
attd	For system security, Avaya recommends entering **attd** in this field. This routes intercept calls to the attendant and, if the attendant receives several of these, they know a problem exists.

Display Calling Number for Room to Room Caller ID Calls

Valid entries	Usage
y/n	Enter **y** to display the calling number for room to room hospitality calls.

Internal Auto-Answer of Attd-Extended/Transferred Calls

This only applies to digital telephones (except BRI) with a headset or speakerphone capability.

Valid entries	Usage
attd-extended	Enter **attd-extended** to enable IAA for only attendant extended calls.
both	Enter **both** to enable IAA for station transferred and attendant extended calls.
none	Enter **none** to disable IAA for all calls.
transferred	Enter **transferred** to enable IAA for only station transferred calls.

Music (or Silence) On Transferred Trunk Calls

Valid entries	Usage
all	Enter **all** to allow all transferred trunk calls to receive music until the call is answered if the Music-on-Hold feature is available.
no	Enter **no** if trunk callers are to hear music (or silence if Music-on-Hold is not administered) while waiting to be transferred, and then ringback as soon as the transfer is completed till the call is answered.
call-wait	Enter **call-wait** if trunk calls transferred to stations that require the call to wait hear music (if administered); all other transferred trunk calls receive ringback tone.

Music/Tone on Hold

If you use equipment that rebroadcasts music or other copyrighted materials, you might be required to obtain a copyright license from, or pay fees to, a third party. You can purchase a Magic OnHold system, which does not require such a license, from Avaya or our business partners. This field does not appear if **Tenant Partitioning** is **y** on the System Parameters Customer-Options (Optional Features) screen. In that case, use the **Tenant** screen to establish Music on Hold.

Valid entries	Usage
music **tone** **none**	Indicates what a caller hears while on hold. Default is **none**. When **music** is entered, the **Type** field appears to define the music type.

Off-Premises Tone Detect Timeout Interval (seconds)

Valid entries	Usage
5 to **25**	The number of seconds a call progress tone receiver (CPTR) tries to detect dial tone from a trunk during dialing. Once the time-out interval occurs, the call either outpulses on the trunk or gets intercept treatment depending on the setting of the **Outpulse Without Tone** field on page 6 of this screen.

Port

Appears when **Music/Tone on Hold** is **music** and **Type** is **port**. Enter the necessary characters to indicate the port number that provides Music-on-Hold access.

Valid entries	Usage
01 to **03** (DEFINITY CSI) or **1** to **64** (S87XX/S8300 Servers)	First and second characters are the cabinet number
A to **E**	Third character is the carrier
0 to **20**	Fourth and fifth character are the slot number
01 to **04** (Analog TIE trunks) **01** to **31**	Six and seventh characters are the circuit number
1 to **80** (DEFINITY CSI) or **1** to **250** (S87XX/S8300 Servers)	Gateway
V1 to **V9**	Module
01 to **31**	Circuit

Protocol for Caller ID Analog Terminals

Determines the protocol/tones sent to a Caller ID telephone.

Valid entries	Usage
Bellcore	Enter **Bellcore** for Bellcore protocol with 212 modem protocol tones. Used in the U.S. and similar countries.
V23-Bell	Enter **V23-Bell** for Bellcore protocol with V.23 modem tones. Used in Bahrain and similar countries.

Self Station Display Enabled

Use this field to control the use of the **inspect** button for digital display telephones.

Self Station Display allows a user to display the primary extension associated with a digital display telephone. There are two methods: (1) enter a feature access code (FAC), and (2) use the **inspect** button. In either case, the display shows the primary extension associated with the telephone where the FAC or **normal** or **exit** button is entered. In the case of the FAC, the display continues until a display-altering event occurs (for instance, going on-hook or receiving an incoming call). In the case of the **inspect** button, the display continues until the user presses the **normal** or **exit** button or until a display-altering event occurs.

Valid entries	Usage
y	The primary extension does display when the **inspect** button is pressed.
n	The extension does not display when the **inspect** button is pressed.

Trunk-to-Trunk Transfer

Regulations in some countries control the settings for this field. See your Avaya technical support representative for assistance.

Valid entries	Usage
all	Enter **all** to enable all trunk-to-trunk transfers. This allows telephone users to set up trunk-to-trunk transfer, go on-hook without disconnecting the call, and forward the call to a remote location. This value is required for SIP Enablement Services (SES) support.
restricted	Enter **restricted** (restricted public) to restrict all public trunks (CO, WATS, FX, CPE, DID, and DIOD).
none	Enter **none** to restrict all trunks (except CAS and DCS) from being transferred.

Type

This field appears when **Music/Tone on Hold** is set to **music**.

Note:
If the **Tenant Partitioning** field on the System Parameters Customer-Options (Optional Features) screen is set to **y**, you cannot administer the **Music/Tone on Hold** field. If the **Tenant Partitioning** field on the System Parameters Customer-Options (Optional Features) screen set to **y**, you must use the **Music Sources** screen to assign music to a port.

Valid entries	Usage
ext **group** **port**	● Indicate whether the source for Music on Hold is an announcement extension, an audio group, or a port on a VAL board. ● Type **ext** and the corresponding extension number of the integ-mus announcement/audio source. ● Type **group** and the corresponding Music-on-Hold analog group number. ● Type **port** and the corresponding location of the Music-on-Hold analog/aux-trunk source. **Note:** After a valid value is entered, a blank field appears for entry of the appropriate source identifier (extension number, audio group number, or port number).

Field descriptions for page 2

Figure 92: Feature-Related System Parameters screen

```
change system-parameters features                                   Page 2 of x
                        FEATURE-RELATED SYSTEM PARAMETERS

LEAVE WORD CALLING PARAMETERS
                            Maximum Number of Messages Per Station: 10
               Maximum Number of External Calls Logged Per Station: 0
                     Message Waiting Indication for External Calls? n
  Stations with System-wide Retrieval Permission (enter extension)
    1:1234567890123   9:1234567890123   17:1234567890123   25:1234567890123
    2:               10:                18:                26:
    3:               11:                19:                27:
    4:               12:                20:                28:
    5:               13:                21:                29:
    6:               14:                22:                30:
    7:               15:                23:
    8:               16:                24:

                   Prohibit Bridging Onto Calls With Data Privacy? n
                        Enhanced Abbreviated Dial Length (3 or 4): 3
            Default Multimedia Outgoing Trunk Parameter Selection: 2x64
```

LEAVE WORD CALLING PARAMETERS

Maximum Number of Messages Per Station

Valid entries	Usage
0 to **125**	The maximum number of LWC Messages that can be stored by the system for a telephone at a given time.

Maximum Number of External Calls Logged Per Station

When an external call is not answered, the server running Communication Manager keeps a record of up to 15 calls (provided information on the caller identification is available) and the telephone’s message lamp lights. The telephone set displays the names and numbers of unsuccessful callers

.

Valid entries	Usage
0 to **15**	The maximum number of calls that can be logged for each user. The assigned number cannot be larger than the entry in the **Maximum Number of Messages Per Station (when MSA not in service)** field.

Message Waiting Indication for External Calls

Provides a message waiting indication when external calls are logged.

Valid entries	Usage
y	The message waiting indication for a particular station is on whenever an external call is logged.
n	The log of external calls has no impact on the message waiting indication.

Default Multimedia Outgoing Trunk Parameter Selection

Does not appear on S87XX Series IP-PNC.

Valid entries	Usage
2x56	Sets default parameter for bandwidth and bearer for all video calls.
2x64	

Enhanced Abbreviated Dial Length (3 or 4)

The administrator might not be able to use all entry slots because of system capacity constraints.

Valid entries	Usage
3	A value of **3** makes 1000 Enhanced List entries available to the administrator
4	A value of **4** makes 10,000 entries available.

Prohibit Bridging Onto Calls with Data Privacy

Valid entries	Usage
y/n	Enter **y** to protect calls from bridge-on by any party, including Service Observing, Intrusion, Verify, and Bridging.

Stations With System-wide Retrieval Permission (enter extension)

An extension must be removed from this list before the station is removed from the system. The server running Communication Manager refers to the extensions on this list as "super-retrievers."

Valid entries	Usage
An assigned extension	Enter up to 10 telephone extension numbers that can retrieve LWC Messages or External Call Log records for all other telephones. A VDN extension is not allowed.
attd	An entry of **attd** gives retrieval permission to all attendants.

Field descriptions for page 3

Figure 93: Feature-Related System Parameters screen

```
change system-parameters features                              page 3 of x

                         FEATURE-RELATED SYSTEM PARAMETERS
TTI/PSA PARAMETERS

   WARNING! SEE USER DOCUMENTATION BEFORE CHANGING TTI STATE

         Terminal Translation Initialization (TTI) Enabled? y_
              TTI State: ________        TTI Security Code:
         Enhanced PSA Location/Display Information Enabled?
                          Default COR for Dissociated Sets:
                             CPN, ANI for Dissociated Sets:
           Unnamed Registrations and PSA for IP Telephones?
               Customer Telephone Activation (CTA) Enabled?
  Don't Answer Criteria for Logged off IP/PSA/TTI Stations? n
                     Hot Desking Enhancement Station Lock? n

EMU PARAMETERS
          EMU Inactivity Interval for Deactivation (hours): 1

CALL PROCESSING OVERLOAD MITIGATION
Restrict Calls:
```

TTI/PSA PARAMETERS

CPN, ANI for Dissociated Sets

Appears when the **Default COR for Dissociated Sets** field is non-blank. Specifies the ISDN calling party number (CPN), R2-MFC ANI, and CAMA CESID applied to calls made from PSA dissociated sets, if no system-wide calling party information has been administered for those protocols on their respective administration screens.

Valid entries	Usage
1 to 20 digits	Enter the calling party number or automatic number identification for calls made from dissociated telephones.

Customer Telephone Activation (CTA) Enabled

Valid entries	Usage
y	Enter **y** if you want the Customer Telephone Activation feature for your system.
n	Enter n if you do not want the Customer Telephone Activation feature for your system.

Default COR for Dissociated Sets

Appears when the **Terminal Translation Initialization (TTI) Enabled** field is **y**.

Valid entries	Usage
0 to **995** or blank	Specify the Class of Restriction (COR) that the system uses for calls made from dissociated telephones.

Don’t Answer Criteria for Logged Off IP/PSA/TTI Stations

Use this field to control call process handling for logged-off IP/PSA/TTI Terminals.

Valid entries	Usage
y	If this field is set to **y(es)**, the caller hears a ringback tone on a call to a logged-off IP/PSA/TTI terminal. If a coverage path is administered, the coverage for “Don’t Answer” is used. If Enhanced Call Forwarding is administered, the "Don’t Answer" path is used.
n	If this field is set to **n(o)**, the caller hears a busy tone on a call to a logged-off IP/PSA/TTI terminal. If a coverage path is administered, the coverage for “busy” is used. If Enhanced Call Forwarding is administered, the “Busy” path is used.

Hot Desking Enhancement Station Lock

The enhanced handling of Station Lock is controlled by the **Hot Desking Enhancement Station Lock** field.

Valid entries	Usage
y/n	To enable the feature, select **y**. To disable the feature, select **n**. The default value is **n**. **x** is an invalid entry.
System	Hot Desking Enhancement (HDE) is controlled by the same named field on the System-Parameters Features screen.

For more information on Hot Desking Enhancement (HDE), see *Avaya Aura™ Communication Manager Feature Description and Implementation,* 555-245-205.

Enhanced PSA Location/Display Information Enabled

Appears when the **Terminal Translation Initialization (TTI) Enabled** field is **y**.

Valid entries	Usage
y	Enter **y**, if you want the system to display: ● PSA login and associated station information when a station is PSA associated. ● PSA logout and the port when a station is PSA dissociated.
n	Enter n if you do not want the system to display PSA information.

Terminal Translation Initialization (TTI) Enabled

For more information on TTI, see Terminal Translation Initialization in *Avaya Aura™ Communication Manager Feature Description and Implementation*, 555-245-205. You should contact your Avaya technical support representative before making changes to TTI settings.

Valid entries	Usage
y	Enter **y** to start ACTR, TTI, and PSA transactions (extension and telephone moves between ports). You can administer this field only if the **Terminal Trans. Init. (TTI)** field on the System Parameters Customer-Options (Optional Features) screen is **y**.
n	Enter **n** to remove existing TTI port translations and make sure no new TTI port translations are generated.

TTI Security Code

Appears when the **Terminal Translation Initialization (TTI) Enabled** field is **y**.

Valid entries	Usage
1 to 7 digits	Enter a one-digit to seven-digit number that a TTI user must use when the user accesses TTI from a telephone or data terminal. The system displays this field only when the **Terminal Translation Initialization (TTI)** field is **y**.

TTI State

Appears when the **Terminal Translation Initialization (TTI) Enabled** field is **y**. Enter the type of port translation that you want for the system to use for unadministered digital ports. The default is **voice**.

Valid entries	Usage
data	Enter **data**, if you want a stand-alone data module to the be the TTI port translation for your system. The activation and deactivation sequence is entered at data terminal.
resume	Enter **resume**, if you want TTI to be available after TTI has been manually suspended. The state of TTI returns to the state that it was in before TTI was manually suspended.

Valid entries	Usage
suspend	Enter **suspend** to make TTI voice or TTI data translations temporarily unavailable. The system does not remove existing TTI translations.
voice	Enter **voice**, if you want voice or voice/data terminal to be the TTI port translation for the system. The activation and deactivation sequence is entered from a telephone.

Unnamed Registrations and PSA for IP Telephones

Valid entries	Usage
y/n	Enter **y** to allow IP telephones to use the Personal Station Access (PSA) feature, and allow IP telephones to register into the state sometimes known as "PSA dissociated," "TTI unmerged," or "TTI state," but which is called "Unnamed Registered" in the H.323 standards.

EMU PARAMETERS

EMU Inactivity Interval for Deactivation (hours)

Use this field to administer a system-wide administrable interval for EMU deregistration at the visited switch. The allowable entries are the digits between 1 and 24 for hours, or blank. An entry of 1 means that after 1 hour of inactivity, the telephone is dropped from the visited home server. Where the entry is blank, the timer is not used and the visited station remains active until deregistration by another means occurs. This timer is applicable to inter and intra-Communication Manager EMU registrations.

Note:
If SES is enabled for your system, this field is used as the inactivity timer for SIP Visiting Users. For more information on SES and SIP telephones, see *SIP Support in Avaya Aura™ Communication Manager Running on Avaya S8xxx Servers,* 555-245-206.

Valid entries	Usage
1 to **24** or blank	Enter the interval, in hours, after which a visiting user is dropped due to inactivity. Default is **1**.

CALL PROCESSING OVERLOAD MITIGATION

Restrict Calls

Indicate the type of calls to block first during overload traffic conditions on the system.

Valid entries	Usage
stations-first	Deny new traffic generated by internal stations, allowing inbound calls only (works best in call center environments).
all-trunk-first	Deny all out-bound calls to trunks, tie-lines and stations, and all station-originated calls.
public-trunks-first	Deny all in-bound calls from trunks and tie-lines.

Field descriptions for page 4

Figure 94: Feature-Related System Parameters screen

```
 display system-parameters features                            Page 4 of x
                        FEATURE-RELATED SYSTEM PARAMETERS
          Reserved Slots for Attendant Priority Queue: 5
                           Time before Off-hook Alert: 10
               Emergency Access Redirection Extension:
 Number of Emergency Calls Allowed in Attendant Queue: 5
 Maximum Number of Digits for Directed Group Call Pickup:4
               Call Pickup on Intercom Calls? y        Call Pickup Alerting? n
                   Extended Group Call Pickup: none

  Deluxe Paging and Call Park Timeout to Originator? n
 Controlled Outward Restriction Intercept Treatment: tone
Controlled Termination Restriction (Do Not Disturb): tone
          Controlled Station to Station Restriction: tone
AUTHORIZATION CODE PARAMETERS         Authorization Code Enabled? y
                                       Authorization Code Length: 7
                            Authorization Code Cancellation Symbol: #
                                         Attendant Time Out Flag? n
                                      Display Authorization Code? y
               Controlled Toll Restriction Replaces: none
```

Authorization Code Enabled

Enables the Authorization Codes feature on a system-wide basis.

Valid entries	Usage
y	In the **Authorization Code Enabled** field, type **y**.

Authorization Code Length

Defines the length of the Authorization Codes users need to enter. To maximize the security of user's system, Avaya recommends to make each authorization code to the maximum length allowed by the system.

Valid entries	Usage
7	In the **Authorization Code Length** field, type **7**.

Authorization Code Cancellation Symbol

The # symbol a caller must dial to cancel the 10-second wait period during which the user can enter an authorization code.

Valid entries	Usage
#	In the **Authorization Code Cancellation Symbol** field, leave the default of #.

Attendant Time Out Flag

Attendant Time Out Flag indicates that a call is not to be routed to the attendant if a caller does not dial an authorization code within 10 seconds or dials an invalid authorization code.

Valid entries	Usage
n	In the **Attendant Time Out Flag** field, leave the default of **n**.

Display Authorization Code

Prevents the authorization code from displaying on telephone sets thus maximizing security.

Valid entries	Usage
n	In the **Display Authorization Code** field, type **n**.

Note:

These commands `Authorization Code Enabled, Authorization Code Length`, `Authorization Code Cancellation Symbol, Attendant Time Out Flag` & `Display Authorization Code` are available only when enabled in Figure 94: Feature-Related System Parameters screen on page 296.

Controlled Outward Restriction Intercept Treatment

Enter the type of intercept treatment the caller receives when the call is outward restricted.

Valid entries	Usage
announcement	Provides a recorded announcement to calls that cannot be completed as dialed. You select and record the message. The calling party receives indication that the call is receiving Intercept Treatment. Enter the extension number for the announcement in the associated field. **Note:** If entering a Multi-Location Dial Plan shortened extension, note the following: When entering a Multi-Location Dial Plan shortened extension in a field designed for announcement extensions, certain administration end validations that are normally performed on announcement extensions are not done, and resultant warnings or submittal denials do not occur. The shortened extensions also do not appear in any display or list that shows announcement extensions. Extra care should be taken to administer the correct type of announcement for the application when assigning shortened extensions.
attendant	Allows attendants to provide information and assistance to outgoing calls that cannot be completed as dialed or that are transferred to incomplete or restricted stations.
extension	Enter the extension number for the extension in an associated field. Cannot be a VDN extension.
tone	Provides a siren-type tone to internal calls that cannot be completed as dialed

Controlled Station-to-Station Restriction

Enter the type of intercept treatment the caller receives when the call is placed to a restricted telephone.

Valid entries	Usage
announcement	If announcement is entered, an associated extension number field displays. Enter the extension of the restricted telephone in the field.
attendant	Intercepted calls are redirected to the attendant.
extension (cannot be a VDN extension)	If extension is entered, an associated extension number field displays. Enter the extension of the restricted telephone in the field.
tone	Intercepted calls receive intercept (siren) tone.

Controlled Termination Restriction (Do Not Disturb)

Enter the type of intercept treatment the caller receives when the call is placed to a termination restricted telephone.

Valid entries	Usage
announcement	If **announcement** is entered, complete an associated extension number field.
attendant	Redirects intercepted calls to the attendant.
coverage	Redirects intercepted calls to coverage.
extension	If **extension** is entered, complete an associated extension number field. Cannot be a VDN extension,
tone	Provides a siren-type tone to calls that cannot be completed as dialed.

Deluxe Paging and Call Park Timeout to Originator

Paged calls that are to be parked require separate activation of the Call Park feature. All parked calls that time out return to the attendant.

Valid entries	Usage
y	Enter **y** to enable the Loudspeaker Paging - Deluxe feature that essentially integrates the Loudspeaker Paging and Call Park features. All parked calls that time out (not answered by paged party) return to the parking party.
n	Enter **n** to enable the Loudspeaker Paging feature.

Emergency Access Redirection Extension

Valid entries	Usage
An assigned extension	Enter the assigned extension number (can be a VDN) where emergency queue overflow redirects.

Number of Emergency Calls Allowed in Attendant Queue

Valid entries	Usage
0 to **75**	Enter the number of emergency calls allowed in the attendant queue before additional calls are routed to the backup extension.

Reserved Slots for Attendant Priority Queue

Valid entries	Usage
2 to **342**	Enter the number of calls that can go in to the emergency queue.

Time Before Off-Hook Alert

Valid entries	Usage
1 to **3000** seconds	Enter the time in seconds that a telephone with an Off-Hook Alert Class of Service can remain off-hook (after intercept tone has started) before an emergency call is sent to the attendant.

AUTHORIZATION CODE PARAMETERS

Attendant Time Out Flag

If this field is not enabled, the caller receives Intercept tone. This flag affects only remote users or incoming calls over trunks requiring an authorization code. This field only appears if **Authorization Codes Enabled** is **y**.

Valid entries	Usage
y/n	Enter **y** if a call is to be routed to the attendant if the caller does not dial an authorization code within 10 seconds or dials an invalid authorization code.

Authorization Code Cancellation Symbol

Enter the symbol a caller must dial to cancel the 10-second wait period during which the user can enter an authorization code. This field only appears when **Authorization Code** is **y**.

Valid entries	Usage
#	Enter the cancellation code **#** if the main and tandem servers/switches are both of the same type.
1	Enter the cancellation code **1** if an Avaya System 85 or DIMENSION PBX switch is part of the complex/network.

Authorization Code Length

This field only appears and must be completed if **Authorization Codes Enabled** is **y**. This is the number of digits that must be assigned to the **Authorization Code (AC)** field on the Authorization Code screen.

SECURITY ALERT:
You enhance your system's security by using the maximum length for your authorization code.

Valid entries	Usage
4 to 13 digits	Enter a number that defines the number of digits (length) in the **Authorization Code** field.

Authorization Codes Enabled

This field cannot be administered if Authorization Codes is not enabled on the System Parameters Customer-Options (Optional Features) screen.

SECURITY ALERT:
To maintain system security, Avaya recommends that Authorization Codes be used.

Valid entries	Usage
y/n	Enter **y** to enable Authorization Codes on a system-wide basis.

Controlled Toll Restriction Intercept Treatment

Appears when the **Controlled Toll Restriction Replaces** field is **outward** or **station-to-station**. This field applies an intercept treatment to a toll call during the call processing.

Valid entries	Usage
announcement	A sub-field appears to the right if **announcement** is used. If the entry is **announcement**, enter the assigned announcement extension.
attendant	Intercepted calls are redirected to the attendant.
extension	A sub-field appears to the right if **extension** is used. If the entry is **extension**, enter the extension assigned to station or individual attendant.
tone	Intercepted calls receive intercept (siren) tone.

Controlled Toll Restriction Replaces

This field activates the Controlled Toll Restriction feature.

Valid entries	Usage
outward **station-station** **none**	The value that you choose for this field is replaced by controlled toll restriction. In other words, if you choose station-station, you cannot use station-station restrictions unless you reset this field.

Display Authorization Code

This field applies only to DCP, not to BRI or hybrid sets.

SECURITY ALERT:
To enhance your system's security, set **Display Authorization Code** to **n**.

Valid entries	Usage
y	Enter **y** to allow authorization code digits to display on the set during the dialing.
n	Enter **n** if these digits should not display.

Field descriptions for page 5

Figure 95: Feature-Related System Parameters screen

```
change system-parameters features                                  page 5 of x
                          FEATURE-RELATED SYSTEM PARAMETERS

SYSTEM PRINTER PARAMETERS
  Endpoint: ____            Lines Per Page: 60       EIA Device Bit Rate:


SYSTEM-WIDE PARAMETERS
                                     Switch Name: ________________
            Emergency Extension Forwarding (min): 10
          Enable Inter-Gateway Alternate Routing? n
Enable Dial Plan Transparency in Survivable Mode? n
                              COR to Use for DPT:

MALICIOUS CALL TRACE PARAMETERS
               Apply MCT Warning Tone? n     MCT Voice Recorder Trunk Group: ___
      Delay Sending Release (seconds)?
SEND ALL CALLS OPTIONS
       Send All Calls Applies to: station    Auto Inspect on Send All Calls? n

UNIVERSAL CALL ID
     Create Universal Call ID (UCID)? n      UCID Network Node ID: ___
```

SYSTEM PRINTER PARAMETERS

The system printer is the printer dedicated to support scheduled reports.

EIA Device Bit Rate

This field does not appear for S87XX Servers

.

Valid entries	Usage
1200 **2400** **4800** **9600**	Enter the required printer speed setting.

Endpoint

Valid entries	Usage
Data module extension	Does not appear for S87XX Series IP-PNC. Associated with the System printer.
eia	Does not appear for S87XX Series IP-PNC. If the DCE jack is used to interface the printer.
SYS_PRNT	Use this value if the system printer is connected over a TCP/IP link, and the link is defined as SYS_PRNT on the IP Services screen.
blank	

Lines Per Page

Valid entries	Usage
24 to **132**	Enter the number of lines per page required for the report.

SYSTEM-WIDE PARAMETERS

COR to Use for DPT

Use this field to indicate the Class of Restriction to use for the Dial Plan Transparency feature (DPT).

Valid entries	Usage
station	This is the default. The FRL of the calling station determines whether that station is permitted to make a trunk call and if so, which trunk(s) it is eligible to access.
unrestricted	The first available trunk preference pointed to by ARS routing is used.

Emergency Extension Forwarding (min)

If an emergency call should drop (get disconnected), the public safety personnel attempts to call back. If the ELIN that was sent was not equivalent to the caller's extension number, the return call would ring some other set than the one that dialed 911. To overcome that limitation, you can automatically forward that return call to the set that placed the emergency call for an administered period of time.

This Emergency Extension Forwarding only applies if the emergency location extension number is an extension on the same PBX as the extension that dialed 911. Customers who have several PBXs in a campus should assign emergency location extensions accordingly.

This field sets the Emergency Extension Forwarding timer for all incoming trunk calls if an emergency call gets cut off (drops).

Valid entries	Usage
0 to **999**	Type a number between 0 and 999 that represents the time (in minutes) that an incoming trunk call forwards to the extension that made the initial 911 call. The default value for both new installs and upgrades is **10**.

Note:
If a user at the emergency location extension (the extension that made the initial 911 call) manually turns off the Call Forwarding feature, the feature is off no matter how many minutes might remain on the timer.

Enable Dial Plan Transparency in Survivable Mode

Use this field to enable/disable Dial Plan Transparency (DPT) without changing or removing other feature administration associated with DPT.

Valid entries	Usage
y/n	Enter **y** to enable the Dial Plan Transparency feature should a media gateway register with a local survivable processor (LSP), or a port network register with an Enterprise Survivable Server (ESS). Default is **n**.

Enable Inter-Gateway Alternate Routing

For more information on Inter-Gateway Alternate Routing, see *Administering Network Connectivity on Avaya Aura™ Communication Manager*, 555-233-504.

Valid entries	Usage
y/n	Enter **y** to enable the Inter-Gateway Alternate Routing feature. Default is **n**.

Switch Name

Valid entries	Usage
Any keyboard character	Enter up to 20 alpha-numeric characters for identification.

MALICIOUS CALL TRACE PARAMETERS

Apply MCT Warning Tone

Valid entries	Usage
y/n	Enter **y** to provide an audible tone to the controlling station when an MCT recorder is actively recording a malicious call.

Delay Sending Release (seconds)

This field specifies the amount of time DEFINITY waits before sending an ISDN release message in response to receiving an ISDN disconnect message. This field appears only if, on the System Parameters Customer-Options (Optional Features) screen, the **Malicious Call Trace** field is **y**.

Valid entries	Usage
0 to **30**	Enter the number in increments of 10.

MCT Voice Recorder Trunk Group

Assign the trunk group for MCT voice recorders.

Valid entries	Usage
1 to **666** or blank	group number for DEFINITY CSI
1 to **2000** or blank	group number for S87XX Series IP-PNC

SEND ALL CALLS OPTIONS

Auto Inspect on Send All Calls

Valid entries	Usage
y	If set to **y**, allows you to be presented automatically with Calling Party information for calls which are silently alerting their station because of the Send-All-Calls feature.
n	If set to **n**, you are not guaranteed a Calling Party display for calls sent directly to Coverage by the Send-All-Calls feature.

Send All Calls Applies to

Valid entries	Usage
station	If set to **station**, any call to that station, regardless of the number dialed, causes calls to that station's own extension to be sent immediately to Coverage, or causes calls to different extensions assigned to the station as bridged appearances to become Ring-Ping notification if **Redirect Notification** field is **y**.
extension	When set to **extension**, only the calls sent to that extension are placed to coverage.

UNIVERSAL CALL ID

Create Universal Call ID (UCID)

Valid entries	Usage
y	If set to **y**, DEFINITY generates UCID for each call when necessary.
n	If set to **n**, the DEFINITY does not generate a UCID for any call.

UCID Network Node ID

Enter a number unique to this server/switch in a network of switches.

Valid entries	Usage
1 to **32767** or blank	This number is an important part of the UCID tag and must be unique to the server/switch.

Field descriptions for page 6

Figure 96: Feature-Related System Parameters screen

```
change system-parameters features                                    page 6 of x
                          FEATURE-RELATED SYSTEM PARAMETERS
       Public Network Trunks on Conference Call: 5             Auto Start? n
    Conference Parties with Public Network Trunks: 6            Auto Hold? n
 Conference Parties without Public Network Trunks: 6       Attendant Tone? y
         Night Service Disconnect Timer (seconds): 180      Bridging Tone? n
                 Short Interdigit Timer (seconds): 3      Conference Tone? n
              Unanswered DID Call Timer (seconds): ____    Intrusion Tone? n
              Line Intercept Tone Timer (seconds): 30    Special Dial Tone? n
                 Long Hold Recall Timer (seconds):     Mode Code Interface? n
                      Reset Shift Timer (seconds): 0
      Station Call Transfer Recall Timer (seconds): 0      Recall from VDN? n
                               DID Busy Treatment: tone

               Allow AAR/ARS Access from DID/DIOD? _
                 Allow ANI Restriction on AAR/ARS? _
      Use Trunk COR for Outgoing Trunk Disconnect? _
                  7405ND Numeric Terminal Display? n           7434ND? n
DISTINCTIVE AUDIBLE ALERTING
             Internal: 1   External: 2   Priority: 3
                         Attendant Originated Calls:
    DTMF Tone Feedback Signal to VRU - Connection: _    Disconnection: _
```

7405ND Numeric Terminal Display

Valid entries	Usage
y/n	If enabled, this allows you to use 7405ND in the **Type** field of the Station screen. This is not an actual telephone type, but you can use this to define ports for certain types of voice messaging systems. This numeric display setting sends only numbers, and not names, to the messaging system.

7434ND

Valid entries	Usage
y/n	If enabled, this allows you to use 7434ND in the **Type** field of the Station screen. This is not an actual telephone type, but you can use this to define ports for certain types of messaging systems. Use this value if your voice messaging system operates in Bridged Mode.

Allow AAR/ARS Access from DID/DIOD

Valid entries	Usage
y/n	Enter **y** to allow calls for DID and DIOD type trunk groups to complete calls using ARS or AAR.

Allow ANI Restriction on AAR/ARS

(For Russia only) If a call is placed over a Russian shuttle trunk or a Russian rotary trunk via an AAR or ARS entry with the **ANI Req** field set to **r**, then ANI is requested just like a **y** entry. However, if the ANI request fails, the call immediately drops. All other trunk types treat the **r** entry as a **y**.

Valid entries	Usage
y	The **ANI Req** field on the AAR and ARS Digit Analysis Table and the AAR and ARS Digit Conversion Table permits the additional value of **r**(estricted).
n	The **ANI Req** field on the two screens takes only the current values of **n** and **y**.

Attendant Originated Calls

Valid entries	Usage
internal **external** **priority**	Indicate which type of ringing applies to attendant-originated calls.

Attendant Tone

Valid entries	Usage
y/n	Enter **y** to provide call progress tones to the attendants.

Auto Hold

Valid entries	Usage
y/n	Enter **y** to enable the Automatic Hold feature on a system-wide basis.

Auto Start

If this field is enabled, the **Start** buttons on all attendant consoles are disabled.

Valid entries	Usage
y/n	Enter **y** to enable the Automatic Start feature.

Bridging Tone

Valid entries	Usage
y/n	Enter **y** to apply a bridging tone when calls are bridged on primary extensions.

Conference Parties with Public Network Trunks

If the value of the **Public Network Trunks on Conference Call** field is **0**, this field does not appear on the screen.

Valid entries	Usage
3 to **6**	Specify the maximum number of parties allowed in a conference call involving a public network subscriber.

Conference Parties without Public Network Trunks

Valid entries	Usage
3 to **6**	Enter a number to specify the maximum number of parties allowed in a conference call involving no public network trunks.

Conference Tone

Note:
Bridging and Conference Tones are not supported by all countries. If these tones are enabled for countries other than Italy, Belgium, United Kingdom, or Australia, the tones are equivalent to no tone (silence) unless the tone is independently administered or customized on the Tone Generation screen.

Valid entries	Usage
y/n	Enter **y** to provide conference tone as long as three or more calls are in a conference call.

DID Busy Treatment

Specifies how to handle a direct inward dialing (DID) call to a busy station.

Valid entries	Usage
attendant	Call is routed to attendant.
tone	Caller hears a busy tone.

Intrusion Tone

Valid entries	Usage
y/n	Enter **y** to apply an intrusion tone (executive override) when an attendant intrudes on the call.

Line Intercept Tone Timer (seconds)

Valid entries	Usage
0 to **60**	Enter a number to specify how long an analog station user can wait after hearing warning tone without going on hook, before the station is placed in the lockout state.

Long Hold Recall Timer (seconds)

You can administer the system to remind a user that a call has been on hold for too long.

Valid entries	Usage
0 to **999**	Enter a number between **0** and **999**; **0** deactivates the timer. This value is the number of seconds a call can be on hold before the system re-alerts the user to remind them of the call.

Mode Code Interface

Note:
If you make a change to this field, you must log off and log back on to effect the permission changes to get to the Mode Code Related System Parameters on page 640.

Valid entries	Usage
y/n	A **y** entry allows you to use the Mode Code Voice Mail System Interface to connect the server running Communication Manager over a DTMF interface to INTUITY AUDIX or other vendors' voice-mail systems.

Night Service Disconnect Timer (seconds)

Valid entries	Usage
10 to **1024** or blank	Enter a number or blank to indicate how long a trunk call can be unanswered during night service before being disconnected. The trunk must not have Disconnect Supervision for this timer to apply.

Public Network Trunks on Conference Call

Indicates the number of public network trunks allowed on a conference call.

Valid entries	Usage
0 to **5**	If this field is **0**, the **Conference Parties with Public Network Trunks** field does not appear on the screen.

Recall from VDN

This feature is available when **Vectoring (Basic)** and **Vectoring (Prompting)** are set to **y**. Use this field to Indicate whether or not a call that is transferred to a VDN and then routed to a station is recalled to the originating station after the **Station Call Transfer Recall Timer** expires.

Valid entries	Usage
y	Calls are recalled from a VDN when the **Station Call Transfer Recall Timer** expires.
n	Calls are not recalled from a VDN when the **Station Call Transfer Recall Timer** expires.

Reset Shift Timer (seconds)

Used only for station-to-station calls or private network calls using ISDN trunks.

Valid entries	Usage
0 to **255**	Specifies the number of seconds that reset shift dial tone is audible before busy tone is heard. Reset shift dial tone allows the user to dial a new extension by dialing one new digit that replaces the last digit of the extension previously dialed. The new digit replaces the last digit of the extension previously dialed. Enter **0** to disable this feature.

Short Interdigit Timer (seconds)

Valid entries	Usage
3 to **9**	Enter a number to limit the time that digit analysis waits for the next digit when it has predicted that all the digits have already been collected.

Special Dial Tone

Special dial tone notifies an analog-telephone user if certain features are still active when the user goes off-hook. These features include:

- Call Forwarding
- Send All Calls
- Do Not Disturb

Valid entries	Usage
y/n	Enter **y** to use the Special Dial Tone. You must have a TN2182 circuit pack.

Station Call Transfer Recall Timer (seconds)

Allows a user-transferred call (station-to-station, a trunk call, or a DCS call) to re-terminate with priority ringing back to the station user who initiates the transfer operation if the transfer-to party does not answer the call within the administered Station Call Transfer Recall timer.

Valid entries	Usage
0 to **999**	Enter the time in seconds before a call redirects back to the station user who initiated the transfer operation. Enter **0** to disable this feature.

Unanswered DID Call Timer (seconds)

Enter number or blank to limit how long a DID call can remain unanswered before routing to the DID/TIE/ISDN Intercept Treatment feature. This timer interacts with the nonadministrable 50 second Wait for Answer Supervision Timer (WAST). The WAST timer overrides this field. Thus if this field is set to a value equal to or greater than 50 seconds, the caller receives intercept tone instead of the normal attendant or announcement treatment that is given when the Unanswered DID Call Timer expires before the WAST. If the Unanswered DID Call Timer expires while the DID call is being processed by call vectoring, the timer is ignored. See Wait Answer Supervision Timer in this section.

Valid entries	Usage
A number between **10** and **1024**	Enter a number to indicate how long a DID call can remain unanswered before routing to the DID/TIE/ISDN Intercept Treatment feature.
blank	Disables the timer.

Use Trunk COR for Outgoing Trunk Disconnect

Use this field to indicate whether the Outgoing Trunk Disconnect Timer is set based on the COR of the originating station or of the trunk group. If enabled, the timer is based on the COR of the trunk, not the originating caller's station.

Valid entries	Usage
n	Default. The Outgoing Trunk Disconnect Timer to be set based on the COR of the originating station. This is the default.
y	Enter **y** to enable the Outgoing Trunk Disconnect Timer to be set based on the COR of the trunk instead of the originating station.

DISTINCTIVE AUDIBLE ALERTING

The following Distinctive Audible Alerting fields appear when **Tenant Partitioning** on the System Parameters Customer Options screen is **n**. Use these fields to administer distinctive ring patterns for your system.

Attendant Originated Calls

This field appears when **Tenant Partitioning** on the System Parameters Customer Options screen is **n**.

Valid entries	Usage
internal **external** **priority**	Indicates which type of ringing (defined above) applies to attendant-originated calls. Default is **external**.

Distinctive Audible Alerting (Internal, External, Priority)

This field appears when **Tenant Partitioning** on the System Parameters Customer Options screen is **n**.

This is also known as Distinctive Ringing. Enter the number of rings for **Internal**, **External**, and **Priority** calls. For virtual stations, this applies to the mapped-to physical telephone. Defaults are as follows:

- **1**: Internal calls
- **2**:External and attendant calls
- **3**:Priority calls

Note:
SIP Enablement Services (SES) messaging includes the ring types internal, external, intercom, auto-callback, hold recall, transfer recall, or priority. In Communication Manager, types intercom, auto-callback, hold recall, and transfer recall are treated as priority.

Valid entries	Usage
1	1 burst, meaning one burst of ringing signal per period
2	2 bursts, meaning two bursts of ringing signal per period
3	3 bursts, meaning two bursts of ringing signal per period

DTMF Tone Feedback Signal to VRU - Connection, Disconnection

This field appears only if **DTMF Feedback Signals for VRU** on the System Parameters Customer-Options (Optional Features) screen is **y**.

Valid entries	Usage
0 to **9**, *, #, **A, B, C, D**	Enter the code to connect or disconnect the VRU. This can be a single digit, or a combination such as *99 to connect, #99 to disconnect. The tones must be programmed at the VRU as well.
blank	Blank means that no tone is to be sent to the VRU.

Field descriptions for page 7

Figure 97: Feature-Related System Parameters screen

```
change system-parameters features                              page 7 of x
                          FEATURE-RELATED SYSTEM PARAMETERS

CONFERENCE/TRANSFER

               Abort Transfer?                   No Dial Tone Conferencing?
        Transfer Upon Hang-Up?         Select Line Appearance Conferencing?
Abort Conference Upon Hang-Up?                                      Unhold?
   No Hold Conference Timeout:     Maximum Ports per Expanded Meet-me Conf:
                                    External Ringing for Calls with Trunks?
ANALOG BUSY AUTO CALLBACK
                Without Flash?                        Announcement:
                                         Voice Mail Hunt Group Ext:

AUDIX ONE-STEP RECORDING
                            Recording Delay Timer (msec):
Apply Ready Indication Tone To Which Parties In The Call?
  Interval For Applying Periodic Alerting Tone (seconds):
```

CONFERENCE/TRANSFER

Abort Conference Upon Hang-Up

Allows DCP, hybrid, IP, wireless, or ISDN-BRI telephone users to abort the conference operation when they hang up.

Valid entries	Usage
y/n	Enter **y** to change a call placed on soft-hold in the conference-pending status to hard-held status if the user hangs up.

Abort Transfer

Stops the transfer operation whenever a user presses a non-idle call appearance button in the middle of the transfer operation, or when they hang up. If both the **Abort Transfer** and **Transfer Upon Hang-Up** fields are **y** and you press the **transfer** button and then dial the complete transfer-to number, hanging up the telephone transfers the call. You must select another non-idle call appearance to abort the transfer. If the **Transfer Upon Hang-Up** field is **y**, hanging up completes the transfer. Requires DCP, Hybrid, IP, ISDN-BRI or wireless telephones.

Valid entries	Usage
y/n	Enter **y** to abort the transfer a call by pressing the **Transfer** button, dialing the desired extension, and then hanging up or selecting another non-idle call appearance. The user must press the **Transfer** button again to complete the process unless **Transfer Upon Hang-up** is also set to **y**.

External Ringing for Calls with Trunks

Use this field to specify existing ringing behavior or external call ringing behavior on external trunk calls that are transferred or conferenced by stations or Attendants, or extended by the Attendant to an "on-switch" extension.

Valid entries	Usage
y	Enter **y** to activate external ringing for transferred external trunk calls.
n	Enter **n** to use existing ringing behavior. This is the default value.

Maximum Ports per Expanded Meet-me Conf

This field allows you to administer the maximum number of conferees in an Expanded Meet-me Conference. This is a system-wide limit (that is, not administrable on a per Expanded-Meet-me VDN basis). This field is hidden if **Maximum Number of Expanded Meet-me Conference Ports** is **0** on the System Parameters Customer-Options **(Optional Features)** screen.

Valid entries	Usage
3 to **300**	Enter the maximum number of parties allowed for each conference on your system.

No Dial Tone Conferencing

When another line is on hold or alerting, No Dial Tone Conferencing eliminates dial tone while setting up a conference.

Valid entries	Usage
y/n	Enter **y** to activate No Dial Tone Conferencing.

No Hold Conference Timeout

Controls the timeout of No Hold Conference call setup. The system Answer Supervision timer should be set to a value less than this.

Valid entries	Usage
20 to **120**	Enter the number of seconds.

Select Line Appearance Conferencing

Use this field to specify that the user can use the line appearance rather than the **Conf** button to include a call in a conference. If a user is on a call, and another line is on hold or an incoming call alerts on another line, the user can press the **Conf** button to bridge the calls together. Using the select line appearance capability, the user can press a line appearance button to complete a conference instead of pressing the **Conf** button a second time.

Valid entries	Usage
y/n	Enter **y** to activate Select Line Appearance Conferencing

Transfer Upon Hang-Up

Using this feature, users can transfer a call by pressing the **Transfer** button, dialing the required extension, and then disconnecting.

Valid entries	Usage
y/n	Enter **y** so users can transfer a call by pressing the **Transfer** button, dialing the desired extension, and then disconnecting. The user can also wait to disconnect, speak with the other party, then press **Transfer** again to complete the process. With this field set to **y**, users of the Call Park FAC can park a call without having to press the **Transfer** button a second time. **NOTE:** This feature does not work for SIP endpoints. For SIP endpoints, the transferring user must join both the ends of the call.

Unhold

Allows the user to press the **hold** button on a telephone to release a hold (if no other line appearance is on hold or alerting). This does not apply to BRI telephones or attendant consoles.

Valid entries	Usage
y/n	Enter **y** to activate the unhold capability

ANALOG BUSY AUTO CALLBACK

With the Analog Busy Auto Callback Without Flash (ACB) feature enabled, when a caller places a call through an analog station, and the called station is busy and has no coverage path nor forwarding, then an announcement plays, announcing that the station is busy and prompting the caller to enter **1** for ACB or **2** to cover to a voice mail hunt group extension.

Announcement

Appears only if the **Without Flash** field is **y**.

Valid entries	Usage
Extension number	Enter the extension of the announcement you want to play for the ACB feature. This field cannot be left blank.

Voice Mail Hunt Group Ext

Appears only if the **Without Flash** field is **y**.

Valid entries	Usage
Extension number	Enter a voice mail hunt group extension to which the call is to be forwarded if the user enters 2 at the ACB announcement prompt.

Without Flash

Provides automatic callback for analog stations without flashing the hook. It is applied only when the called station is busy and has no other coverage path or call forwarding. The caller can enable the automatic callback without flashing the hook or entering the feature access code.

Note:
If the **Analog Busy Auto Callback Without Flash** field is set to **y**, the **Busy Auto Callback without Flash** field on the Station screen defaults to **y** (enabled) for all analog station types that allow Analog Auto Callback.

Valid entries	Usage
y/n	Enter **y** to provide automatic callback for a calling analog station without flashing the hook.

AUDIX ONE-STEP RECORDNG

On stations administered with this feature button, this feature allows users to activate and deactivate the recording of active calls to their Audix with the press of one button.

Apply Ready Indication Tone To Which Parties In The Call

Use this field to administer who hears the ready indication tone

Valid entry	Usage
all	All parties in the call hear the tone. The default value is all.
initiator	Only the initiator hears the tone. The initiator is the user who activates the Audix one-step recording feature.
none	None of the parties in the call hear the tone.

Note:
When the **Apply Ready Indication Tone To Which Parties In The Call** field is set to *initiator*, all parties in the call hear the ready indication tone. When the **Apply Ready Indication Tone To Which Parties In The Call** field is set to *all* and the **Interval For Applying Periodic Alerting Tone (seconds)** field is set to *0*, the parties in the call do not hear the ready indication tone and the alerting tone.

Interval For Applying Periodic Alerting Tone (seconds)

Appears only if the **Apply Ready Indication Tone To Which Parties In The Call** field is set to **all**.

Valid entries	Usage
0 to **60**	Enter a number from zero to 60 for the number of seconds desired between alerting tones, where zero disables the tone. The default value is a 15 second interval.

Recording Delay Timer (msecs)

Valid entries	Usage
0 to **4000** in increments of 100	Use this field to administer a delay interval before starting audix recording.

Field descriptions for page 8

Figure 98: Feature-Related System Parameters screen

```
change system-parameters features                                Page   8 of  x
                        FEATURE-RELATED SYSTEM PARAMETERS

ISDN PARAMETERS

 Send Non-ISDN Trunk Group Name as Connected Name? n
 Display Connected Name/Number for ISDN DCS Calls? n
       Send ISDN Trunk Group Name on Tandem Calls? n


                         QSIG/ETSI TSC Extension:
 MWI - Number of Digits Per Voice Mail Subscriber: 7

                             National CPN Prefix:
                        International CPN Prefix:
                               Pass Prefixed CPN: ASAI? n   VDN/Vector? n
    Unknown Numbers Considered Internal for AUDIX? n
             USNI Calling Name for Outgoing Calls? n
               Path Replacement with Measurements? y
                  QSIG Path Replacement Extension:
   Send QSIG Path Replacement Conf. Event to ASAI? y
            Path Replace While in Queue/Vectoring? n
```

ISDN PARAMETERS

Display Connected Name/Number for ISDN DCS Calls

Valid entries	Usage
y/n	Enter **y** to display the connected name/number (if received) for ISDN DCS calls.

Feature Plus Ext

Valid entries	Usage
A valid extension	Administration of this field is required for proper termination of some Feature Plus signaling. For example, Message Waiting Indication (MWI) requires this extension in order to send the indication to the appropriate server running Communication Manager. Appears only if the **ISDN Feature Plus** field is **y** on the System Parameters Customer-Options (Optional Features) screen.

International CPN Prefix

Allows you to apply prefixes to international calling numbers for display at receiving telephones. This is useful for those telephones that use or implement call back features based on incoming call numbers. When an ISDN-PRI call arrives, the incoming call setup is analyzed for: (1) whether the Type of Address (TOA) is national or international, and (2) whether the Numbering Plan Identifier (NPI) is Unknown or ISDN/Telephony. This administered prefix is applied to international calls. Prefixing applies to any subsequent display on the same server when the call is transferred, covered, or forwarded. The same prefixing applies to outgoing ISDN-PRI calls when the connected number information is returned and meets the same TOA and NPI criteria. The prefix plus the calling/connected number digit string is limited to 15 digits, with truncation occurring at the least significant digits.

Valid entries	Usage
1 to 5 digits, (**0** to **9**), * and # or blank	Enter a number that allows you to apply prefixes to international calling numbers for display.

Maximum Length

Appears only if the **Unknown Numbers Considered Internal for AUDIX** field is **y**. Indicates the maximum length of an unknown private number. Any unknown number longer than the administered value is considered external. This field cannot be blank when it appears.

Valid entries	Usage
1 to **20**	Enter a number for the maximum length of an unknown private number.

MWI - Number of Digits Per Voice Mail Subscriber

Appears only if the **Basic Supplementary Services** field or the **ISDN Feature Plus** field on the System Parameters Customer-Options (Optional Features) screen is **y**. This field provides an indication of the number of digits per AUDIX subscriber.

Note:
For QSIG-MWI, these routing digits and inserted digits must screen a digit string that, when analyzed and processed, routes to a Signaling Group supporting QSIG-TSCs. Once a QSIG TSC is established (from a message- center server/ switch to a served-user switch), then every lamp update message places the **Inserted Digits** field (from the Message Waiting Indication Subscriber Number Prefixes screen) in front of the AUDIX subscriber number to screen a complete QSIG network number for the served user.

For Feature Plus MWI, the routing digits and inserted digits must screen a digit string that routes over an SSF trunk to the Feature Plus extension on the remote (served user) switch. The **Inserted Digits** field must include the Feature Plus extension.

Valid entries	Usage
3 to **7**	Enter a value that corresponds to the digit string length of subscribers translated in the Message Center entity. For instance, if the Message Center entity is AUDIX, the value in this field must match the value of the **Extension Length** field on the Switch Interface Administration screen of AUDIX.

National CPN Prefix

Allows you to apply prefixes to national calling numbers for display at receiving telephones. This is useful for those telephones that use or implement call back features based on incoming call numbers. When an ISDN-PRI call arrives, the incoming call setup is analyzed for: (1) whether the Type of Address (TOA) is national or international, and (2) whether the Numbering Plan Identifier (NPI) is Unknown or ISDN/Telephony. This administered prefix is applied to national calls. Prefixing applies to any subsequent display on the same server when the call is transferred, covered, or forwarded. The same prefixing applies to outgoing ISDN-PRI calls when the connected number information is returned and meets the same TOA and NPI criteria. The prefix plus the calling/connected number digit string is limited to 15 digits, with truncation occurring at the least significant digits.

Valid entries	Usage
1 to 5 digits, (**0** to **9**), * and # or blank	Enter a number that allows you to apply prefixes to national calling numbers for display.

Pass Prefixed CPN to ASAI

Passes Calling Party Number information (CPN) to ASAI. The prefixed number is not passed on to other adjuncts, Call Detail Recording, or servers/switches.

Valid entries	Usage
y/n	Enter **y** to pass CPN information to ASAI.

Path Replacement While in Queue/Vectoring

Valid entries	Usage
y/n	Enter **y** to allow Path Replacement after queue/vector processing has started. Depending on the version of CMS you are using, some calls can go unrecorded if you enable this capability. Contact your Avaya technical support representative for more information.

Path Replacement with Measurements

Valid entries	Usage
y/n	Allows QSIG path replacement or DCS with Reroute to be attempted on measured calls.

QSIG Path Replacement Extension

Enter the extension for the system to use as part of the complete number sent in the Path Replacement Propose message.

Valid entries	Usage
Extension	Enter an unused extension that conforms to your dial plan.

QSIG/ETSI TSC Extension

Valid entries	Usage
Enter any valid, unassigned extension.	This is the phantom endpoint extension for QSIG Call Independent Signaling Connections (CISCs), which are similar to NCA Temporary Signaling Connections (TSCs) (both incoming and outgoing). ETSI protocol TSCs as well as QSIG TSCs are supported.

Send Custom Messages Through QSIG?

Valid entries	Usage
y/n	Enter **y** to provide appropriate display information, for example for the Posted Messages feature, over QSIG links.

Send ISDN Trunk Group Name on Tandem Calls

Valid entries	Usage
y/n	Enter **y** to provide consistent display information regardless of trunk type. If set to **y**, provides only trunk group name.

Send Non-ISDN Trunk Group Name as Connected Name

Valid entries	Usage
y/n	Enter **y** to send a name of the non-ISDN trunk group as the connected name when a call routes from ISDN to non-ISDN and the call is answered.

Unknown Numbers Considered Internal for AUDIX

Appears when, on the System Parameters Customer-Options (Optional Features) screen, either the **ISDN-PRI** or **ISDN-BRI Trunks** field is **y**. This field controls the treatment of an ISDN number whose numbering plan identification is "unknown" in a QSIG centralized AUDIX arrangement. This field works in conjunction with the Calling Party Number to INTUITY AUDIX field on the Hunt Group screen. The **Calling Party Number to INTUITY AUDIX** field on the Hunt Group screen must be **y** for this field to have an effect.

Valid entries	Usage
y	The unknown number is considered "internal" and AUDIX tries to find a calling party name match for the digit string. If a name match is found, AUDIX provides the calling party's name. If no name is found, AUDIX provides the calling party's telephone number.
n	The unknown number is considered "external" and AUDIX provides the calling party's telephone number.

USNI Calling Name for Outgoing Calls?

Valid entries	Usage
y/n	Enter **y** to send a name on outgoing calls over NI PRI trunks. Important: Be sure you have validated that your service provider's central office is capable of accepting calling name information from Communication Manager in this way. For example, if the central office has a 5ESS, it must be a generic 5EXX or later. Failure to validate the central office capability might cause the central office to drop outgoing calls from your Avaya S8XXX Server. In this case, change the value in this field to **n**. Enter **n** to prevent sending calling name information with outgoing calls over NI PRI trunks. **n** is this field overrides a **y** in the **Send Name** field of the outgoing Trunk Group screen.

PARAMETERS FOR CREATING QSIG SELECTION NUMBERS

Level 1 Code

Enter the first level regional code of the Avaya S8XXX Server in the network. Administer this field carefully. Communication Manager does not check to ensure you have entered a code that supports your entry in the **Network Level** field. You cannot enter anything in this field unless the **Network Level** field is set to **1** or **2**.

Valid entries	Usage
0 to **9**	Enter up to 5 digits.
blank	Because blank regional codes are valid, an entry is not required if the **Network Level** field is **1** or **2**.

In QSIG standards, this level 1 code is called the Level 0 Regional Code.

Level 2 Code

Enter the second level regional code of the Avaya S8XXX Server in the network. Administer this field carefully. The system does not check to ensure you have entered a code that supports your entry in the **Network Level** field. You cannot enter anything in this field unless the **Network Level** field is set to **2**.

Valid entries	Usage
0 to **9**	Enter up to 5 digits.
blank	Because blank regional codes are valid, an entry is not required if the **Network Level** field is 2.

In QSIG standards, this level 2 code is called the Level 1 Regional Code.

Network Level

Enter the value of the highest regional level employed by the PNP network. Use the following table to find the relationship between the network level and the Numbering Plan Identification/ Type of Number (NPI/TON) encoding used in the QSIG PartyNumber or the Calling Number and Connected Number IEs.

Valid entries	Usage
0	NPI - PNP TON - local
1	NPI - PNP TON - Regional Level 1
2	NPI - PNP TON - Regional Level 2
blank	If this field is blank and the **Send Calling Number** and/or **Send Connected Number** field is **y** or **r** with **private** specified for the **Numbering Format** field on the ISDN Trunk Group screen, the Calling Number and/or Connected Number IEs is not sent. If the field is left blank but **private** has been specified in the **Numbering Format** field on the ISDN Trunk Group screen, the Identification Number (PartyNumber data type) is sent for QSIG PartyNumbers encoded in ASN.1-defined APDUs. In this case, the ASN.1 data type containing the PartyNumber (PresentedAddressScreened, PresentedAddressUnscreened, PresentedNumberScreened, or PresentedNumberUnscreened) is sent marked as **PresentationRestricted** with **NULL** for the associated digits.

Field descriptions for page 9

Figure 99: Feature-Related System Parameters screen

```
change system-parameters features                                Page   9 of  x
                        FEATURE-RELATED SYSTEM PARAMETERS

CPN/ANI/ICLID PARAMETERS
   CPN/ANI/ICLID Replacement for Restricted Calls: restricted
  CPN/ANI/ICLID Replacement for Unavailable Calls: Unavailable

DISPLAY TEXT
    Identity When Bridging: station
     User Guidance Display? n

INTERNATIONAL CALL ROUTING PARAMETERS
                 Local Country Code:
          International Access Code:

ENBLOC DIALING PARAMETERS
   Enable Enbloc Dialing without ARS FAC? y      Minimum Digit Length: 4

CALLER ID ON CALL WAITING PARAMETERS
     Caller ID on Call Waiting Delay Timer (msec): 200
```

CPN/ANI/ICLID PARAMETERS

CPN/ANI/ICLID Replacement for Restricted Calls

Valid entries	Usage
up to 15 characters	Enter a text string to replace the restricted numbers on the display.

CPN/ANI/ICLID Replacement for Unavailable Calls

Valid entries	Usage
up to 15 characters	Enter a text string to replace the unavailable numbers on the display.

DISPLAY TEXT

Identity When Bridging

Use this field to determine whether the telephone display shows the literal identity of the bridged appearance or the virtual identity.

Note:
When you choose the **station** option, you must update the **Numbering — Public/Unknown Format** screen with the Extension Codes of the stations that display the caller's or answering party's assigned identification.

Valid entries	Usage
principal	The location from which the caller is bridging in. This is the default.
station	The caller's and the answering party's assigned identification.

User Guidance Display

Use this field to determine whether the telephone display shows user guidance messages.

Valid entries	Usage
y/n	Enter **y** to enable display of user guidance messages on the telephone. To disable the feature, select **n**. The default value is **n**. **x** is an invalid entry.

INTERNATIONAL CALL ROUTING PARAMETERS

Local Country Code

Valid entries	Usage
1 to 3 digits or blank	Enter a valid PSTN E.164 country code for this node. The default is blank (no SBS signaling trunk groups are administered). For example, for an SBS node in the United States, enter **1**. For a list of country codes, see the International Telecommunications Union "List of ITU-T Recommendation E.164 Assigned Country Codes .

International Access Code

Valid entries	Usage
1 to 5 digits or blank	Enter the access code required by the PSTN to route calls out of the country. This code is included with the telephone number received from the SBS terminating node if the Local Country Codes of the originating and terminating nodes are different. The default is blank (no access code is needed). For example, for an SBS node in the United States, enter **011**.

Note:
Once administered, these fields cannot be cleared until all trunk groups administered for SBS signaling have been removed. For details, see the **Trk Grp(s)** and **Signaling Group** screens. If the international call routing parameters are not administered on the **system-parameters features** screen and SBS is enabled on a trunk screen, a warning is displayed: `Must set INTERNATIONAL CALL ROUTING parameters on system-parameters features`.

ENBLOC DIALING PARAMETERS

Enable Enbloc Dialing without ARS FAC

Valid entries	Usage
y/n	Enter **y** to enable Enbloc Dialing without the need to dial a FAC. Default is **n**.

Minimum Digit Length

This field appears only when **Enable Enbloc Dialing without ARS FAC** is **y**.

Valid entries	Usage
1 to **20**	Enter the number of digits before Enbloc Calling Treatment is activated. Default is extension length plus 1.

CALLER ID ON CALL WAITING PARAMETERS

Caller ID on Call Waiting Delay Timer (msec)

Valid entries	Usage
5 to **1275** in increments of 5	Enter the desired delay in 5-millisecond intervals. Default is **200**.

Field descriptions for page 10

Figure 100: Feature-Related System Parameters screen

```
change system-parameters features                               Page  10 of  x
                        FEATURE-RELATED SYSTEM PARAMETERS

                 Pull Transfer: n            Update Transferred Ring Pattern? n
         Outpulse Without Tone? y              Wait Answer Supervision Timer? n
         Misoperation Alerting? n               Repetitive Call Waiting Tone? n
    Allow Conference via Flash? y
 Vector Disconnect Timer (min):       Network Feedback During Tone Detection? y
                                     System Updates Time On Station Displays? n

                    Station Tone Forward Disconnect: silence
                            Level Of Tone Detection: precise
          Charge Display Update Frequency (seconds): 30
                           Date Format on Terminals: mm/dd/yy
                        Onhook Dialing on Terminals? y
               Edit Dialing on 96xx H.323 Terminals? y
                  Allow Crisis Alert Across Tenants? y


ITALIAN DCS PROTOCOL
  Italian Protocol Enabled? y
   Apply Intercept Locally? y      Enforce PNT-to-PNT Restrictions? n
```

Allow Conference via Flash

Valid entries	Usage
y	Enter **y** to allow an analog station to use flash to conference calls together.
n	Enter **n** to prevent this.

Charge Display Update Frequency (seconds)

This applies only if you use Advice of Charge or Periodic Pulse Metering with display functions.

Valid entries	Usage
10 to **60** or blank	The amount of time (in seconds) between charge-display updates. Frequent display updates might have considerable performance impact. If the duration of a call is less than the Charge Display Update Frequency, the display will not automatically show charge information. To see charge information for a call, the user must have a **disp-chrg** button and must press the button before the call drops.

Date Format on 607/2400/4600/6400 Terminals

The format of the date as displayed on the telephones.

Valid entries	Usage
mm/dd/yy	month/day/year
dd/mm/yy	day/month/year
yy/mm/dd	year/month/day

Edit Dialing on 96xx H.323 Terminals

Edit Dialing feature allows an end-user to pre-dial a number when the telephone is on-hook.

Valid entries	Usage
y/n	To enable Edit Dialing select **y**. To disable the feature select **n**. The default value is **n**.

For more information on Edit Dialing, see *Avaya Aura™ Communication Manager Feature Description and Implementation,* 555-245-205.

Hear Zip Tone Following VOA?

This tone alerts a telephone user that the announcement has completed and a caller is now connected. CallMaster set and attendant console users hear double zip tone following the announcement. All other telephone users hear single zip tone.

Note:
This field does not effect auto-answer zip tone heard prior to the VOA.

Valid entries	Usage
y	Enter **y** to play zip tone following a VDN of Origin Announcement (VOA).
n	Enter **n** if you do not want zip tone following a VOA.

Intercept Treatment on Failed Trunk Transfers

Valid entries	Usage
y	Enter **y** to provide intercept treatment to calls failing trunk transfers.
n	Enter **n** to drop these calls.

Level of Tone Detection

For the most part, this option is no longer required in today's switching environment. It might be useful if your users are having difficulty placing outgoing calls due to inaccurate detection of network dial tone.

Valid entries	Usage
broadband	This is the least exact of the levels of tone detection. If Communication Manager detects any tone at all, it interprets this as dial tone.
medium	The server running Communication Manager interprets any tone which has a continuous "on" period of longer than 1 second as dial tone. Otherwise, the server accepts whatever the tone detector circuit pack reports.
precise	Communication Manager accepts whatever the tone detector circuit pack reports.

Misoperation Alerting

Misoperation Alerting should not be enabled if Call Prompting is optioned.

Valid entries	Usage
y	Enter **y** for misoperation recall alerting on multi-appearance stations, analog stations, and attendant consoles.
n	Enter **n** for standard misoperation handling without recall alerting.

Network Feedback During Tone Detection

Valid entries	Usage
y/n	Enter **y** to provide audible feedback to the user while the system attempts to detect dial tone.

On-hook Dialing on 607/2400/4600/6400/8400 Terminals

For 6400/8400, 607, 2420, 2410, and 4600 telephone users with speakerphones.

Valid entries	Usage
y/n	Enter **y** allows users to dial without lifting the handset. If you enable this, users hear dial tone when they press the **Speaker** button, even if the handset is on-hook.

The next four fields control station-to-switch recall signal timing. If a flashhook interval (recall window) is required, the upper and lower bounds of the interval can be administered. An on-hook that lasts for a period of time greater than or equal to the lower bound and less than or equal to the upper bound is treated as a recall flash. If an interval is not required, the **Disconnect Timing** value must be administered. An on-hook that lasts for a period of time less than this value is ignored; greater than or equal to this value is regarded as a disconnect. Regardless, an on-hook lasting 50 to 150 ms coming from a 2500-type set is always treated as a digit pulse unless **Ignore Rotary Digits** is **y** for that station.

Outpulse Without Tone

Valid entries	Usage
y	Enter **y** to indicate the server outpulse digits even when a dial tone has not been received.
n	Enter "**n**" if the calling party should receive intercept tone if no dial tone is detected.

Pull Transfer

Valid entries	Usage
y/n	Enter **y** to enable the Pull Transfer feature on a system-wide basis. This allows either the transferring or transferred-to party to press the **Transfer** button to complete the transfer operation

Repetitive Call Waiting Interval (sec)

This field appears when the **Repetitive Call Waiting Tone** field is **y**.

Valid entries	Usage
1 to **99**	Enter a number to specify the number of seconds between call waiting tones.

Repetitive Call Waiting Tone

Valid entries	Usage
y/n	Enter **y** to indicate that a repetitive call waiting tone be provided to the called party for all types of call waiting access.

Station Tone Forward Disconnect

Tone Forward Disconnect applies to any station other than one administered as a data endpoint, an attendant console, a BRI telephone, an auto answer, or as an Outgoing Call Management (OCM) agent.

Valid entries	Usage
busy **intercept** **silence**	When a station is the last party remaining off-hook on a call, that station receives the indicated tone or silence until that station is placed on-hook, or until the tone has played for 45 seconds and is followed by silence.

System Updates Time On Station Displays

This does not apply to telephones (such as BRI telephones) where the user sets the time.

Valid entries	Usage
y/n	Enter **y** to have the system automatically update the time on display telephones when background maintenance is run (for example, when the set is plugged in).

Update Transferred Ring Pattern

Valid entries	Usage
y/n	Enter **y** to change the ringing pattern from internal to external when an internal station transfers an external call. If most of your calls go through an attendant, you might want to set this to **y**, so your users can distinguish an external call.

Vector Disconnect Timer (min)

Enter the number of minutes, or blank that a trunk should remain connected to a vector.

Valid entries	Usage
1 to **240**	The number of minutes that you enter determines when the trunk is disconnected if the **Disconnect Supervision-In** or **Disconnect Supervision-Out** fields on the Trunk Group screen are **n**.
blank	Enter blank if you do not want Communication Manager to initiate a disconnect.

Wait Answer Supervision Timer

See Unanswered DID Call Timer (seconds) for more information.

Valid entries	Usage
y	Enter **y** to enable this feature on a system-wide basis. When **y** is entered in this field, calls to stations unanswered after 50 seconds are dropped.
n	When **n** is entered in this field, unanswered calls drop only when the calling party goes on-hook.

ITALIAN DCS PROTOCOL

The next three fields control the Italian DCS Protocol feature.

Apply Intercept Locally

This field appears if the **Italian Protocol Enabled** field is **y**.

Valid entries	Usage
y/n	Enter **y** to indicate that DID/CO intercept treatment is applied locally instead of on the originating server/switch.

Enforce PNT-to-PNT Restrictions

This field appears if the **Italian Protocol Enabled** field is **y**.

Valid entries	Usage
y/n	Enter **y** to indicate that restrictions and denial of PNT-to-PNT connections are enforced when the EDCS message is unavailable. A **y** in this field means restrictions are enforced.

Italian Protocol Enabled

Valid entries	Usage
y/n	Enter **y** to enable the Italian DCS feature on a system-wide basis.

Field descriptions for page 11

Figure 101: Feature-Related System Parameters screen

```
change system-parameters features                               page 11 of x
                        FEATURE-RELATED SYSTEM PARAMETERS

CALL CENTER SYSTEM PARAMETERS
  EAS
        Expert Agent Selection (EAS) Enabled? n
       Minimum Agent-LoginID Password Length:
         Direct Agent Announcement Extension: ________          Delay: ___
   Message Waiting Lamp Indicates Status For: station
VECTORING
                   Converse First Data Delay: 0       Second Data Delay: 2
              Converse Signaling Tone (msec): 100          Pause (msec): 70_
                    Prompting Timeout (secs): 10
                Interflow-qpos EWT Threshold: 2
   Reverse Star/Pound Digit For Collect Step? n
         Available Agent Adjustments for BSR? n
                           BSR Tie Strategy? 1st_found
  Store VDN Name in Station's Local Call Log? n

SERVICE OBSERVING
             Service Observing: Warning Tone? n      or Conference Tone? n
    Service Observing Allowed with Exclusion? n
            Allow Two Observers in Same Call? n
```

CALL CENTER SYSTEM PARAMETERS

EAS

Delay

Valid entries	Usage
0 to **99**	Only displays if **Expert Agent Selection (EAS)** or **ASAI** on the System Parameters Customer-Options (Optional Features) screen is **y**. Enter the number of seconds (**0** to **99**) the caller will hear ringback before the Direct Agent Announcement is heard by the calling party.

Direct Agent Announcement Delay

Only appears if **Expert Agent Selection (EAS)** or **ASAI Link Core Capabilities** on the System Parameters Customer-Options (Optional Features) screen is **y**.

Valid entries	Usage
0 to **99** or blank	Enter the number of seconds the caller hears ringback before the Direct Agent Announcement is heard by the calling party.

Direct Agent Announcement Extension

Valid entries	Usage
Valid extension	Enter the extension of the direct agent announcement.

Expert Agent Selection (EAS) Enabled

To enable this field, either no ACD or vectoring hunt groups might exist or, existing ACD or vectoring hunt groups must be "skilled." Only appears if **Expert Agent Selection (EAS)** on the System Parameters Customer-Options (Optional Features) screen is **y**.

Valid entries	Usage
y/n	Enter **y** to enable Expert Agent Selection.

Message Waiting Lamp Indicates Status For

Only appears if **Expert Agent Selection (EAS)** on the System Parameters Customer-Options (Optional Features) screen is **y**.

Valid entries	Usage
station	Since you only have one message waiting lamp on a telephone, you need to indicate if the message is for at the telephone extension or the loginID.
loginID	Expert Agent Selection (EAS) must be enabled to use this option.

Minimum Agent-LoginID Password Length

Enter the minimum number of digits that must be administered as an EAS Agent's LoginID password. Only appears if **Expert Agent Selection (EAS)** on the System Parameters Customer-Options (Optional Features) screen is **y**.

Valid entries	Usage
0 to **9**	Entering a **0** or blank indicates no password is required.

VECTORING

Available Agent Adjustments for BSR

Controls the use of BSR available agent adjustments. The **Vectoring (Best Service Routing)** field must be **y** on the System Parameters Customer-Options (Optional Features) screen.

Valid entries	Usage
y/n	Enter **y** to allow adjustments to available agents.

BSR Tie Strategy

This field appears only when **Vectoring (Best Service Routing)** on the **System Parameters Customer-Options (Optional Features)** screen is **y**.

Valid entries	Usage
1st-found	BSR uses the previously selected best choice as the best skill or location. This is the default setting.
alternate	Alternates the BSR selection algorithm when a tie in EWT or available agent criteria occurs. Every other time a tie occurs for calls from the same VDN, the consider step with the tie is selected to send the call instead of the first selected split, skill, or location. This helps balance the routing when the cost of routing remotely is not a concern.

Converse First Data Delay/Second Data Delay

The First Data Delay prevents data from being outpulsed (as a result of a converse vector step) from the system to CONVERSANT before CONVERSANT is ready. The delay commences when the CONVERSANT port answers the call. The Second Data Delay is used when two groups of digits are being outpulsed (as a result of a converse vector step) from the system to CONVERSANT. The Second Data Delay prevents the second set from being outpulsed before CONVERSANT is ready. The delay commences when the first group of digits has been outpulsed. Only appears if **Vectoring (Basic)** on the System Parameters Customer-Options (Optional Features) screen is **y**.

Valid entries	Usage
0 to **9**	Number of seconds for the delay.

Converse Signaling Tone/Pause

Only appears if **Vectoring (Basic)** and **DTMF** on the System Parameters Customer-Options (Optional Features) screen are **y**. In the **Signaling Tone** field, enter the length in milliseconds of the digit tone for digits being passed to the CONVERSANT. In the **Pause** field, enter the length in milliseconds of the delay between digits being passed. The optimum timer settings for the CONVERSANT or IR are 60 msec tone and 60 msec pause.

<table>
<tr><th>Valid entries</th><th>Usage</th></tr>
<tr><td>40 to 2550 (in increments of 10).</td><td rowspan="2">Values entered in the Tone/Pause fields are rounded up or down depending upon the type of circuit pack used to outpulse the digits.
• TN742B or later suffix analog board — Tone and pause round up or down to the nearest 25 msecs. For example, a 130 msec tone rounds down to 125 msecs, a 70 msec pause rounds up to 75 msec for a total of 200 msecs per tone.
• TN464F, TN767E or later suffix DS1 boards — Tone and pause round up to the nearest 20 msecs. For example, a 130 msec tone rounds up to 140 msecs, a 70 msec pause rounds up to 80 msecs for a total of 220 msecs per tone.
If a circuit pack has been used for end-to-end signalling to the CONVERSANT, and has then been used to send digits to a different destination, the CONVERSANT timers might stay in effect. To reset your timers to the system default, pull and reseat the circuit pack.</td></tr>
<tr><td>100</td></tr>
</table>

Interflow-qpos EWT Threshold

Displays only if, on the System Parameters Customer-Options (Optional Features) screen, the **Lookahead Interflow (LAI)** field is **y**. Part of enhanced Look-Ahead Interflow. Any calls predicted to be answered before this threshold will not be interflowed (therefore saving CPU resources).

Valid entries	Usage
0 to **9** or blank	Number of seconds for this threshold

Prompting Timeout (secs)

Only appears if **Vectoring (Prompting)** on the System Parameters Customer-Options (Optional Features) screen is **y**.

Valid entries	Usage
4 to **10**	Enter the number of seconds before the `Collect Digits` command times out for callers using rotary dialing.

Reverse Star/Pound Digit for Collect Step

The "*" is interpreted as a "caller end-of-dialing indicator and the "#" is an indicator to clear all digits previously entered by the caller for the current "collect" vector step.

Valid entries	Usage
y/n	Enter **y** to reverse the star and pound digits by the "collect" vector step. If set to **y**, it does not affect any other DEFINITY vector step or other non-ACD DEFINITY feature (such as ARS) in that the "*" and "#" digit-processing is unchanged.

Store VDN Name in Station's Local Call Log

Specifies if Communication Manager sends a message telling the telephone to store the VDN name or the calling party's name in the station call log for any of the following telephones:

- **2420**
- **4610**
- **4620**

- **4625**

Valid entries	Usage
y	Communication Manager sends a message telling the telephone to store the VDN name in the station call log.
n	Communication Manager sends a message telling the telephone to store the calling party's name in the station call log. This is the default setting.

SERVICE OBSERVING

Allow Two Observers in Same Call

Use this field to set, on a system-wide basis, the number of service observers allowed in a call to two.

Valid entries	Usage
y/n	When set to **y**, two service observers can monitor the same EAS Agent LoginID or station extension, and up to two service observers can be on the same two-party call or in a conferenced call having more than two parties.

Service Observing: Warning Tone

Service Observing (Basic) on the System Parameters Customer-Options (Optional Features) screen must be **y** before this field can be administered.

CAUTION:
The use of Service Observing features might be subject to federal, state, or local laws, rules or regulations or require the consent of one or both of the parties to the conversation. Customers should familiarize themselves and comply with all applicable laws, rules, and regulations before using these features.

Valid entries	Usage
y/n	Enter **y** to assign a warning tone to be given to telephone users and calling parties whenever their calls are being monitored using the Service Observing feature. This field cannot be set to **y** when **or Conference Tone?** is set to **y**.

or Conference Tone

Service Observing (Basic) on the System Parameters Customer-Options (Optional Features) screen must be **y** before this field can be administered.

Valid entries	Usage
y/n	Enter **y** to assign a conference tone to be given to telephone users and calling parties whenever their calls are being monitored using the Service Observing feature. This field cannot be set to **y** when **or Warning Tone?** is set to **y**.

Service Observing Allowed with Exclusion

Allows Service Observing of a station with Exclusion active, either by Class Of Service or by manual activation of Exclusion. Default is **n**.

Valid entries	Usage
y	Enter **y** to allow Service Observing of a station with Exclusion active, either by COS or by manual activation of Exclusion.
n	Observing towards a station with Exclusion active is denied, or if Exclusion is activated by a station while being observed, all bridged parties including the observer are dropped. This is the default.

Field descriptions for page 12

Figure 102: Feature-Related System Parameters screen

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

AGENT AND CALL SELECTION
                     MIA Across Splits or Skills? n
                      ACW Agents Considered Idle? y
                      Call Selection Measurement: current-wait-time
Service Level Supervisor Call Selection Override? y
                             Auto Reserve Agents:
        Copy ASAI UUI During Conference/Transfer?
ASAI
Call Classification After Answer Supervision? n      Send UCID to ASAI? n

CALL MANAGEMENT SYSTEM
                       REPORTING ADJUNCT RELEASE
                                  CMS <appl mis>: R13.1
                                   IQ <appl ccr>:

                 ACD Login Identification Length: 0
                           BCMS/VuStats LoginIDs?
               BCMS/VuStats Measurement Interval: hour
       BCMS/VuStats Abandon Call Timer (seconds):
                 Validate BCMS/VuStats Login IDs? n
                        Clear VuStats Shift Data: on-login
             Remove Inactive BCMS/VuStats Agents? n
```

AGENT AND CALL SELECTION

ACW Agents Considered Idle

Valid entries	Usage
y/n	Enter **y** to have agents who are in After Call Work included in the Most-Idle Agent queue. This means that ACW is counted as idle time. Enter **n** to exclude ACW agents from the queue.

Auto Reserve Agents

When a critical skill is not meeting its service level, auto-reserve puts agents in standby for their other skills to ensure that there is an available agent when the next call arrives for the critical skill. When an agent becomes available, all of his or her assigned skills are checked to see if any auto-reserve skills are not meeting their target service level. If so, the agent is made available only in those skills.

Valid entries	Usage
all	Puts an agent on stand-by for all skills.
none	Agent is not on stand-by for any additional skills.
secondary-only	Puts an agent on stand-by only for secondary skills.

Call Selection Measurement

This field determines how Communication Manager selects a call for an agent when the agent becomes available and there are calls in queue.

For information on Business Advocate, contact your Avaya representative or see the *Avaya Business Advocate User Guide*, 07-300653.

Valid entries	Usage
current-wait-time	**Current Wait Time** selects the oldest call waiting for any of the agent's skills.
predicted-wait-time .	**Predicted Wait Time** is a feature of Business Advocate.

Copy ASAI UUI During Conference/Transfer

Displays when, on the System Parameters Customer-Options (Optional Features) screen, either the **ASAI Interface** or **ASAI Proprietary Adjunct Links** field is **y**.

Valid entries	Usage
y/n	Enter **y** to copy user-to-user (UUI) information during a conference or transfer calls.

Note:
When this field is set to **y**, the system actually copies *all* UUI information, not just ASAI UUI. Copying only occurs during a human-initiated conference or transfer. Communication Manager does not copy the UUI if the conference or transfer is initiated by ASAI.

MIA Across Splits or Skills

Valid entries	Usage
y/n	Enter **y** to remove an agent from the MIA queue for all the splits/skills/hunt groups that they are available in when the agent answers a call from any of his or her splits/skills/hunt groups.

Service Level Maximizer Algorithm

This field displays only if **Service Level Maximizer** on the **System Parameters Customer-Options (Optional Features)** screen is **y**.

Valid entries	Usage
weighted	Use the WSL algorithm for Service Level Maximizer calculations.
actual	Use the ASL algorithm for Service Level Maximizer calculations. This is the default.

Service Level Supervisor Call Selection Override

This field determines whether Communication Manager changes agents' call handling preferences when a skill using Service Level Supervisor exceeds its Level 1 threshold.

For information on Business Advocate, contact your Avaya Account Executive or see the *Avaya Business Advocate User Guide*, 07-300653.

Valid entries	Usage
y	Enter **y** if you want to override the normal call handling preferences of a skill's assigned agents in this situation.
n	Enter **n** if you do not want to override agents' normal call handling preferences when the skill exceeds its Level 1 threshold. Service Level Supervisor requires Expert Agent Selection and Business Advocate.

ASAI

Call Classification After Answer Supervision?

For use with ASAI Outbound Call Management (OCM).

Valid entries	Usage
y/n	Enter **y** to force the server running Communication Manager to rely on the network to provide answer/busy/drop classification to the server. After the call has been answered, a call classifier can be added to perform answering machine, modem, and voice answering detection. The default value **n** always connects a classifier after call setup for determining call progress and answer. ISDN progress messages generally take precedence.

Send UCID to ASAI

Valid entries	Usage
y/n	Enter **y** to enable transmission of Universal Call ID (UCID) information to ASAI.

CALL MANAGEMENT SYSTEM

REPORTING ADJUNCT RELEASE

CMS (appl mis)

Valid entries	Usage
R12	CMS R12 is connected to the mis1 link, and to the mis2 link for a second CMS. The **IQ** field must be blank.
R13	CMS R13 is connected to the mis1 link, and to the mis2 link for a second CMS. The **IQ** field must be blank.

Valid entries	Usage
R13.1	CMS R13.1 is connected to the mis1 link, and to the mis2 link for a second CMS. Reporting adjuncts CMS, Avaya IQ, or both can be connected..
R14	CMS R14 is connected to the mis1 link, and to the mis2 link for a second CMS. Reporting adjuncts CMS, Avaya IQ, or both can be connected.
blank	A CMS system is not connected. This is the default. If any entry on the dial plan is set to greater than 7 digits, this field must be blank unless SA9062 is active and CMS is R14.1 or later. CMS only supports a maximum of 7 digits in reports.
R14.1	CMS R14.1 is connected to the mis1 link, and to the mis2 link for a second CMS. This release or later is required to activate the Special Application SA9062 to allow permissive use with Communication Manager Expanded Dial Plan (EDP-allowing extensions greater than 7 digits in the dial plan). This allows CMS to be connected with or without Avaya IQ. If any extensions grater than 7 digits are received by CMS, the left most digit in excess of 7 is deleted, leaving the right most 7 digits for tracking and reporting. Avaya IQ tracks and reports on the full EDP extensions.
R15	CMS R15 is connected to the mis1 link, and to the mis2 link for a second CMS.

IQ (appl ccr)

Valid entries	Usage
4.0, 5.0	Enter the release of the Avaya IQ system that is connected to the ccr1 link, and to the ccr2 link for a second Avaya IQ. EAS and UCID must be active before this screen is submitted for Avaya IQ connection. With Communication Manager 3.1, the IQ field does not appear and Avaya IQ is connected as a second CMS system with the CMS release field set to R13.1. With Communication Manager 4.x, 5.0, 5.1, or 5.2, either Avaya IQ Release 4.0 or 5.0 is valid depending on whether CMS is connected and the release of CMS. For full support with Communication Manager 5.2, Avaya IQ 5.0 is required. Reporting adjuncts CMS, Avaya IQ, or both can be connected. Only CMS R13.1 or later is allowed with Avaya IQ. When the CMS is set to R13.1, the Avaya IQ Add Resource Screen field for specifying the Communication Manager software release must be set to Communication Manager 3.1. With the CMS field set to R14 or R14.1, the Avaya IQ Add Resource Screen field for specifying the Communication Manager software release must be set to Communication Manager 4.0 or 5.0. With the CMS field set to R15, the Avaya IQ must be release 5.0 and the Add Resource Screen field for specifying the Communication Manager software release must be set to Communication Manager 5.2.
blank	An IQ system is not connected. This is the default.

OTHER CALL MANAGEMENT SYSTEM FIELDS

ACD Login Identification Length

Enter the number of digits for an ACD Agent Login ID if **Expert Agent Selection (EAS)** on the System Parameters Customer-Options (Optional Features) screen is **n**. If **BCMS/VuStats Login IDs** is **y**, the ACD Login ID length must be greater than 0. This field identifies an ACD agent to CMS. The number you enter in this field must equal the number of characters in the agent's login ID.

Valid entries	Usage
0 to **9**	For CMS, this field cannot be **0**.

BCMS/VuStats Abandon Call Timer (seconds)

Valid entries	Usage
1 to **10** or blank	Specify the number of seconds before calls are considered abandoned. Calls with talk time that is less than this number (and that are not held) are tracked by BCMS and displayed by VuStats as ABAND calls.

BCMS/VuStats LoginIDs

This feature can be used when EAS is not optioned, or in addition to EAS login IDs. When this field is **y**, both BCMS and CMS use the same login ID for an agent.

Valid entries	Usage
y/n	Enter **y** to administer valid agent login IDs to monitor call activity by agent.

BCMS/VuStats Measurement Interval

You can enter **half-hour** or **hour** for polling and reporting measurement data if the **BCMS (Basic)** and/or the **VuStats** on the System Parameters Customer-Options (Optional Features) screen is **y**.

Valid entries	Usage
half-hour **hour**	There are a maximum of 25 time slots available for measurement intervals. If **hour** is specified, an entire day of traffic information is available for history reports; otherwise, only half a day is available. This does not affect daily summaries as they always reflect traffic information for the entire day. The interval can be changed at any time, but does not go into effect until the current interval completes.

Clear VuStats Shift Data

Valid entries	Usage
on-login	Enter **on-login** to clear shift data for an agent when the agent logs in.
at-midnight	Enter **at-midnight** to clear shift data for all agents at midnight.

Remove Inactive BCMS/VuStats Agents

Valid entries	Usage
y	Agents are removed from reports when they have no staff time during the previous 7 days.
n	Agents remain on the report even if they have no staff time for any period of time.

Validate BCMS/VuStats Login IDs

Valid entries	Usage
y	Enter **y** to allow entry only of login-IDs that have been entered on the BCMS Login-ID screen.
n	Enter **n** to allow entry of any ACD login of the proper length.

Field descriptions for page 13

Figure 103: Feature-Related System Parameters screen

```
change system-parameters features                                Page  13 of  x
                        FEATURE-RELATED SYSTEM PARAMETERS

 CALL CENTER MISCELLANEOUS
                         Clear Callr-info: next-call
        Allow Ringer-off with Auto-Answer? n
          Service Level Algorithm for SLM: actual
    Reporting for PC Non-Predictive Calls? n


          Interruptible Aux Notification Timer (sec): 3
        Interruptible Aux Deactivation Threshold (%): 95

  ASAI

        Call Classification After Answer Supervision? n
                                   Send UCID to ASAI? n
```

CALL CENTER MISCELLANEOUS

Allow Ringer-off with Auto-Answer

Valid entries	Usage
y/n	Enter **y** to allow a user to use the **ringer-off** feature button to prevent ringing on EAS auto-answer calls.

Clear Callr-info

Use this field to specify when the collected digits Callr-Info display is to be removed from the agent/station display.

Valid entries	Usage
leave-ACW	Leaves the display up while the agent is in ACW (After-call work mode).
next-call	Clears the display when the next call is received. This is the default.
on-call-release	Clears the display on the 2nd line of a two-line display as soon as the call is released, either because of receiving call disconnect or the agent/ station user pressing the release button.

Interruptible Aux Notification Timer (sec)

Specifies the number of seconds the endpoint interruptible aux notifications, the flashing lamp, display, or tone, are on before an auto-in-interrupt or manual-in-interrupt agent is made available. This delay makes sure that an agent is not immediately made available when he presses an **interruptible aux** button. Also, this delay provides a brief period to an agent already in interruptible Aux mode before that agent is made available automatically. Valid entries are **1** to **9**. Default is **3**.

Field descriptions for page 14

Figure 104: Feature-Related System Parameters screen

```
change system-parameters features                               Page  14 of  x
                        FEATURE-RELATED SYSTEM PARAMETERS
 REASON CODES
                         Aux Work Reason Code Type: none
                           Logout Reason Code Type: none
                   Two-Digit Aux Work Reason Codes? n

 REDIRECTION ON IP CONNECTIVITY FAILURE
                Switch Hook Query Response Timeout:
       Auto-answer IP Failure Aux Work Reason Code: 0

 MAXIMUM AGENT OCCUPANCY PARAMETERS
                Maximum Agent Occupancy Percentage: 100
      Maximum Agent Occupancy Aux Work Reason Code: 9

 FORCED AGENT LOGOUT PARAMETERS
     Maximum Time Agent in ACW before Logout (sec):
                     ACW Forced Logout Reason Code: 0
              Clock Time Forced Logout Reason Code: 0
```

REASON CODES

Aux Work Reason Code Type

Valid entries	Usage
none	Enter **none** if you do not want an agent to enter a Reason Code when entering AUX work.
requested	Enter **requested** if you want an agent to enter a Reason Code when entering AUX mode but do not want to force the agent to do so. To enter **requested** the **Reason Codes** and **EAS** on the System Parameters Customer-Options (Optional Features) screen must be **y**.
forced	Enter **forced** to force an agent to enter a Reason Code when entering AUX mode. To enter **forced**, the **Reason Codes** and **EAS** on the System Parameters Customer-Options (Optional Features) screen must be **y**.

Logout Reason Code Type

Valid entries	Usage
none	Enter **none** if you do not want an agent to enter a Reason Code when logging out.
requested	Enter **requested** if you want an agent to enter a Reason Code when logging out but do not want to force the agent to do so. To enter **requested** the **Reason Codes** and **EAS** on the System Parameters Customer-Options (Optional Features) screen must be **y**.
forced	Enter **forced** to force an agent to enter a Reason Code when logging out. Enter **forced** to force an agent to enter a Reason Code when entering AUX mode. To enter **forced**, the **Reason Codes** and **EAS** on the System Parameters Customer-Options (Optional Features) screen must be **y**.

Two-Digit Aux Work Reason Codes

Valid entries	Usage
y/n	Enter **y** to enable two-digit Reason Codes for agent state changes to Aux Work. Default is **n**.

REDIRECTION ON IP CONNECTIVITY FAILURE

Switch Hook Query Response Timeout

Valid entries	Usage
500 to **5000** (msec)	Assign the time on a system basis that the call processing will wait for a response to the switch hook query before Return on IP Connectivity Failure (ROIF) is triggered. For details on selecting an appropriate timeout period, see *Avaya Aura™ Call Center 5.2 Automatic Call Distribution (ACD) Reference*, 07-602568
blank	ROIF is not active.

Auto-answer IP Failure AUX Reason Code

Valid entries	Usage
0 to **99**	Enter the reason code assigned for auto-answer IP failure, as the reason the agent was put into Aux Work.

MAXIMUM AGENT OCCUPANCY PARAMETERS

The Maximum Agent Occupancy (MAO) threshold is a system-administered value that is applied across all administered agents and is based on the total percentage of agent time in call service. MAO data is derived from the same calculations that are used to derive Least Occupied Agent (LOA).

When an agent who exceeds the specified MAO threshold attempts to become available, he or she is automatically placed in AUX mode for the reason code administered for this purpose. When the occupancy for such pending agents drops below the MAO, they are released from AUX mode and made available. To use MAO, Expert Agent Selection (EAS) must be enabled. For more information on MAO, see *Avaya Aura™ Call Center 5.2 Call Vectoring and Expert Agent Selection (EAS) Reference*, 07-600780.

Maximum Agent Occupancy Percentage

Valid entries	Usage
0 to **100**	Enter the percentage for MAO. Default is **100**.

Maximum Agent Occupancy AUX Reason Code

Valid entries	Usage
0 to **99**	Enter a reason code value. Default is **9**. A different reason code can be used for this purpose, but Avaya recommends that you do *not* use reason code 0.

FORCED AGENT LOGOUT PARAMETERS

Clock Time Forced Logout Reason Code

Valid entries	Usage
0 to **9**	Set the reason code for the Forced Agent Logout by Clock Time feature.

Maximum Time Agent in ACW before Logout (sec.)

This field is used for setting a maximum time the agent can be in ACW on a per system basis. You can only change the default if **Expert Agent Selection (EAS) enabled?** is set to **y** on the Feature-Related System Parameters screen, and the **Call Center Release** field on the System Parameters Customer-Options (Optional Features) screen is set to **3.0** or later. When this timer expires, the agent is logged out. This system option applies only to EAS configurations.

.

Valid entries	Usage
30 to **9999** or blank	Indicate the maximum time an agent can be in ACW before being automatically logged out. Default is blank, meaning no timeout.

ACW Forced Logout Reason Code

This field is used to specify the reason for logging out the agent due to timeout in ACW when the Reason Codes feature is active. You can only change the default if, on the System Parameters Customer-Options (Optional Features) screen, **Reason Codes** is set to **y**, and the **Call Center Release** field is set to 3.0 or later. Additionally, the **Expert Agent Selection (EAS) enabled?** field on the Feature-Related System Parameters screen must be set to **y**.

.

Valid entries	Usage
0 to **9**	Enter a reason code value. Default is **0**.

Field descriptions for page 15

Figure 105: Feature-Related System Parameters screen

```
change system-parameters features                              Page 15 of x
                        FEATURE-RELATED SYSTEM PARAMETERS

SPECIAL TONE
                                  Special Dial Tone? n
          Special Dial Tone for Digital/IP Stations: none

REDIRECTION NOTIFICATION
                           Display Notification for Do Not Disturb? n
                           Display Notification for Send All Calls? n
                             Display Notification for Call Forward? n
                    Display Notification for Enhanced Call Forward? y
                         Display Notification for a locked Station? n
         Display Notification for Limit Number of Concurrent Calls? n
                          Display Notification for Posted Messages? n
                                Scroll Status messages Timer(sec.):





Chained Call Forwarding? n
```

Chained Call Forwarding?

Valid entries	Usage
y/n	To enable chained call forwarding, select **y**. To disable chained call forwarding, select **n**. The default value is **n**.

SPECIAL TONE

Special Dial Tone

Valid entries	Usage
y/n	Enter **y** to enable an audible tone indicating the station is locked. Default is **n**.

Special Dial Tone for Digital / IP Stations

Valid entries	Usage
all none non-display	Specify the type of dial tone to be used for digital or IP stations.

REDIRECTION NOTIFICATION

Display Notification for Do Not Disturb?

Valid entries	Usage
y/n	To enable notification, select **y**. To disable notification, select **n**. The default value is **n**.

Display Notification for Send All Calls?

Valid entries	Usage
y/n	To enable notification, select **y**. To disable notification, select **n**. The default value is **n**.

Display Notification for Call Forward?

Valid entries	Usage
y/n	To enable notification, select **y**. To disable notification, select **n**. The default value is **n**.

Display Notification for Enhanced Call Forward?

Valid entries	Usage
y/n	To enable notification, select **y**. To disable notification, select **n**. The default value is **n**.

Display Notification for a locked Station?

Valid entries	Usage
y/n	To enable notification, select **y**. To disable notification, select **n**. The default value is **n**.

Display Notification for Limit Number of Concurrent Calls?

Valid entries	Usage
y/n	To enable notification, select **y**. To disable notification, select **n**. The default value is **n**.

Display Notification for Posted Messages?

Valid entries	Usage
y/n	To enable posting of messages, select **y**. To disable posting of messages, select **n**. The default value is **n**.

Scroll Status messages Timer (sec.)

Valid entries	Usage
5 to 10	The default value is **blank**, which deactivates the scrolling. The status information shows the feature that has the highest priority and is enabled. The features in order of decreasing priority are: – Do Not Disturb – Send All Calls – Call Forward – Posted Messages – LNCC – Station lock

Field descriptions for page 16

Figure 106: Feature-Related System Parameters screen

```
change system-parameters features                              Page 16 of x
                        FEATURE-RELATED SYSTEM PARAMETERS

AUTOMATIC EXCLUSION PARAMETERS

                       Automatic Exclusion by COS? y
                Automatic Exclusion Coverage/Hold? y
            Automatic Exclusion with Whisper Page? y
                              Recall Rotary Digit: 2
                    Password to Change COR by FAC: *
         Duration of Call Timer Display (seconds): 3

WIRELESS PARAMETERS
  Radio Controllers with Download Server Permission (enter board location)
    1.         2.         3.         4.         5.

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

AUTOMATIC EXCLUSION PARAMETERS

Automatic Exclusion by COS

Activates automatic exclusion automatically by class of service when a user goes off-hook on a station with an assigned **Exclusion** button. This works only for stations on the local server running Communication Manager.

Valid entries	Usage
y	Enables automatic exclusion by a class of service.
n	Exclusion operates normally.

Automatic Exclusion Coverage/Hold

Appears when the **Automatic Exclusion by COS** field is **y**.

Valid entries	Usage
y	The principal can bridge onto the call by pressing the appropriate bridged appearance button. And, if the coverage point places the exclusion call on hold, the principal can retrieve the call.
n	If a coverage point has answered a call and there is active exclusion on the call, the principal cannot bridge onto the call. And, if the coverage point places the exclusion call on hold, the principal cannot retrieve the call.

Automatic Exclusion with Whisper Page

Appears when the **Automatic Exclusion by COS** field is **y**.

Valid entries	Usage
y	The whisper page goes through to an excluded call.
n	The whisper page is denied when a station attempts to whisper page to a station that is on an excluded call.

Duration of Call Timer Display

Administer a call timer button on the Station screen.

Valid entries	Usage
3 to **30**	Enter the length of time (in 3 second increments) that the call information remains on display after the call is terminated.

Password to Change COR by FAC

Appears if, on the System Parameters Customer-Options (Optional Features) screen, the **Change COR by FAC** field is **y**. Avaya recommends using this password option.

Valid entries	Usage
4 to 8 digits	Requires the password option.
blank	Disables the password option.

Recall Rotary Digit

This establishes the digit to use for rotary telephones to receive recall dial tone. Dialing this digit simulates switch-hook flash so that users of rotary telephones can use features such as conference and transfer. The telephone must also be administered to use the recall rotary digit.

Valid entries	Usage
0 to **9**	Enter the digit users can dial to generate recall dial tone. Use a number that is not the first digit in normal dialing patterns.

WIRELESS PARAMETERS

Radio Controllers with Download Server Permission

Enter the necessary characters for the port location of the circuit pack containing the radio controllers with download server permission.

Valid entries	Usage
01 to **03** (DEFINITY CSI) or **1** to **64** (S87XX/S8300 Servers)	First and second characters are the cabinet number
A to **E**	Third character is the carrier
0 to **20**	Fourth and fifth characters are the slot number

IP PARAMETERS

Direct IP-IP Audio Connections

Allows direct audio connections between IP endpoints.

Valid entries	Usage
y/n	Enter to **y** to save on bandwidth resources and improve sound quality of voice over IP transmissions.

IP Audio Hairpinning

Allows IP endpoints to be connected through the IP circuit pack in the server.

Valid entries	Usage
y/n	Enter **y** to allow IP endpoints to be connected through the IP circuit pack in the Avaya S8XXX Server in IP format, without going through the Avaya DEFINITY TDM bus. Default is **n**.

RUSSIAN MULTI-FREQUENCY PACKET SIGNALING

Re-try

The **Re-try** field applies to outgoing Russian MFP trunks. It allows the server running Communication Manager to resend Russian MFP calling party number and dialed number information to the CO. The server resends the information only once over another outgoing trunk port of the same trunk group if Communication Manager receives a message that the information was received incorrectly by the CO. The switch also sends Russian MFP information over another trunk port if Communication Manager does not receive a timely response for the information.

Valid entries	Usage
y/n	Enter **y** to resend address information on outgoing Russian MFP trunks.

T2 (Backward Signal) Activation Timer (secs)

The **T2 (Backward Signal) Activation Timer (secs)** field applies to outgoing Russian MFP trunks. This field sets the number of seconds that Communication Manager waits for confirmation after sending calling party number and dialed number information on outgoing Russian MFP trunks

Valid entries	Usage
5 to **20**	Enter the number of seconds the system waits to receive confirmation after sending the address information on outgoing Russian MFP trunks.

Field descriptions for page 17

Figure 107: Feature-Related System Parameters screen

```
change system-parameters features                              page 17 of x
                       FEATURE-RELATED SYSTEM PARAMETERS

INTERCEPT TREATMENT PARAMETERS
Invalid Number Dialed Intercept Treatment: announcement  7700
Invalid Number Dialed Display: Invalid Number
Restricted Number Dialed Intercept Treatment: announcement  7701
Restricted Number Dialed Display: Restricted No.
Intercept Treatment On Failed Trunk Transfers? n

WHISPER PAGE
Whisper Page Tone Given To: all

6400/8400/2420J LINE APPEARANCE LED SETTINGS
                    Station Putting Call On Hold: green   wink
                   Station When Call is Active: green   solid
       Other Stations When Call Is Put On Hold:
            Other Stations When Call Is Active:
                                       Ringing:
                                          Idle:
         Display Information With Bridged Call?
                            Pickup On Transfer?
```

INTERCEPT TREATMENT PARAMETERS

Invalid Number Dialed Intercept Treatment

Enter the type of intercept treatment the end-user hears after dialing an invalid number.

Valid entries	Usage
announcement	Provides a recorded announcement when the end-user dials an invalid number. You select and record the message. Enter the extension number for the announcement in the associated field.
tone	Provides intercept tone when the end-user dials an invalid number. This is the default.

Invalid Number Dialed Display

This field shows a name in either Latin or Asian characters for an invalid number calling in.

Valid entries	Usage
Letters, spaces, numerals, and special characters.; maximum 15 characters	This field supports both a NAME1 and a NAME2 value. A NAME1 value directs the system to use the table of names that contains Latin characters, which can be displayed. Type a value of NAME2 to direct the system to use the UTF-8 table of names, which contains non-ASCII characters suitable for Asian language names.

Restricted Number Dialed Intercept Treatment

This field controls whether an announcement or an intercept tone is played when an end-user dials an number restricted from them due to COS, COR, or FRL restrictions. Enter the type of intercept treatment the end-user hears after dialing a restricted number.

Valid entries	Usage
tone	Provides intercept tone when the end-user dials an restricted number. This is the default.
announcement	Provides a recorded announcement when the end-user dials a restricted number. You select and record the message. Enter the extension number for the announcement in the associated field.

Restricted Number Dialed Display

This field controls whether the system displays any string of alphanumeric characters assigned for calls that are denied because of COS/COR, or FRL restrictions.

Valid entries	Usage
Letters, spaces, numerals, and special characters.; maximum 15 characters	This field supports both a NAME1 and a NAME2 value. A NAME1 value directs the system to use the table of names that contains Latin characters, which can be displayed. Type a value of NAME2 to direct the system to use the UTF-8 table of names, which contains non-ASCII characters suitable for Asian language names.

Intercept Treatment on Failed Trunk Transfers

Valid entries	Usage
y	Enter **y** to provide intercept treatment to calls failing trunk transfers.
n	Enter **n** to drop these calls.

WHISPER PAGE

Whisper Page Tone Given To

Use this field to indicate who should hear a Whisper Page.

Valid entries	Usage
all	All parties hear the Whisper Page.
paged	The whisper page feature sends a beep to the paging and the paged party.

DIGITAL STATION LINE APPEARANCE LED SETTINGS

WARNING:
The following fields only change the LED operation for 84xx and 64xx model telephones. When the LED operation is changed using any of these fields, then IP Agent and IP Softphone using a station type of 84xx or 64xx does not work. For station types other than 84xx or 64xx, a change to the LEDs using these fields does not affect either IP Agent or IP Softphone.

Note:
The system generates a warning if the default values of the LED Settings field are changed. The warning message states `WARNING: Avaya Softphone does not operate correctly if this value is changed`. You can see this warning message if you are running Communication Manager 3.1 or later.

Station Putting Call On Hold

Use this field to control the LED color and flash rate on the 8400 and 6400 series telephones for a call held on a Primary or Bridged Appearance. The LED for the color not selected is turned OFF. The default values are **green** and **wink**.

Valid entries	Usage
green or **red**	Indicate whether the LED is green or red.
off **wink** **inverse-wink** **flash** **flutter** **broken-flutter** **steady**	Select the flash rate for a call on hold.

Station When Call is Active

Use this field to control the red LED on the 8400 and 6400 series telephones, for a station active on a call. The default value is **steady**.

Valid entries	Usage
steady	When the value is steady, Communication Manager controls the red LED.
off	When the value is off, the red LED is always OFF.

Other Stations When Call Is Put On Hold

Use this field to control LED options for the other stations with a Bridged Appearance that has been placed on hold (for example, the user of this station has not pushed the hold button). The default values are **green** and **wink**.

Note:
This field is for a DCP bridged appearance LED color and flash rate when a call on a bridged appearance is put on hold by another party on the DCP bridged appearance. Additionally, this field only applies to 8400 and 6400 series telephones. The 2400 series telephone uses icons rather then LEDs. Correct operation in the Japanese environment requires the administrator to select the values **red** and **flash** for this field.

Valid entries	Usage
green or **red**	Indicate the color of the LED. Default is **green**.
off **wink** **inverse-wink** **flash** **flutter** **broken-flutter** **steady**	Select the flash rate for the LED. Default is **wink**.

Other Stations When Call Is Active

Use this field to control a DCP bridged appearance LED for those non-active parties with a bridged appearance that is active. The default value is **green**.

Note:
This field only applies to 8400 and 6400 series telephones. The 2400 series telephone uses ICONs rather then LEDs. Correct operation in the Japanese environment requires the administrator to select the value **red** for this field.

Valid entries	Usage
green or **red**	Select the LED color. Default is **green**.

Ringing

Use this field to control the LED color and flash rate while a call is ringing.

Note:
This field only applies to 8400 and 6400 series telephones. The 2400 series telephone uses icons rather then LEDs. Correct operation in the Japanese environment requires the administrator to select the values **red** and **wink** for this field. The default values are **green** and **flash**.

Valid entries	Usage
green or **red**	Indicate the LED color.
off **wink** **inverse-wink** **flash** **flutter** **broken-flutter** **steady**	Indicate the flash rate.

Idle

Use this field to control the LED of a station that is idle. The default value is **steady**.

Note:
This field only applies to 8400 and 6400 series telephones. The 2400 series telephone uses icons rather then LEDs. This value controls the red LED. Correct operation in the Japanese environment requires the administrator to select the value **off** for this field.

Valid entries	Usage
steady	LED is on. This is the default.
off	LED is off.

Display Information With Bridged Call

Use this field to control whether or not name and number for a bridged call are displayed on the telephone of the called party. A **y** entry indicates that the information is to be displayed; this field does not in any way control the content of the display.

Valid entries	Usage
y/n	Type **y** to display the name and number for an incoming call to the bridged appearance. Default value is **n**.

Pickup on Transfer

Valid entries	Usage
y	Enter **y** to allow bridged appearances of a station to pick up a call on hold because of a transfer.
n	Bridged appearances of another station are NOT allowed to pick up a call on hold because of a transfer.

Field descriptions for page 18

Figure 108: Feature-Related System Parameters screen

```
change system-parameters features                                Page  18 of  18
                        FEATURE-RELATED SYSTEM PARAMETERS

IP PARAMETERS

                   Direct IP-IP Audio Connections? y
                             IP Audio Hairpinning? n


CALL PICKUP
  Maximum Number of Digits for Directed Group Call Pickup: 4
                   Call Pickup on Intercom Calls? y     Call Pickup Alerting? n
     Temporary Bridged Appearance on Call Pickup? y     Directed Call Pickup? n
                      Extended Group Call Pickup: none
                   Enhanced Call Pickup Alerting? y
 Enhanced Call Pickup Delay Timer (sec.) Display: 5     Audible Notification: 5

                        Display Information With Bridged Call? n
  Keep Bridged Information on Multiline Displays During Calls? y
                  PIN Checking for Private Calls? n
```

IP PARAMETERS

Direct IP-IP Audio Connections

Allows direct audio connections between H.323 endpoints. For SIP Enablement Services (SES) trunk groups, this is the value that allows direct audio connections between SES endpoints.

Valid entries	Usage
y/n	Enter to **y** to allow direct connections.

IP Audio Hairpinning?

Allows IP endpoints to be connected through the IP circuit pack in the server.

Valid entries	Usage
y/n	Enter **y** to allow IP endpoints to be connected through the Avaya S8XXX Server's IP circuit pack in IP format, without first going through the Avaya DEFINITY TDM bus. Default is **n**.

PIN Checking for Private Calls

The PIN Checking for Private Calls feature now has a new field added to system-parameters features screen as shown in Figure 108: Feature-Related System Parameters screen on page 377 that allows the user to enable the feature.

Valid entries	Usage
y/n	To enable PIN Checking for Private Calls select **y**. To disable the feature select **n**. The default value is **n**.

CALL PICKUP

Maximum Number of Digits for Directed Group Call Pickup

Valid entries	Usage
1 to **4**	The maximum number of digits are accepted for the pickup group number. The pickup group number is considered complete when you enter the maximum number of digits or enter the # symbol. Default is **4**. For more information, see the **Directed Group Call Pickup Access Code** field on Feature Access Code (FAC) screen.

Call Pickup Alerting

This provides pickup group members with a visual indication on the Call Pickup status lamp of calls eligible to be answered via Call Pickup

.

Valid entries	Usage
y/n	Enter **y** to enable Call Pickup Alerting on a system-wide basis.

Call Pickup on Intercom Calls

Valid entries	Usage
y	Enter **y** to allow a user's or Agent LoginID's call, ringing as an intercom call, to be picked up using the Call Pickup or Directed Call Pickup features. This field controls the use of this feature throughout the system.
n	Enter **n** to prevent the use of these features to pickup an intercom call.

Directed Call Pickup

Feature use by individual stations, attendants, or EAS agents can be controlled by COR.

Valid entries	Usage
y	Enter **y** to allow use of the Directed Call Pickup feature across the system.
n	Enter **n** to prevent feature use.

Extended Group Call Pickup

Enables call pickup groups to answer calls directed to another call pickup group.

Valid entries	Usage
flexible	Flexible feature version supporting a one-to-n (pickup group-to-extended pickup group) mapping.
simple	Simple feature version with a one-to-one pickup group-to-extended pickup group mapping supported.
none	Extended group call pickup not supported.

Temporary Bridged Appearance on Call Pickup

Valid entries	Usage
y	Enter **y** to allow a temporary bridged appearance for calls answered with the Call Pickup or Directed Call Pickup features. This field controls this capability on a system-wide basis.
n	Enter **n** to prevent the temporary bridged appearance of calls answered with these features.

Enhanced Call Pickup Alerting

Valid entries	Usage
y/n	To enable the feature, select **y**. To disable the feature, select **n**. The default value is **n**.

Enhanced Call Pickup Delay Timer (sec.) Display

Valid entries	Usage
1 to 15	Enter the time in seconds, any numeric value in the range 1 to 15.

Enhanced Call Pickup Delay Timer (sec.)

Valid entries	Usage
1 to 15	Enter the time in seconds, any numeric value in the range 1 to 15.

Audible Notification

Valid entries	Usage
y/n	Enter the time in seconds, any numeric value in the range 1 to 15.

Firmware Station Download

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

Field descriptions for page 1

Figure 109: Firmware Station Download screen

```
 change firmware station-download                                  page 1 of x

                              FIRMWARE STATION DOWNLOAD

Source File:

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

Beginning Station:                  Ending Station:
```

Source File

Valid entries	Usage
up to 32 alphanumeric characters	Display only field. This field displays the name of the file specified on the TFTP Server Configuration screen, and which exists in system memory.

Schedule Download

Valid entries	Usage
y/n	Enter **y** to schedule a time for firmware download to multiple DCP stations.

Start Date/Time

This field appears only when the **Schedule Download** field is set to **y**.

Valid entries	Usage
mm, dd, yyyy; hh, mm	Enter the month, day, year, and time at which you want the firmware download to begin.

Stop Date/Time

This field appears only when the **Schedule Download** field is set to **y**.

Valid entries	Usage
mm, dd, yyyy; hh, mm	Enter the month, day, year, and time at which you want the firmware download to end.

Continue Daily Until Completed

Valid entries	Usage
y/n	Enter **y** if you want the system to execute the firmware download each day at the scheduled time until all specified telephones have received the firmware.

Download Set Type

Valid entries	Usage
2410 DCP **2420 DCP**	Display only. Indicates the set type (2410 or 2420) of DCP telephones to which firmware is to be downloaded.

Beginning Station

Valid entries	Usage
up to 8 digits	Enter the first extension number in the range of telephones to which you want to download the firmware. Up to 1000 stations can be included in a scheduled download.

Ending Station

Valid entries	Usage
up to 8 digits	Enter the last extension number in the range of telephones to which you want to download firmware. Up to 1000 stations can be included in a scheduled download.

Group Paging Using Speakerphone

Use this screen to assign digital speakerphones to a paging group. Users can page all the telephones in the group simultaneously by dialing the group's extension.

Field descriptions for page 1

Figure 110: Group Paging Using Speakerphone screen

```
change group-page 1                                              Page 1 of x
                          GROUP PAGING USING SPEAKERPHONE

    Group Number: 1                            Group Extension: 1234567890123
      Group Name:                                          COR: 1
GROUP MEMBER ASSIGNMENTS                                    TN: 1
  Extension       Name                    Extension       Name
  1234567890123   12345678901234567890    1234567890123   12345678901234567890

 1: 41752           x41752 4a1823 24port 17:
 2: 41153                                18:
 3: 41750    st2 4a1802                  19:
 4: 41529    Prince Charles              20:
 5: 41527    x41752 port 4a1805          21:
 6: 41706    EXT 41706                   22:
 7: 41534    Thunder x41534              23:
 8:                                      24:
 9:                                      25:
10:                                      26:
11:                                      27:
12:                                      28:
13:                                      29:
14:                                      30:
15:                                      31:
16:                                      32:
```

COR

Valid entries	Usage
0 to **995**	Enter a class of restriction. In order to page the group, users' class of restriction must give them calling permission for the group's class of restriction.

Ext

Valid entries	Usage
An extension number	Assign a telephone to the group by entering its extension number in this field.

Group Extension

Valid entries	Usage
An extension number	Assign the extension users dial to page the members of this group.

Group Name

Valid entries	Usage
1 to 27 characters	Enter a name that's informative to users, because it appears on callers' telephone displays when they page the group. **NOTE:** Avaya BRI stations support ASCII characters only. They do not support non-ASCII characters, such as Eurofont or Kanafont. Therefore, if you use non-ASCII characters in any Communication Manager **Name** field, such characters do not display correctly on a BRI station.

Group Number

This field displays the identifying number the server running Communication Manager assigns to the group when it is created.

Name

When you save your changes, Communication Manager fills in this display field with the name assigned to each extension on the Station screen.

Note:
Avaya BRI stations support ASCII characters only. They do not support non-ASCII characters, such as Eurofont or Kanafont. Therefore, if you use non-ASCII characters in any Communication Manager **Name** field, such characters do not display correctly on a BRI station.

TN

This field allows group paging to be partitioned by tenant. Enter the tenant number for this paging group.

Related topics

See Group Paging in *Avaya Aura™ Communication Manager Feature Description and Implementation,* 555-245-205, for more information.

Holiday Table

Use this screen to define individual holidays or holiday ranges.

Field descriptions for page 1

Figure 111: Holiday Table screen

```
change holiday table x                                         Page   1 of x
                                    HOLIDAY TABLE

Number:  1                    Name:

______START______      ______END________
Month Day Hour Min     Month Day Hour Min     Description
__    __  __   __      __    __  __   __      _____________________________
__    __  __   __      __    __  __   __      _____________________________
__    __  __   __      __    __  __   __      _____________________________
__    __  __   __      __    __  __   __      _____________________________
__    __  __   __      __    __  __   __      _____________________________
__    __  __   __      __    __  __   __      _____________________________
__    __  __   __      __    __  __   __      _____________________________
__    __  __   __      __    __  __   __      _____________________________
```

Description

Valid entries	Usage
Up to 27 characters.	Enter a phrase to describe the holiday.

End Day

Valid entries	Usage
1 to **31**	Enter the ending day of the holiday.

End Hour

Valid entries	Usage
0 to **23**	Enter the ending hour of the holiday using a 24-hour clock.

End Min

Valid entries	Usage
0 to **59**	Enter the ending minute of the holiday.

End Month

Valid entries	Usage
1 to **12**	Enter the ending month of the holiday.

Name

Display-only field identifying the name of the table.

Valid entries	Usage
Up to 27 characters	Description of the holiday table.

Number

Display-only field identifying the holiday table number.

Valid entries	Usage
1 to **10**	Holiday table number.

Start Day

Valid entries	Usage
1 to **31**	Enter the starting day of the holiday.

Start Hour

Valid entries	Usage
0 to **23**	Enter the starting hour of the holiday using a 24-hour clock.

Start Min

Valid entries	Usage
0 to **59**	Enter the starting minute of the holiday.

Start Month

Valid entries	Usage
1 to **12**	Enter the starting month of the holiday.

Hospitality

This screen is used to implement the system parameters associated with the hospitality features. To use and administer the Hospitality-related features, **Hospitality** must be **y** on the System Parameters Customer-Options (Optional Features) screen. Contact your Avaya representative for assistance.

Field descriptions for page 1

Figure 112: Hospitality screen

```
change system-parameters hospitality                                Page   1 of   x
                              HOSPITALITY

                       Message Waiting Configuration: act-nopms
               Controlled Restrictions Configuration: act-nopms
               Housekeeper Information Configuration: act-nopms
                     Number of Housekeeper ID Digits: 0
                                    PMS Log Endpoint:
                           Journal/Schedule Endpoint:
             Client Room Coverage Path Configuration: act-nopms
              Default Coverage Path for Client Rooms:
             Forward PMS Messages to Intuity Lodging? n

                              PMS LINK PARAMETERS
                                       PMS Endpoint:
          PMS Protocol Mode: transparent ASCII mode? n
               Seconds before PMS Link Idle Timeout: 20
Milliseconds before PMS Link Acknowledgment Timeout: 500
                   PMS Link Maximum Retransmissions: 3
           PMS Link Maximum Retransmission Requests: 3
                   Take Down Link for Lost Messages? y
```

Client Room Coverage Path Configuration

This indicates whether the server and the Property Management System (PMS) exchange coverage path information for guest stations.

Valid entries	Usage
act-nopms	The message is acknowledged (MESSAGE ACK), but no action is taken.
act-pms	If active (**act-pms**), the server and PMS exchange and accept coverage path information. This field does not apply to normal mode. When upgrading from a release that does not support this feature, the field is set to **act-pms** if the PMS protocol mode is administered for transparent or ASCII mode.

Controlled Restrictions Configuration

This indicates whether controlled restriction information is being exchanged between the server and the PMS.

Valid entries	Usage
act-nopms	The message is acknowledged (MESSAGE ACK), but no action is taken.
act-pms	The server and the PMS exchange and accept controlled restriction information.

Default Coverage Path for Client Rooms

This applies only to stations with a "client room" class of service in the "occupied" mode. This field is used for transparent or ASCII mode. The value in this field is also used during a translation save as the coverage path for each station with "client room" class of service.

Valid entries	Usage
1 to **9999** or blank	Enter the coverage path assigned when the server receives a check-out message for a valid extension or a new check-in.

Forward PMS Message to INTUITY Lodging

This field is used only in ASCII mode.

Valid entries	Usage
y	PMS-to-INTUITY messages are sent through the server.
n	PMS-to-INTUITY messages are sent directly to the Avaya INTUITY Lodging system.

Housekeeper Information Configuration

This indicates whether housekeeper information is being exchanged between the server and the PMS.

Valid entries	Usage
act-nopms	The message is acknowledged (MESSAGE ACK), but no action is taken.
act-pms	If active (**act-pms**), the server and PMS exchange and accept housekeeper information.

Journal/Schedule Endpoint

This is a valid data extension number that is assigned to the data module connected to the Journal/Schedule printer.

Valid entries	Usage
Valid data extension number	Cannot be a VDN extension. This extension can be the same as the PMS/Log printer and both sets of reports can be printed on the same printer. This extension is dialed by the server to send journal information or schedule reports to the printer.
PMS_LOG	Use this value if the printer is connected over a TCP/IP link, and this link is defined as PMS_LOG on the IP Services screen.
PMS_JOURNAL	Use this value if the printer is connected over a TCP/IP link, and this link is defined as PMS_JOURNAL on the IP Services screen.
blank	

Message Waiting Configuration

This indicates whether message waiting notification requests and changes are being exchanged between the server and the PMS.

Valid entries	Usage
act-nopms	The message is acknowledged (MESSAGE ACK), but no action is taken.
act-pms	Message waiting is active on the server and information between the PMS and server is being transmitted.

Number of Housekeeper ID Digits

Valid entries	Usage
0 to **6**	Enter the number of digits that the housekeeper must dial for identification.

PMS Log Endpoint

This is a valid data extension number that is assigned to the data module connected to the PMS/Log printer.

Valid entries	Usage
Valid data extension	Cannot be a VDN extension. This extension is dialed by the server to send housekeeping and PMS events to the printer.
PMS_LOG	Use this value if the printer is connected over a TCP/IP link, and this link is defined as PMS_LOG on the IP Services screen.
PMS_JOURNAL	Use this value if the printer is connected over a TCP/IP link, and this link is defined as PMS_JOURNAL on the IP Services screen.
blank	

PMS LINK PARAMETERS

ASCII mode

The **PMS Protocol Mode** field must be **transparent**.

Valid entries	Usage
y/n	Enter **y** when the ASCII-only mode is being used for the PMS message set.

Milliseconds Before PMS Link Acknowledgment Timeout

This regulates how quickly the system responds to a message from the PMS (also known as "pace timing"). This value is also used as the "inquiry message" (ENQ) time-out value. In most cases, keep this as short as possible.

Valid entries	Usage
100 to **1500**	Enter the time in milliseconds the system waits for an acknowledgment from the PMS indicating it correctly received a message.

PMS Link Maximum Retransmission Requests

Valid entries	Usage
1 to **5**	Enter the number of times that the server allows the PMS to request acknowledgment for a message that it sent.

PMS Link Maximum Retransmissions

Valid entries	Usage
1 to **5**	Enter the number of times that the server retransmits a message to the PMS in response to a negative acknowledgment, or sends an inquiry for acknowledgment from the PMS before giving up on the message.

PMS Log Endpoint

Valid entries	Usage
Valid extension	Enter the data extension number the server dials to access PMS. Cannot be a VDN extension.
PMS	Use this value if the PMS is connected over a TCP/IP link, and this link is defined as PMS on the IP Services screen.
blank	

PMS Protocol Mode

Valid entries	Usage
normal	Indicate the message protocol mode used between the server and PMS. Coordinate this option with your PMS vendor.
transparent	

Seconds Before PMS Link Idle Timeout

Valid entries	Usage
5 to **20**	Enter the idle time in seconds that the server waits for an acknowledgment from the PMS before the server enters link failure mode from the PMS transmission link.

Take Down Link for Lost Messages

Valid entries	Usage
y/n	Enter **y** to cause the PMS link to come down if messages are being lost. Monitor your PMS error log if you use n.

Field descriptions for page 2

Figure 113: Hospitality screen

```
change system-parameters hospitality                           Page   2 of x

               HOSPITALITY
          Dual Wakeup? y    Daily Wakeup? y  VIP Wakeup? y
                                VIP Wakeup Per 5 Minutes: _____
                        Room Activated Wakeup With Tones?
                Time of Scheduled Wakeup Activity Report: ______
                 Time of Scheduled Wakeup Summary Report: ______
        Time of Scheduled Emergency Access Summary Report: ______
                                       Announcement Type:
                       Integrated Announcement Extension:
      Length of Time To Remain Connected To Announcement: ____
         Extension To Receive Failed Wakeup LWC Messages: ______
        Routing Extension On Unavailable Voice Synthesis: ______
                Display Room Information in Call Display?
                      Automatic Selection of DID Numbers?
                     Custom Selection of VIP DID Numbers?
                               Number of Digits from PMS:
                                        PMS Sends Prefix?
                   Number of Digits in PMS Coverage Path:
                                  Digit to Insert/Delete:
```

Announcement Ports

This field appears only when the **Announcement Type** field is **voice-synthesis**. For the **voice-synthesis** announcement type, this indicates the equipment location of two ports on the voice synthesizer circuit pack. Enter the necessary characters.

Valid entries	Usage
01 to **03** (DEFINITY CSI) or **1** to **64** (S87XX/S8300 Servers)	First and second characters are the cabinet number
A to **E**	Third character is the carrier
0 to **20**	Fourth and fifth character are the slot number
01 to **04** (Analog TIE trunks) **01** to **31**	Six and seventh characters are the circuit number
1 to **80** (DEFINITY CSI) or (S87XX/S8300 Servers)	Gateway

Valid entries	Usage
V1 to **V9**	Module
01 to **31**	Circuit

Announcement Type

This indicates the type of automatic wakeup announcement the hotel guest receives. Allowable entries are as follows:

Valid entries	Usage
external	Applicable when using an announcement adjunct. If **external** is used, complete the **Auxiliary Board for Announcement** field.
integrated	Applicable when using the TN750B or TN750C announcement circuit pack. If **integrated** is used, complete the **Integrated Announcement Extension** field. The extension you enter must be a valid integrated announcement extension (administered on the Recorded Announcements screen) or a VDN.
mult-integ	Multi-integrated; applicable when using the TN750B or TN750C announcement circuit pack. **mult-integ** allows the automatic wakeup feature to use integrated announcement circuit packs to play any one of multiple announcements to different extensions during a wakeup call. If **mult-integ** is used, complete the **Default Announcement Extension** field. The extension you enter must be a valid integrated announcement extension (administered on the Recorded Announcements screen) or a VDN.
voice-synthesis	If **voice-synthesis** is used, complete the **Announcement Ports** field.
music-on-hold	If **music-on-hold** is used, no other field appears.
silence	If **silence** is used, no other field appears.

Automatic Selection of DID Numbers

This field assigns a 2 to 5-digit number to a guest's telephone number that is not associated with the room number.

Valid entries	Usage
y/n	Enter **y** to use the Automatic Selection of DID Numbers for Guest Rooms feature.

Auxiliary Board for Announcement

This field appears only when the **Announcement Type** field is **external**. This indicates the equipment location of an auxiliary trunk circuit that connects to the external announcement equipment. Enter the necessary characters.

Valid entries	Usage
01 to **03** (DEFINITY CSI) or **1** to **64** (S87XX/S8300 Servers)	First and second characters are the cabinet number
A to **E**	Third character is the carrier
0 to **20**	Fourth and fifth character are the slot number
01 to **04** (Analog TIE trunks) **01** to **31**	Six and seventh characters are the circuit number
1 to **80** (DEFINITY CSI) or **1** to **250** (S87XX/S8300 Servers)	Gateway
V1 to **V9**	Module
01 to **31**	Circuit

Custom Selection of VIP DID Numbers

This field allows you to select the DID number assigned to a room when a guest checks in. This field can be accessed only if the **Automatic Selection of DID Numbers** field is **y**.

Valid entries	Usage
y/n	Enter **y** to allow you to select the DID number assigned to a room when a guest checks in.

Daily Wakeup

Valid entries	Usage
y/n	Enter **y** if each extension can request daily wakeup calls.

Default Announcement Extension

This field appears only when the **Announcement Type** field is **mult-integ**. This indicates the default wakeup announcement extension when using the integrated announcement circuit pack.

Valid entries	Usage
valid extension or VDN	Enter the extension of the announcement you want to use for default wakeup calls.

Digit to Insert/Delete

Enter the leading digit that can be deleted and inserted back as described in the following text. The current PMS message set uses the extension number as the room identifier. In many customer configurations, the leading digit of the extension number is dropped to screen the room number. In order to accommodate PMS devices that are based on room number and not extension, this leading digit can be deleted on messages from Communication Manager to the PMS, and then inserted back on messages from the PMS to Communication Manager.

Note:
The PMS interface supports 3-, 4-, or 5-digit extensions, but prefixed extensions do not send the entire number across the interface. Only the assigned extension number is sent. Therefore, you should not use prefixed extensions for numbers that are also going to use the Digit to Insert/Delete function.

Valid entries	Usage
0 to **9**	Enter the leading digit that can be deleted and inserted back as described in the following text.

Display Room Information in Call Display

This indicates the type of guest room information displayed on telephone displays.

Valid entries	Usage
y	If this field is set to **y**, the telephones display the name and room number. The extension number and room number are not always the same number.
n	If this field is set to **n**, the telephones display the name and extension number.

Dual Wakeup

Valid entries	Usage
y/n	Enter **y** if each extension can request two wakeup calls within one 24-hour time period.

Extension to Receive Failed Wakeup LWC Messages

This indicates where unsuccessful wakeup LWC messages are stored. This is usually administered to an unassigned extension (cannot be a VDN extension) or to the attendant (attd). In addition, a LWC lamp for that extension is usually assigned to the attendant console as an indication of failed wakeup calls.

Valid entries	Usage
Unassigned extension	Enter the extension where unsuccessful wakeup LWC messages are stored.

Integrated Announcement Extension

This field appears only when the **Announcement Type** field is **integrated**. This indicates the wakeup announcement extension when using the integrated announcement circuit pack.

Valid entries	Usage
Valid extension or VDN	Enter the extension of the announcement you want to use for wakeup calls.

Length of Time to Remain Connected to Announcement

This applies only after the guest has heard the announcement completely one time, but continues to listen.

Valid entries	Usage
0 to **300**	Enter the length of time in seconds that a hotel guest will be connected to an announcement.

Number of Digits from PMS

This indicates the number of digits being sent from the PMS to the server to identify room numbers.

Note:
If the **Number of Digits from PMS** field is blank and the **PMS Sends Prefix** field is set to **n**, the server does not support an extension that starts with **0**.

Valid entries	Usage
1 to **4**	When using normal mode, digits **1** through **4** are valid.
1 to **5**	When using transparent or ASCII mode, digits **1** through **5** are valid.
blank	If using mixed numbering in the server, leave this field blank.

Number of Digits in PMS Coverage Path

This indicates whether the coverage paths are **3** or **4** digits long. There can be up to 9999 coverage paths.

Valid entries	Usage
3 to **4**	Indicate whether the coverage paths are 3 or 4 digits long.

PMS Sends Prefix

This indicates if the PMS sends a prefix digit to the server as part of the room numbering plan.

Note:
If the **PMS Sends Prefix** field is set to **n** and the **Number of Digits from PMS** field is blank, the server does not support an extension that starts with **0**.

Valid entries	Usage
y/n	Enter **y** or **n** to indicate if the PMS sends a prefix digit to the server as part of the room numbering plan.

Room Activated Wakeup with Tones

Valid entries	Usage
y/n	Enter **y** if wakeup calls can be activated via tones that prompt users for the time they wish to waken. This allows room activated wakeup calls without the use of a speech synthesizer or a display telephone.

Routing Extension on Unavailable Voice Synthesis

Valid entries	Usage
Assigned extension (cannot be a VDN extension) or **attd**	A call is placed to this extension if a voice synthesis port is not available during voice synthesis entry of wakeup requests.

CAUTION:
Set the following reports for a time other than when the system does its scheduled maintenance tests. To make sure the times do not overlap, enter the command `display system-parameters maintenance` and check when the system is set to run tests.

Time of Scheduled Emergency Access Summary Report

This indicates the time of day that the Emergency Access Summary Report is printed on the Journal/ Schedule printer.

Valid entries	Usage
hh:mm:am/pm	Enter the time where hh=hour, mm=minute, am/pm=A.M. or P.M.

Time of Scheduled Wakeup Activity Report

This indicates the time of day that the Wakeup Activity Report is printed on the Journal/Schedule Printer. This report summarizes the wakeup activity for each extension that had wakeup activity for the past 24 hours.

Valid entries	Usage
hh:mm:am/pm	Enter the time where hh=hour, mm=minute, am/pm=A.M. or P.M.

Time of Scheduled Wakeup Summary Report

This indicates the time of day that the Wakeup Summary Report is printed on the Journal/Schedule printer. This report gives an hour-by-hour summary of the number of scheduled wakeup calls and a list of extensions to which wakeup calls were attempted but did not complete during the hour.

Valid entries	Usage
hh:mm:am/pm	Enter the time where hh=hour, mm=minute, am/pm=A.M. or P.M.

VIP Wakeup

Valid entries	Usage
y/n	Enter **y** if each extension can request VIP wakeup calls.

VIP Wakeups Per 5 Minutes

This field appears if the **VIP Wakeup** field is **y**.

Valid entries	Usage
1 to **50**	Enter the number of VIP Wakeup calls allowed in a 5-minute interval.

Field descriptions for page 3

Figure 114: Hospitality screen

```
change system-parameters hospitality                              Page 3 of x

                                   HOSPITALITY

ROOM STATES:

                  Definition for Rooms in State 1: Rooms in State 1
                  Definition for Rooms in State 2: Rooms in State 2
                  Definition for Rooms in State 3: Rooms in State 3
                  Definition for Rooms in State 4: Rooms in State 4
                  Definition for Rooms in State 5: Rooms in State 5
                  Definition for Rooms in State 6: Rooms in State 6

HOSPITALITY FEATURES

                                            Suite Check-in? n
```

ROOM STATES

Definition for Rooms in State 1 through 6

Enter up to a 30-character definition for each room status. For example, you could define state 1 as ‘clean, ready to use’ and state 2 as ‘occupied, needs cleaning.’

The definitions for room states (Field descriptions for page 3), are for Attendant Room Status only. If you are not using Attendant Room Status, you do not need to complete these fields.

HOSPITALITY FEATURES

Suite Check-in

This field allows attendants to have the system automatically check-in several related extensions with one `check-in` command.

Valid entries	Usage
y/n	Enter **y** to use the Suite Check-in feature. See Hospitality in *Avaya Aura™ Communication Manager Feature Description and Implementation,* 555-245-205, for more information.

Hunt Group

Hunt groups allows calls to be answered by users (agents) at a predefined group of telephones or devices.

Use the Hunt Group screen to create a hunt group, identified by a hunt group number, and to assign hunt group member users by their extension numbers.This screen can also be used to implement associated features such as Automatic Call Distribution (ACD) and Hunt Group Queuing. The total number of pages can vary, depending on system configuration. See the *Avaya Aura™ Communication Manager Hardware Description and Reference*, 555-245-207, for the maximum number of hunt groups supported by each configuration.

When a call comes into a hunt group, the system checks for the busy or idle status of extension numbers in the hunt group when answering. A Uniform Call Distribution (UCD) type hunt group selects the "most idle" extension in the group when answering a new call. A Direct Department Calling (DDC) type hunt group selects the first available extension (in the administered sequence) when answering a new call. Expert Agent Distribution (EAD), used only with Expert Agent Selection (EAS), selects the "most idle" agent or the "least occupied" agent with the highest skill level for the call’s skill.

Note:
Vector controlled splits/skills can be called directly via the split/skill extension (instead of calling a VDN mapped to a vector that terminates the call to a vector controlled split/skill); however, the calls do not receive any announcements, be forwarded, redirect to coverage, or intraflow/interflow to another hunt group.

The Hunt Group screen can vary according to system configuration and values populating particular fields.The following figures show several ways that page 1 of the Hunt Group screen might appear. The descriptions that follow the figures include all fields shown in all variations of Page 1.

Field descriptions for page 1

Figure 115: Hunt Group screen when Queue is y

```
change hunt-group n                                              Page 1 of X
                                    HUNT GROUP

           Group Number: 4__                                          ACD? _
             Group Name: ____________________________               Queue? y
                                                               Queue Limit:____
        Group Extension: ________                                  Vector? _
             Group Type: _______                            Coverage Path: ____
                     TN: ___                    Night Service Destination: ____
                    COR: _                                MM Early Answer? _
          Security Code: ____                       Local Agent Preference? _
       ISDN Caller Disp: ________


Calls Warning Threshold: ___   Port: x____  Extension: ____
 Time Warning Threshold: ___   Port: x____  Extension: ____
```

Figure 116: Hunt Group screen when Queue and Vector are n

```
change hunt-group n                                               Page 1 of X
                                    HUNT GROUP

            Group Number: ___                                         ACD? n
              Group Name: ____________________________              Queue? n
         Group Extension: ____                                     Vector? n
              Group Type: ________                          Coverage Path: ____
                      TN: ___                  Night Service Destination: _____
                     COR: _                               MM Early Answer? _
           Security Code: ____                      Local Agent Preference? _
     ISDN Caller Display: ________
```

The two **Extension** fields display only when the **Calls Warning Port** and the **Time Warning Port** fields are **x**.

Figure 117: Hunt Group screen when Queue and Vector are y

```
change hunt-group n                                            Page 1 of X
                                      HUNT GROUP

            Group Number: ___                                         ACD? n
              Group Name: ____________________________              Queue? y
         Group Extension: ________                                 Vector? y
              Group Type: ______
                      TN: ___
                     COR: _                                MM Early Answer?
           Security Code: ____                      Local Agent Preference? _
      ISDN Caller Display: ________


 Calls Warning Threshold: ___    Port: x___    Extension: ____
  Time Warning Threshold: ___    Port: x___    Extension: ____
```

The two **Extension** fields display only when the **Calls Warning Port** and the **Time Warning Port** fields are x.

Figure 118: Hunt Group screen when Queue is y and Vector is n

```
change hunt-group n                                              Page 1 of X
                                      HUNT GROUP

            Group Number: ___                                         ACD? n
              Group Name: ____________________________              Queue? n
         Group Extension: ________                                 Vector? n
              Group Type:                                   Coverage Path: ____
                      TN:                         Night Service Destination: _____
                     COR:                                  MM Early Answer?
           Security Code: ____                      Local Agent Preference? _
        ISDN Caller Disp: ________


 Calls Warning Threshold: ___    Port: x___    Extension: ____
  Time Warning Threshold: ___    Port: x___    Extension: ____
```

ACD

Indicates whether Automatic Call distribution is used. This field cannot be set to **y** if, on the System Parameters Customer-Options (Optional Features) screen, the **ACD** field is **n**.

Valid entries	Usage
y	The hunt group function as an ACD split/skill. AUDIX hunt groups can function as ACD splits/skills.
n	This feature is not desired, even if, on the System Parameters Customer-Options (Optional Features) screen, the **ACD** field is **y**. When the hunt group is assigned as an ACD split/skill, the hunt group members serve as ACD agents. The agents in this split/skill must log in to receive ACD split/skill calls. If this hunt group is on a remote server running Communication Manager and using the AUDIX in a DCS feature, then enter **n**.

Calls Warning Threshold

Appears if the **Queue** field is **y**. Enter the number of calls that can be queued before the System flashes the queue status (feature buttons assigned on agents telephones) and the optional Auxiliary Queue Call Warning Threshold lamp assigned to the split/skill. These lamps are lighted steadily when at least one call is in queue and the threshold has not yet been reached

Valid entries	Usage
1 to **999** and must be less than or equal to the queue length or blank	This field must not be left blank if **Calls Warning Port** is assigned a port number.

(Calls Warning) Extension

Appears if the **Queue** field is **y** and when the **Calls Warning Port** and the **Time Warning Port** fields are **x**. An extension is needed when an X is placed in **Calls Warning Port**. This extension can be used by the Terminal Translation Initialization (TTI) feature to assign a port to this extension from the port itself. Once **Calls Warning Port** is assigned a valid port (either via TTI or the `change hunt-group` command), then the extension is removed and considered unassigned.

Valid entries	Usage
Extension	Enter an unassigned extension. This field cannot be blank.

(Calls Warning) Port

Appears if the **Queue** field is **y**. Enter the seven-character port number assigned to connect the optional external Auxiliary Queue Call Warning Threshold lamp that flashes when the number of calls in queue has exceeded the queue warning threshold (assigned in **Calls Warning Threshold**).

This port is assigned to an Analog Line circuit pack or given an **x** designation if an extension is used. Enter the necessary characters.

Valid entries	Usage
01 to **03** (DEFINITY CSI) or **1** to **64** (S87XX/S8300 Servers)	First and second characters are the cabinet number
A to **E**	Third character is the carrier
0 to **20**	Fourth and fifth character are the slot number
01 to **04** (Analog TIE trunks) **01** to **31**	Six and seventh characters are the circuit number

Note:
For example, 01A0612 is in cabinet 01, carrier A, slot 06, and circuit number (port) 12.

COR

Enter the class of restriction (COR) number that reflects the desired restriction for the hunt group. If this is a hunt group supporting the AUDIX in a DCS feature, the CORs on the Hunt Group screen on each server running Communication Manager must be the same.

Valid entries	Usage
0 to **995**	Enter the class of restriction (COR) number that reflects the desired restriction for the hunt group.

Coverage Path

Enter a coverage path number. This assigns a coverage path for the hunt group. The coverage path is assigned using the Coverage Path screen. Does not appear if the **Vector** field is **y**.

Valid entries	Usage
1 to **999**	Enter a coverage path number.
t1 to **t999**	Time of day table
blank	

Group Extension

Enter an unused extension number to be assigned to the hunt group. The field cannot be blank.

Valid entries	Usage
0 to **9**	Unassigned extension

Group Name

This field identifies the hunt group.

Valid entries	Usage
27-character string	Enter a character string that uniquely identifies the group (for example, “parts dept,” “purchasing,” or “sales dept”). **Note:** For 4610SW, 4620SW, 4621SW, and 4622SW, Sage, Spark, and 9600-series Spice telephones, the **Group Name** field has an associated optional native name field that is supported by the Unicode language display. The native name field is accessible through the Integrated Management Edit Tools such as Avaya Site Administration (ASA). Unicode is also an option for the 2420J telephone when **Display Character Set** on the **System Parameters Country-Options** screen is **katakana**. For more information on the 2420J, see *2420 Digital Telephone User's Guide*, 555-250-701. Avaya BRI stations support ASCII characters only. They do not support non-ASCII characters, such as Eurofont or Kanafont. Therefore, if you use non-ASCII characters in any Communication Manager **Name** field, such characters do not display correctly on a BRI station.

Group Number

This is a display-only field when the screen is accessed using an administration command such as `add` or `change`.

Group Type

The group types available depend on what is administered on your System Parameters Customer-Options (Optional Features) screen for **Automatic Call Distribution (ACD)**, **Expert Agent Selection (EAS)** and **Business Advocate**. The table below shows what group types are available depending on your configuration.

Each option uses a different method to select an extension or agent for a call when two or more extensions or agents are available. The second table shows how calls are handled for each group type.

	circ	ddc	ucd-mia	ead-mia	ucd-loa	ead-loa	pad	slm
ACD=n	x	x						
ACD, Split, Vector = n/y		x	x					
ACD, Skill, Vector = n/ y			x	x				x
ACD, Skill, Vector = y Advocate or Elite			x	x	x	x		
ACD, Skill, Vector = y Dynamic Advocate			x	x	x	x	x	

Valid entries	Usage
circ	Enter **circ** (circular) when the call should be routed in a "round-robin" order. The order in which you administer the extensions determines the order that calls are directed. The server running Communication Manager keeps track of the last extension in the hunt group to which a call was connected. The next call to the hunt group is offered to the next extension in the circular list, independent of how long that extension has been idle. You cannot use circular hunting with automatic call distribution, queues, or vectors.
ddc	Enter **ddc** when the call should be routed to the first extension or ACD agent assigned in the ACD split. Group type ddc is also known as "hot seat" distribution. "ddc" distribution is not available when the group is administered as a skill.
ucd-mia **ucd-loa**	When **ucd-mia** or **ucd-loa** is entered, a call routes to the most-idle agent based on when the agent finished the most recent call (**ucd-mia**), or the least occupied agent based on agent occupancy (**ucd-loa**). Enter **ucd-mia** or **ucd-loa** if the hunt group has an AUDIX message. One of these entries is required when supporting the Outbound Call Management feature and when the **Controlling Adjunct** field is **asai**.
ead-mia **ead-loa**	When **ead-mia** or **ead-loa** is entered, a call routes to the available agent with the highest skill level for the call. If two or more agents with equal skill levels are available, Communication Manager routes the call to the most-idle agent based on when the agent finished the most recent call ("ead-mia"), or the least occupied agent based on agent occupancy ("ead-loa"). This allows a call to be distributed to an agent best able to handle it if multiple agents are available.
pad	Enter **pad** (percent allocation distribution) to select an agent from a group of available agents based on a comparison of the agent's work time in the skill and the agent's target allocation for the skill.
slm	Enter **slm** when you want to: ● Compare the current service level for each SLM-administered skill to a user-defined call service level target and identify the skills that are most in need of agent resources to meet their target service level. ● Identify available agents and assess their overall opportunity cost, and select only those agents whose other skills have the least need for their service at the current time.

ISDN/SIP Caller Disp

This field is required if, on the System Parameters Customer-Options (Optional Features) screen, the **ISDN-PRI**, **ISDN-BRI and SIP Trunks** field is **y**.

Valid entries	Usage
grp-name **mbr-name**	Enter **grp-name** or **mbr-name** to specify whether the hunt group name or member name, respectively, will be sent to the originating user. **NOTE:** Avaya BRI stations support ASCII characters only. They do not support non-ASCII characters, such as Eurofont or Kanafont. Therefore, if you use non-ASCII characters in any Communication Manager **Name** field, such characters do not display correctly on a BRI station.
blank	Displays the VDN name to the originating user. If the **ISDN-PRI,** the **ISDN-BRI, or the SIP Trunks** field is **n**, this field must be blank.

Local Agent Preference

Use this field whether the routing of an incoming ACD call to an idle agent should be done by matching the location number of the incoming caller's station or trunk to the location number of an idle agent. This field can only be set to **y** if:

- the **Call Center Release** field on the System Parameters Customer-Options (Optional Features) screen is set to 3.0 or later
- the **Expert Agent Selection** and the **Multiple Locations** fields on the System Parameters Customer-Options (Optional Features) screen are **y**
- the hunt group is defined as a skill hunt group (the **Skill?** field on page 2 of the Hunt Group screen is set to **y**)

Valid entries	Usage
y/n	Enter **y** to indicate that an incoming ACD call to an idle agent should be routed by matching the location number of the incoming caller's station or trunk to the location number of an idle agent. Default is **n**.

Interruptible Aux Deactivation Threshold

This field specifies the maximum level of service above which the system deactivates the Interruptible Aux feature.

Use the Interruptible Aux threshold and the Interruptible Aux Deactivation Threshold fields to maintain a buffer between the two levels of service that trigger the Interruptible Aux feature.

The values of the Interruptible Aux

threshold and the Interruptible Aux Deactivation Threshold fields must have a difference of one unit.

The valid entries for this field depend on the value of the Interruptible Aux threshold field:

- For calls-warning-threshold, the valid entries are from 0 to 998.
- For service-level-target, the valid entries are from 0 to 100.
- For time-warning-threshold, the valid entries are from 0 to 998.
- For none, the system does not display the Interruptible Aux Deactivation Threshold field.

Interruptible Aux Threshold

This field specifies the minimum level of service below which the system must enable the Interruptible Aux feature.

Use the Interruptible Aux feature to make the Expert Agent Selection (EAS) agents in the AUX work mode available to receive calls. This feature works only if the agents have an interruptible reason code.

Valid entries	Usage
service-level-target	Use this option to enable the Interruptible Aux feature when the service level drops below the administered percentage of calls within the specified period. For example, if you set the **Service Level Target (% in sec)** field to 90% calls in 30 seconds, the system enables the Interruptible Aux feature if the service drops to 89% calls in 30 seconds.
calls-warning-threshold	Use this option to enable the Interruptible Aux feature when the number of calls in the queue for a hunt group exceeds the specified number of calls. For example, if you set the **Calls Warning Threshold** field to 20, the system enables the Interruptible Aux feature if the number of calls in the queue exceeds 20.
time-warningthreshold	Use this option to enable the Interruptible Aux feature when the oldest call is in the queue for longer than the specified number of seconds. For example, if you set the **Time Warning Threshold** field to 60 seconds, the system enables the Interruptible Aux feature if the duration of the oldest call in the queue for a hunt group exceeds 60 seconds.
none	Use this option if you do not want to administer the Interruptible Aux feature.

Valid entry	Usage

Valid entry	Usage

MM Early Answer

This field applies for systems using Multimedia Call Handling only.

Valid entries	Usage
y/n	The system begins to answer an H.320 call and establish an audio channel before offering the conversion call to the hunt group. This starts billing for the call when the call is first put into queue.

Night Service Destination

Enter the destination where calls to this split redirects when the split is in the night service mode. Not all features work correctly if this is not a local extension. Does not appear if the **Vector** field is **y**.

Valid entries	Usage
An assigned extension number (can be a VDN extension)	Enter the destination where calls to this split redirects when the split is in the night service mode.
attd	An attendant group code.
blank	

Queue

Specifies a queue for the hunt group.

Valid entries	Usage
y/n	Enter **y** so the hunt group is served by a queue.

Queue Limit

This field appears when the **Queue** field is set to **y**.

Valid entries	Usage
1 to **999**	Enter a limit to the number of calls that will queue.
unlimited	The system dynamically allocates the queue slots from a common pool on an as-needed basis.

Security Code

Enter a security code (password) used for the Demand Print feature.

Valid entries	Usage
3 to 8 digits	Enter the password for the Demand Print feature.

Time Warning Threshold

Appears if the **Queue** field is **y** and when the **Calling Warning Port** and the **Time Warning Port** fields are **x**. Enter the time in seconds that a call can remain in the queue before the system flashes the Queue status lamps (feature buttons assigned members telephones) and the Auxiliary Queue Time Warning lamp assigned to this split/skill.

Valid entries	Usage
0 to **999** or blank	An entry of **0** provides a warning whenever a call is queued.

(Time Warning) Extension

Appears if the **Queue** field is **y**. An extension is needed when an **x** is placed in **Time Warning Port**. This extension can be used by the Terminal Translation Initialization (TTI) feature to assign a port to this extension from the port itself. Once **Time Warning Port** is assigned a valid port (either via TTI or the `change hunt-group` command), then the extension is removed and considered unassigned.

Valid entries	Usage
Extension	Enter an unassigned extension. This field cannot be blank.

(Time Warning) Port

Appears if the **Queue** field is **y**. Enter the seven-character port number assigned to the Auxiliary Queue Time Warning lamp that flashes when the time entered in **Time Warning Threshold** has been reached by a call in queue.

Note:
This port is assigned to an Analog Line circuit pack or given an **X** designation if an extension is used. Enter the necessary characters.

Valid entries	Usage
01 to **03** (DEFINITY CSI) or **1** to **64** (S87XX/S8300 Servers)	First and second characters are the cabinet number
A to **E**	Third character is the carrier
0 to **20**	Fourth and fifth character are the slot number
01 to **04** (Analog TIE trunks) **01** to **31**	Six and seventh characters are the circuit number

For example, 01A0612 is in cabinet 01, carrier A, slot 06, and circuit number (port) 12.

TN

Valid entries	Usage
1 to **100**	Enter the Tenant Partition number.

Vector

See example screens for fields that display when this field is **y**.

Valid entries	Usage
y/n	Enter **y** to indicate that this hunt group is vector controlled. On the System Parameters Customer-Options (Optional Features) screen, the **Vectoring-Basic** field must be **y** before **y** can be entered here.

Field descriptions for page 2

Page 2 of the Hunt Group screen can vary according to values for particular fields on Page 1. The screen shown in Figure 119 appears only when the **ACD** field on page 1 is **y**, and the **Queue** and **Vector** fields are **n**. If the **ACD** field is **n**, the Message Center page shown in Figure 121 becomes Page 2 and all subsequent page numbers are decreased by one. The **Timed ACW Interval** field shown below appears only if, on the System Parameters Customer-Options (Optional Features) screen, the **Timed ACW** field is **y**. The screen shown in Figure 120 appears when the **Queue** and **Vector** fields are **y**.

The descriptions that follow the figures include all fields in both variations of Page 2.

Figure 119: Hunt Group screen when ACD is y and Queue and Vector are n

```
change hunt-group n                                                 Page 2 of X
                                  HUNT GROUP
                  Skill? _        Expected Call Handling Time (sec): ___
                    AAS? _          Service Level Target (% in sec):80 in 20
               Measured: ____
   Supervisor Extension: ____
  Priority on Intraflow? _

    Controlling Adjunct: ____




Timed ACW Interval (sec): ___             Maximum Auto Reserve Agents: 0
  Multiple Call Handling: _____________
                                        Redirect on No Answer (rings): ___
                                                      Redirect to VDN: _____
                       Forced Entry of Stroke Counts or Call Work Codes? _
```

Figure 120: Hunt Group screen when Queue and Vector are y

```
change hunt-group n                           P                          Page 2 of X
                                  HUNT GROU
                   Skill?           Expected Call Handling Time (sec):___
                     AAS? _           Service Level Target (% in sec):80 in 20
                Measured: internal
    Supervisor Extension: ____



     Controlling Adjunct: ____



       VuStats Objective: ____
Timed ACW Interval (sec): ___ ___               Maximum Auto Reserve Agents: 0
  Multiple Call Handling: _____________
                                          Redirect on No Answer (rings): ___
                                                        Redirect to VDN: _____
                         Forced Entry of Stroke Counts or Call Work Codes? _
```

AAS

Appears when the **ACD** field is **y**

.

Valid entries	Usage
y/n	Enter **y** if this hunt group is to serve as an Auto-Available Split.

Acceptable Service Level (sec)

Appears if the **ACD** field is **y** and the **Measured** field is **internal** or **both**. This allows BCMS and/or VuStats to report a percentage of calls that were answered within the specified time. This entry is also used by the Business Advocate Service Objective feature.

Valid entries	Usage
0 to **9999** seconds	Enter the number of seconds within which calls to this hunt group should be answered.

Adjunct CTI Link

Appears when the **ACD** field is **y** and the **Controlling Adjunct** field is **asai** or **adjlk**. Enter the appropriate ASAI CTI Link. This field cannot be blank.

Controlling Adjunct

Appears only if the **ACD** field is **y**. If the controlling adjunct is a CONVERSANT voice system (requires an ASAI link), then enter **asai** in this field. (On the System Parameters Customer-Options (Optional Features) screen, the **ASAI Link Core Capabilities** and **Computer Telephony Adjunct Links** fields must be **y** for CallVisor ASAI capability and for an entry other than **none**.)

Valid entries	Usage
none	Indicates that members of the split/skill or hunt group are not controlled by an adjunct processor.
asai	All agent logins are controlled by an associated adjunct and logged-in agents can only use their data terminal keyboards to perform telephone functions (for example, change work state).
adjlk	Computer Telephony Adjunct Links
asai-ip	Indicates ASAI links administered without hardware.
adj-ip	Indicates ASAI adjunct links administered without hardware.

Dynamic Percentage Adjustment

Appears when **ACD** and **Group Type** fields on the Hunt Group screen are **pad** and the **Business Advocate** field is **y** on the System Parameters Customer-Options (Optional Features) screen.

Valid entries	Usage
y/n	Enter **y** to enable automatic adjustments to agents' target allocations as needed to help meet the administered service level targets.

Dynamic Queue Position

Appears when the **ACD** and **Skill** fields are **y** on the Hunt Group screen and the **Business Advocate** field is **y** on the Feature-Related System Parameters screen.

Valid entries	Usage
y/n	Enter **y** to apply the dynamic queue operation to the calls queued to the skill.

Dynamic Threshold Adjustment

Appears when the **ACD** and **Service Level Supervisor** fields on the Hunt Group screen are **y** and the **Business Advocate** field is **y** on the System Parameters Customer-Options (Optional Features) screen.

Valid entries	Usage
y/n	Enter **y** to enable automatic adjustments to overload thresholds to engage reserve agents a bit sooner or a bit later to meet the administered service levels.

Expected Call Handling Time (sec)

Appears if, on the System Parameters Customer-Options (Optional Features) screen, either the **Vectoring (Advanced Routing)** or **Business Advocate** field is **y**. and, on the Hunt Group screen, the **ACD** field is **y**.

.

Valid entries	Usage
1 to **9999** in increments of 1	Establishes the number of seconds for expected call handling. This value is used to initialize Expected Wait Time and is also used by the Business Advocate Percent Allocation feature.

Forced Entry of Stroke Counts or Call Work Codes

Appears when the **ACD** field is **y** and **Controlling Adjunct** field is **none**.

Valid entries	Usage
y/n	Enter **y** so either a Stroke Count or Call Work Code must be entered for each call answered by an agent when in the Manual-In mode.

Inflow Threshold (sec)

Appears only when the **ACD** and **Queue** fields are **y** and **Vector** field is **n**. Enter the number of seconds that a call can remain in the queue before no more calls are accepted by the queue. If **0** is entered, a call is redirected to this split/skill only if there is an available agent.

Valid entries	Usage
0 to **999**	Enter the number of seconds that a call can remain in the queue before no more calls are accepted by the queue.

Level 1 Threshold (sec)

Enter the number of seconds corresponding to the Expected Wait Time (EWT) you want to set for this threshold. For example, if you enter 45 calls whose EWT exceeds 45 seconds are classified as over threshold 1. This field is used with Service Level Supervisor and only appears if the **Service Level Supervisor** field is **y**.

Level 2 Threshold (sec)

Appears if the **ACD** field is **y**. Enter the number of seconds corresponding to the Expected Wait Time (EWT) you want to set for this threshold. For example, if you enter 60 calls whose EWT exceeds 60 seconds are classified as over threshold 2. This field is used with Service Level Supervisor and only appears if the **Service Level Supervisor** field is **y**.

Maximum Auto Reserve Agents

Appears only if the **Group Type** field is **slm**. Set the maximum number of Auto Reserve Agents you want to be available for this skill (hunt group). Any time an auto-reserve skill is in danger of falling below its target service level percent, some of this skill's agents are auto-reserved (kept idle in other skills) so they are available when a new call arrives for this skill. Valid values are **0** to **9**. Default is **0**.

Measured

Provides measurement data for the ACD split/skill collected (internal to the switch) for **VuStats** or **BCMS**. This measurement data is collected for **VuStats** and **BCMS** only if, on the System Parameters Customer-Options (Optional Features) screen, they are **y** and, on the Hunt Group screen, the **ACD** field is **y**.

Valid entries	Usage
internal	If you enter **internal** in this field and on the System Parameters Customer-Options (Optional Features) screen neither the **VuStats** or **BCMS** field is **y**, the system displays the following message: `<value> cannot be used; assign either BCMS or VuStats first` Contact your Avaya representative to assist with any changes you want to make on the System Parameters Customer-Options (Optional Features) screen.
external	Provides measurements made by the Call Management System (external to the server running Communication Manager).
both	Provides measurements collected both internally and externally.
none	Measurement reports for this hunt group are not required.

Multiple Call Handling

Appears only if, on the System Parameters Customer-Options (Optional Features) screen, the **Multiple Call Handling** field is **y** and the **ACD** field on this screen is **y**. This field defines whether the hunt group can have multiple call handling capabilities, and if so, what type.

Valid entries	Usage
none	Agents who are members of that split/skill can only receive an ACD call from that split/skill when the telephone is idle.
on-request	Agents in the Multiple Call Handling split/skill can place a non-ACD or an ACD call on hold and select an available work mode. A queued ACD split/skill or direct agent call then is routed to the agent.
many-forced	An ACD call is delivered automatically to an idle line appearance if the agent is in the Auto-In/Manual-In (MI/AI) work mode and an unrestricted line appearance is available.
	1 of 2

Valid entries	Usage
one-forced	An ACD call is delivered automatically to an idle line appearance if the agent has no other ACD call on the station, is in the Auto-In/Manual-In (MI/AI) work mode, and an unrestricted line appearance is available.
one-per-skill	An ACD call is delivered automatically to an idle line appearance if the agent has no other ACD call for that skill on the station, is in the Auto-In/Manual-In (MI/AI) work mode, and un unrestricted line appearance is available. Valid in an EAS environment and only when the **Skill** field is **y**.
	2 of 2

Priority On Intraflow

Appears if the **ACD** field is **y** and the **Vector** field is **n**.

Valid entries	Usage
y/n	Enter **y** for calls intraflowing from this split to a covering split to be given priority over other calls waiting in the covering split queue.

Redirect on No Answer (rings)

Appears if the **ACD** field is **y**.

Valid entries	Usage
1 to **20**	Enter the maximum number of rings before a call redirects back to the split/skill, or to the administered VDN.
blank	Deactivates Redirect on No Answer.

Redirect to VDN

Appears if the **ACD** field is **y**. To redirect a RONA call to a VDN instead of to the split/skill, enter the extension number of the VDN. The administered VDN must be on-premises and must be administered on the system. The VDN can specify a vector that in turns route to an off-premises VDN. You cannot enter an extension in this field if the **Redirection on No Answer (rings)** field is blank. Direct Agent calls go to the agent’s coverage path if it is administered. If not, the calls go to a VDN.

Valid entries	Usage
Assigned VDN or blank	To redirect a RONA call to a VDN instead of to the split/skill, enter the extension number of the VDN.

Service Level Interval

This field displays only if **Actual** is assigned as the SLM algorithm on the Feature-Related System Parameters screen, and **Group Type** on the Hunt Group screen is set to **slm**. The interval can be set to the same interval used when specifying the target objectives for the application.

Valid entries	Usage
hourly	ASL algorithm calculations for accepted call and total call components are set to 0 at hourly intervals.
daily	ASL algorithm calculations for accepted call and total call components are set to 0 at daily intervals. This is the default.
weekly	ASL algorithm calculations for accepted call and total call components are set to 0 at weekly intervals. The weekly interval starts as 00:00 hours on Sunday.

Service Level Supervisor

Appears if, on the System Parameters Customer-Options **(Optional Features)** screen, the **Business Advocate** field is **y** and, on the Hunt Group screen, the **ACD** and **Skill** fields are **y**. For information on Business Advocate, contact your Avaya representative, or see the *Avaya Business Advocate User Guide*, 07-300653.

Valid entries	Usage
y/n	Enter **y** to use Service Level Supervisor for this skill.

Service Level Target (% in sec)

Appears when the **ACD** field and the **Dynamic Percentage Adjustment** or **Dynamic Threshold Adjustment** field on the Hunt Group screen is **y** and the **Business Advocate** field is **y** on the System Parameters Customer-Options (Optional Features) screen. Also appears when the **Group Type** field on the Hunt Group screen is **slm**, and on the System Parameters Customer-Options (Optional Features) screen, the **Service Level Maximizer** field is set to **y**, and the **Business Advocate** field is set to **n**.

Valid entries	Usage
1 to **99** (percentage) **1** to **9999** (time in seconds)	Enter the percentage and time components of the service level target. After 20 seconds, default service level target values are set at 80%.

Service Objective

Appears when the **Skill** and **Business Advocate** fields on the Feature-Related System Parameters screen are **y** and, on the Hunt Group screen, the **ACD** field is **y**.

Valid entries	Usage
1 to **9999**	Enter the per-skill service objective.

Skill

Appears if, on the System Parameters Customer-Options (Optional Features) screen, the **Expert Agent Selection** field is **y** and, on the Hunt Group screen, the **ACD** field is **y**.

If this field is **y**, then the **Group Type** field must be **ucd** or **ead**.

Valid entries	Usage
y/n	Enter **y** if this hunt group is to be an EAS skill.

SLM Count Abandoned Calls

Appears only if **Actual** is assigned as the SLM algorithm on the Feature-Related System Parameters screen, and Group Type on the Hunt Group screen is **slm**.

Valid entries	Usage
y	Abandoned calls are included in the ASL algorithm calculations for SLM.
n	Abandoned calls are *not* included in the ASL algorithm calculations for SLM. Use this option when reporting for this application does not account for calls that are abandoned while in skill queues.

Supervisor Extension

Appears if the **ACD** field is **y**.

Valid entries	Usage
Valid extension	Enter the extension number (cannot be a VDN number) of the ACD split/ skill supervisor that agents reach when using the Assist feature

Timed ACW Interval (sec)

When a value is entered in this field, an agent in auto-in work mode who receives an ACD call from this hunt group is placed automatically into After Call Work (ACW) when the call drops. Enter the number of seconds the agent should remain in ACW following the call. When the administered time is over, the agent automatically becomes available. Timed ACW cannot be administered if the hunt group is adjunct controlled, is an AUDIX Message Center, or is an auto-available split. The **Timed ACW Interval** field appears if, on the System Parameters Customer-Options **(Optional Features)** screen, the **Timed ACW** field is **y**, and, on the Hunt Group screen, the **ACD** field is **y**.

Note:
This field can be overridden by the settings on the **VDN Timed ACW Interval** field on the Vector Directory Number screen. Coordinate the settings for both fields in setting up delays.

Valid entries	Usage
1 to **9999** or blank	The number of seconds the agent should remain in ACW following the call.

VuStats Objective

Enter a numerical user-defined objective. An objective is a split or skill goal for the call. This could be an agent objective such as a specific number of calls handled or an average talk time. The objective could also be a percent within the service level. The objective appears on the VuStats display and allows agents and supervisors to compare the current performance against the value of the objective for the split or skill.

You can use this value in a customized VuStats display format if, on the VuStats display format screen, the **Object Type** field is either **agent**, **agent-extension**, or **split**.

This field appears if, on the System Parameters Customer-Options (Optional Features) screen, the **VuStats** field is **y** and the **Measured** field is either **internal** or **both** and, on the Hunt Group screen, the **ACD** field is **y**.

Valid entries	Usage
0 to **99999**	Enter a split or skill objective.

Field descriptions for Message Center page

The Hunt Group screen pages and fields can vary according to system configuration and values populating particular fields. The figure below is only an example, and is intended to show most of the fields that might appear on this page of the Hunt Group screen. This example might not show all fields, or might show fields that normally do not appear together. Your own screen might vary from this example according to specific field and system settings. The list of field descriptions that follows the figure is in alphabetical order for quick reference. This list is intended to be comprehensive, and to include information on all fields that might appear.

Figure 121: Hunt Group Message Center screen

```
                                                                Page  2 of  X
                                    HUNT GROUP

                                   Message Center: ________
                                Voice Mail Number: _________________
 Routing Digits (e.g. AAR/ARS Access Code): ____
                         TSC per MWI Interrogation? n
                              Send Reroute Request? y           Provide Ringback?
                                   AUDIX Extension: ________
                         Message Center AUDIX Name: ______
                                          Primary? _
     Calling Party Number to INTUITY AUDIX? _
                                     LWC Reception: _______
                                        AUDIX Name: _______
                             Messaging Server Name: _______

           First Announcement Extension: 1234 ___ Delay (sec): 20
          Second Announcement Extension: ________ Delay (sec): __  Recurring? _
```

AUDIX Name

Enter the name of the AUDIX machine as it appears on the **IP Node Names** screen. Add the AUDIX name to the IP Node Names screen before entering it in this field. For more information on the IP Node Names screen, see *Administering Network Connectivity on Avaya Aura™ Communication Manager*, 555-233-504.

Calling Party Number to INTUITY AUDIX

Appears when the **Message Center** field is **audix** or **rem-vm**.

Valid entries	Usage
y/n	Enter **y** to send the calling party number to INTUITY AUDIX.

First Announcement Delay (sec)

Appears only if the **Queue** field is **y** and the **Vector** field is **n**. Enter the number of seconds that a call remains in queue before the associated first announcement is given the calling party. The call retains its place in the queue while the caller is listening to the recorded announcement. If the call hasn't been answered after the announcement, the caller hears music (for first announcement only) if Music-on-Hold is provided or ringing for as long as the call remains in queue. Appears only if the **Queue** field is **y** and the **Vector** field is **n**.

.

Valid entries	Usage
0 to **99**	When **0** is entered, the first announcement is provided immediately to the caller. This value is set automatically to **0** if there is no queue.
blank	This field must be blank if there is no first announcement.

First Announcement Extension

Appears when the **ACD** and **Queue** fields are **y** and the **Vector** field is **n**.

Note:
If entering a Multi-Location Dial Plan shortened extension, note the following: When entering a Multi-Location Dial Plan shortened extension in a field designed for announcement extensions, certain administration end validations that are normally performed on announcement extensions are not done, and resultant warnings or submittal denials do not occur. The shortened extensions also do not appear in any display or list that shows announcement extensions. Extra care should be taken to administer the correct type of announcement for the application when assigning shortened extensions.

Valid entries	Usage
Enter a recorded announcement extension number.	This is the announcement the caller receives after being in the queue for the time interval specified in First Announcement Delay. If the call hasn't been answered after the announcement, the caller hears music (only after the first announcement) if Music-on-Hold is provided, or ringing for as long as it remains in the queue. If this is the forced first announcement, the caller always hears ringback after the announcement; otherwise, the caller hears music (if provided).
blank	Leaving this field blank indicates there will be no announcement.

LWC Reception

Defines the destination for Leave Word Calling (LWC) messages left for the hunt group.

Valid entries	Usage
audix	If LWC is attempted, the messages are stored in AUDIX. The **Audix Name** field must be filled in too.
msa	Messaging Server Adjunct
spe	If LWC is attempted, the messages are stored in the system processing element (spe).
none	

Message Center

Enter the type of messaging adjunct for the hunt group. Only one hunt group in the system can be administered as **audix**, one as **qsig-mwi**, one as **fp-mwi**, one as **rem-audix**, and as many as six as **qsig-mwi**.

Valid entries	Usage
audix	For AUDIX located on this server running Communication Manager
fp-mwi	Public network allowing AUDIX to be located on another switch; administrable only when the **ISDN Feature Plus** field on the System Parameters Customer-Options (Optional Features) screen is **y**.
msa	Messaging Server Adjunct
msa-vm	For voice-mail system integrated using Mode Codes or Digital Station Emulation
none	Indicates the hunt group does not serve as a message hunt group.
rem-vm	DCS feature allowing voice mail to be located on another server
qsig-mwi	QSIG network allowing voice mail to be located on another server
sip-adjunct	SIP message center server.

Message Center AUDIX Name

Enter the name of the Message Center AUDIX. Appears on Avaya S8300/S87XX Servers if the **Message Center** field is **audix** or **rem-vm**.

Message Center MSA Name

Note:
Administer the IP Node Names screen first.

Enter the name of the Message Center MSA. When it appears, it replaces the **Message Center AUDIX Name** field. Appears on S8300/S87XX Servers if the **Message Center** field is **msa**.

Primary

Appears on Avaya S8300/S87XX Servers if the **Message Center** field is **audix** or **rem-audix**.

Valid entries	Usage
y/n	Enter **y** to indicate that the specified AUDIX is the primary adjunct.

Provide Ringback

Appears only if **Message Center** on the Hunt Group screen is **fp-mwi** or **qsig-mwi**. Use this field if you are using an SBS trunk for the QSIG MWI hunt group. If set to **y**, a call covering to the message center provides ringback to the caller during the coverage interval.

Valid entries	Usage
y/n	When set to **y**, ringback is provided to the calling party until a Connect is received for the call to the Messaging system. Ringback is discontinued upon receipt of the Connect indication. Default is **n**.

Routing Digits (e.g. AAR/ARS Access Code)

Appears only if **Message Center** is **qsig-mwi** or **fp-mwi**. Shows the AAR (most likely for a Message Center type of **qsig-mwi**) or ARS (most likely for a Message Center type of **fp-mwi**) access code which when prepended to the **AUDIX Complete Number** field defines a route to the Message Center switch hunt group containing the line ports to the AUDIX.

Valid entries	Usage
0 to **9**, *****, or **#**	Enter 1 to 4 digits.

Second Announcement Extension

Appears only when the **ACD** and **Queue** fields both are **y** and the **Vector** field is **n**.

Note:
If entering a Multi-Location Dial Plan shortened extension, note the following: When entering a Multi-Location Dial Plan shortened extension in a field designed for announcement extensions, certain administration end validations that are normally performed on announcement extensions are not done, and resultant warnings or submittal denials do not occur. The shortened extensions also do not appear in any display or list that shows announcement extensions. Extra care should be taken to administer the correct type of announcement for the application when assigning shortened extensions.

Valid entries	Usage
Valid extension	Enter the extension number assigned to a recorded announcement.
blank	Leaving the field blank indicates there is no second announcement.

Second Announcement Delay (sec)

Appears only when the **ACD** and **Queue** fields both are **y** and the **Vector** field is **n**. Enter the time in seconds before the call in the queue receives a second recorded announcement or that the second announcement is repeated.

Valid entries	Usage
1 to **99**	Avaya recommends that, if this split/skill or hunt group is a coverage point for another split/skill, this delay should not be more than 15 seconds.
blank	Leave blank if there is no second announcement.

Second Announcement Recurring

Appears only when the **ACD** and **Queue** fields both are **y** and the **Vector** field is **n**.

Valid entries	Usage
y	The second announcement can be repeated.
blank	Leave blank if there is no second announcement.

Send Reroute Request

Appears only when the **Message Center** field is type **QSIG-MWI** and **Supplementary Services with Rerouting** is **y** on the System Parameters Customer-Options (Optional Features) screen.

Valid entries	Usage
y	Rerouting is invoked. This is the default.
n	Rerouting is not invoked when a call covers through a qsig-mwi hunt group.

TSC per MWI Interrogation

Appears when the **Message Center** field is type **QSIG-MWI**. Use this field to control Temporary Signaling Connections (TSCs) used for message waiting interrogations for users that are “local” to the system in which the hunt group is administered.

Valid entries	Usage
y	Communication Manager brings the TSC up, executes the Interrogate operation, and then tears the TSC down.
n	Communication Manager utilizes the existing TSC sending FACILITY messages to request MWI status if the TSC is already set up, or sets up a TSC, and when the interrogation operation is complete, leaves the TSC up, subject to the existing timer. This is the default.

Voice Mail Extension

Appears if the **Message Center** field is **rem-vm**.

Valid entries	Usage
extension	Enter the UDP extension of the voice-mail hunt group on the host server running Communication Manager.

Voice Mail Handle

This field indicates the SIP Enablement Services (SES) handle that can receive voice mail. This field can be left blank if you supply a Voice Mail Number.

Voice Mail Number

Appears only if, on the System Parameters Customer-Options (Optional Features) screen, the **Basic Call Setup** and **Basic Supplementary Services** fields are **y** and the **Message Center** field on this screen is **qsig-mwi** or **fp-mwi**. The **qsig-mwi** selection shows the complete number of the AUDIX hunt group on the Message Center server for QSIG MWI. The **fp-mwi** selection shows the public network number of the AUDIX hunt group on the Message Center server.

Valid entries	Usage
Up to 17 digits	Enter the complete AUDIX dial-up number.

Field descriptions for pages 4 through X

Figure 122: Hunt Group Group Member Assignments screen

```
add hunt-group next                                                     Page x of x
                                 HUNT GROUP

  Group Number: 3        Group Extension: 1234567890123        Group Type:
ucd-mia

   Member Range Allowed: 1 - 1500          Administered Members (min/max): 0 /0
                                               Total Administered Members: 0

GROUP MEMBER ASSIGNMENTS
      Extension     Name                           Extension     Name
      1234567890123 1234567890123456789            1234567890123 1234567890123456789
1457:                                  1470:
1458:                                  1471:
1459:                                  1472:
1460:                                  1473:
1461:                                  1474:
1462:                                  1475:
1463:                                  1476:
1464:                                  1477:
1465:                                  1478:
1466:                                  1479:
1467:                                  1480:
1468:                                  1481:
1469:                                  1482:
```

Administered Members (min/max)

Appears on all member pages. Indicates the minimum and maximum member number administered for this hunt group.

At End of Member List

This display-only field shows the current page is also the last page.

Group Extension

This display-only field shows the extension of the hunt group.

Group Number

This display-only field shows the number of a hunt group.

Group Type

This display-only field shows the type of the hunt group.

Member Range Allowed

The range of allowed members displays on all member pages. These values vary depending on the particular system and/or configuration.

More Members Exist

This display-only field shows there are more members than currently displayed (the current page is not the last page).

Total Administered Members

Appears on all member pages. Indicates the total number of members administered for this hunt group.

GROUP MEMBER ASSIGNMENTS

Ext

A display-only field if the **Controlling Adjunct** field is **asai**. Controlled Agent extensions must be entered on the Adjunct Controlled Agent Table screen. The extension cannot be a VDN. The data module cannot be a member of an ACD split/skill. Use this field when the **Controlling Adjunct** field is **none**.

Valid entries	Usage
Valid extension.	Enter the assigned station or attendant console extension.

Name

This display-only field shows the name assigned to the above extension number when it is administered in the system.

Note:
Avaya BRI stations support ASCII characters only. They do not support non-ASCII characters, such as Eurofont or Kanafont. Therefore, if you use non-ASCII characters in any Communication Manager **Name** field, such characters do not display correctly on a BRI station.

Incoming Call Handling Treatment

Use the Incoming Call Handling Treatment screen to specify call handling for ISDN and SIP Enablement Services (SES) trunk groups. For more information on ISDN trunk groups, see ISDN Trunk Group.

Note:
This page does not appear if, on the ISDN Trunk Group screen, the **Digit Handling (in/out)** field is **overlap** on the "in" side or if the **Direction** field is **outgoing**.

With the Incoming Call Handling Treatment screen, you can specify unique call treatment for different incoming calls on any ISDN trunk group. The number of unique treatments that can be specified in this table and the number of pages vary depending on whether the Usage Allocation Enhancements feature is enabled and on the amount of available system memory.

Unique digit manipulation capabilities, CPN/BN requests, and night service destinations are possible for different types of incoming calls. The unique digit manipulation capabilities can be particularly useful to accommodate different dial plans for different services on an ISDN trunk type with a **Service Type** (field entry) of **cbc** (Call-by-Call). The table can also be used for ISDN trunk groups that are not Call-by-Call. For example, an ISDN group with **Service Type** set to **mega800** could use the Incoming Call Handling Treatment table to distinguish treatment of calls to different 800 numbers based on the Dialed Number Identification Service (DNIS) number that is incoming to Communication Manager.

Each row in the table consists of seven columns. The first three columns (**Service/Feature**, **Called Len**, and **Called Number**) constitute a key that together select which row or unique treatment should apply for an incoming call on the group. The remaining four columns (**Del**, **Insert**, and so on) specify the treatment to be provided for a call that matches the key.

If an incoming call is for a service listed in a row on the table, then that row can specify the treatment for the call, depending on the other two columns of the key. The **Called Len** field is used to continue the row determination. If the number of digits received with the incoming call matches the number of digits in the **Called Len** field for calls to the matched service, then this row might apply. If no other row also contains a matching service and called length, then this row does apply. If another row does exist with the same service and number length, then the **Called Number** field is used to continue the row determination.

If the leading digits received with the incoming call match the digits specified in the **Called Number** field, then this row applies to the call. Therefore, with this table, a unique treatment can be given to any incoming call, even if these calls are to the same service or have the same length of digits. The remaining four fields specify the unique treatment for the call once the row has been determined. Together, the **Del** and **Insert** fields can be used to manipulate the incoming number that is used to route the call. The **Per Call CPN/BN** field appears only for ISDN trunk groups, and can be used to request CPN/BN from AT&T networks for specific calls incoming on the group. The **Night Serv** field also appears only for ISDN trunk groups, and is used to have calls of different types routed to different night destinations when night service is in effect.

The Incoming Call Handling Treatment Table always automatically rearranges to show the precedence order the system uses to pick an entry. Thus, you can easily predict the behavior of the Incoming Call Handling Treatment Table by noting the order in which the entries display. (The entries rearrange after submitting the Trunk Group screen. A subsequent **`change trunk-group`** or **`display trunk-group`** command then shows the rearranged order.)

Communication Manager traverses the table from top to bottom and picks the first entry that matches all the following criteria:

- The Service /Feature, if applicable, matches
- The Called/Length matches
- The Called Number matches

If the administered **Called Length** or **Called Number** is blank, that criterion is considered successful.

Incoming Call Handling Treatment Table entries with a predefined service/feature always appear before entries with a user-defined service/feature. To control the order in which certain entries appear, you must use user-defined services/features for those entries. For example, you can redefine the predefined mega800 service/feature as an identical user-defined entry with the name m800.

User-defined entries are always listed in the reverse order compared to the way they appear on the **Network Facilities** screen. Thus, given two user-defined services/features ABC and XYZ, you can force XYZ to appear before ABC in an Incoming Call Handling Treatment Table by putting **XYZ** after **ABC** on the **Network Facilities** screen.

Note:

DCS features that use the **remote-tgs** button (on the remote server/switch) do not work when the local trunk group deletes or inserts digits on the incoming call. These buttons try to dial a local TAC. Adding or deleting digits defeats this operation and renders the remote feature inoperable. If digit manipulation is needed, use it on the outgoing side, based on the routing pattern. One reason for digit manipulation is insertion of the AAR feature access code (FAC).

These fields are located on the Incoming Call Handling Treatment Table screen.

Field descriptions for page 1

Figure 123: Incoming Call Handling Treatment screen (ISDN)

```
change inc-call-handling-trmt trunk-group 1                         Page 1 of x
                      INCOMING CALL HANDLING TREATMENT

Service/      Called Called          Del Insert            Per Call Night
Feature        Len   Number                                CPN/BN   Service
123456789012345 12 1234567890123456 123 1234567890123456 12345678 1234567890123
tie
tie
tie
tie
tie
tie
tie
tie
tie
tie
tie
tie
tie
tie
tie
tie
tie
```

Note:
For SIP Enablement Services (SES) trunk groups, the **Per Call CPN/BN** and **Night Serv** fields do not appear, as shown in Figure 124.

Figure 124: Incoming Call Handling Treatment screen (SIP)

```
change inc-call-handling-trmt trunk-group1                  Page 1 of X

                        INCOMING CALL HANDLING TREATMENT

     Service/   Called     Called         Del      Insert
     Feature     Len       Number
  _____________  ___   _______________    ___   ______________
  _____________  ___   _______________    ___   ______________
  _____________  ___   _______________    ___   ______________
  _____________  ___   _______________    ___   ______________
  _____________  ___   _______________    ___   ______________
  _____________  ___   _______________    ___   ______________
  _____________  ___   _______________    ___   ______________
  _____________  ___   _______________    ___   ______________
  _____________  ___   _______________    ___   ______________
  _____________  ___   _______________    ___   ______________
  _____________  ___   _______________    ___   ______________
  _____________  ___   _______________    ___   ______________
  _____________  ___   _______________    ___   ______________
  _____________  ___   _______________    ___   ______________
  _____________  ___   _______________    ___   ______________
  _____________  ___   _______________    ___   ______________
  _____________  ___   _______________    ___   ______________
  _____________  ___   _______________    ___   ______________
```

Called Len

Specifies the number of digits received for an incoming call. A blank entry may be used only for the situation in which the Called Number field has been set to blank. When used with the blank entry, this means that any length of digits associated with the Called Party IE of the Incoming SETUP message matches this field. The use of the **0** entry is encouraged for the situation in which the PSTN provider does not provide any 'Number Digits' within the received Called Party IE (such as in Japan). Valid entries are **0** to **21**, or leave blank.

Called Number

Specifies the leading digits received for an incoming call. A blank entry is used as a "wild card" entry and, when used, means that any number associated with the specified Service/Feature can match in this field. Valid entries are up to 16 digits, or leave blank.

Del

Specifies the number of leading digits to be deleted from the incoming Called Party Number. Calls of a particular type can be administered to be routed to a single destination by deleting all incoming digits and then administering the **Insert** field with the desired extension. Valid entries are **1** to **21**, **all**, or leave blank.

Insert

Specifies the digits to be prepended to the front of the remaining digits after any (optional) digit deletion has been performed. The resultant number formed from digit deletion/insertion is used to route the call, provided night service is not in effect. Valid entries are up to 16 characters consisting of a combination from the following: **0** to **9**, *****, **#**, or leave blank.

Per Call CPN/BN

This field appears only for ISDN trunk groups. Specifies when and how to request Calling Party Number (CPN) or Billing Number (BN) for calls of this type. Leave blank when connected to another Avaya S8XXX Server, or when connected to a public network outside North America. Within North America, leave blank when connected to a public network that does not permit customer equipment to request CPN or BN for individual incoming calls. The AT&T Switched Network offers this service under the titles CPN/BN to Terminating End on a Per-Call Basis and ANI (BN) on Request. An entry of **none** indicates Communication Manager will not request either CPN or BN for any incoming calls of this type. Valid entries are **cpn-only**, **bn-only**, **bn-pref** (prefer BN, but accepts CPN), **cpn-pref** (prefer CPN, but accepts BN), **none**, or leave blank. Leave blank when connected to another server/switch or to a network other than the AT&T Switched Network.

Note:
A 4-second delay occurs in terminating the call to the far-end station if the connecting server or switch does not respond to the request.

Night Serv

This field appears only for ISDN trunk groups. Specifies a night service extension (can be a VDN extension) per Service/Feature. An entry other than blank overrides **Night Service** entry on page 1 of the screen. This entry can be overridden by the **Trunk/Member Night Service** entry when provided. Valid entries are an assigned extension, the attendant group access code (**attd**), or leave blank.

Service/Feature

This field is display-only. It is auto-populated with the value entered in the **Service Type** field on the Trunk Group screen.

Note:

An exception occurs when **cbc** is the value in the **Service Type** field on the Trunk Group screen. Because there are several possible values for the **Service/ Feature** field for cbc trunk groups, the field is not display-only, but is available for user entry. Valid **Service/Feature** values for cbc trunk groups can be viewed on the **Network Facilities** screen.

Note also that in addition to pre-defined Services/Features, any user-defined **Facility Type** of **0** (feature), **1** (service), or **2** (incoming) on the **Network Facilities** screen is allowed. For a Service/Feature defined as Type 2, it is this screen that determines which incoming calls are assigned to this Service/ Feature. See the description of the **Network Facilities** screen for details.

Integrated Announcement Boards

You can move integrated announcement boards that have been previously administered on the Announcements/Audio Sources screen to a new board location. You can also display a list of all administered integrated announcement circuit packs.

Field descriptions for page 1

Figure 125: Integrated Announcement Boards screen

```
display integrated-annc-boards                                  Page 1 of x

                         INTEGRATED ANNOUNCEMENT BOARDS
    Last Board Location Saved:

  Board                       Time                   Number of
  Location         Sfx        Remaining     Rate     Recordings        Checksum ID

1:01D09            C          206           32            7            047a
2:02817            C          248           32            1            00db
3:
4:
5:
```

Last Board Location Saved

Valid entries	Usage
Display-only	Applies to TN750 only; not applicable to VAL

Board Location

Valid entries	Usage
Display-only	The physical location of the integrated announcement circuit pack (UUCSS).

Sfx

Valid entries	Usage
Display-only	The circuit pack suffix letter(s).

Time Remaining

Valid entries	Usage
Display-only	The amount of recording time in seconds remaining on the circuit pack at the 64Kb rate.

Rate

Valid entries	Usage
Display-only	The announcement's compression rate.

Number of Recordings

Valid entries	Usage
Display-only	The number of nonzero-length announcement recordings or files on the circuit pack.

Checksum ID

Valid entries	Usage
Display-only	Applies to TN750 only; not applicable to VAL.

Integrated Announcement Translations

Use this screen to change board locations currently administered on the Announcements/Audio Sources screen to a new board location.

Field descriptions for page 1

Figure 126: Change Integrated Announcement Translations screen

```
change integ-annc-brd-loc                                    Page 1 of x

                      CHANGE INTEGRATED ANNOUNCEMENT TRANSLATIONS
          Change all board location translations from board:     to board:


Changing board locations using this command will change
all currently administered "from" board locations on the
Announcements/Audio Sources screen to the "to" board location.
```

Change all board location translations from board

Valid entries	Usage
board; cabinet 1 to 3; carrier A-E; slot 1 to 20; or gateway 1 to 10; module V1 to V9	Enter a VAL board that is currently administered on the Announcements/Audio Sources screen.

to board

Valid entries	Usage
board; cabinet 1 to 3; carrier A to E; slot 1 to 20; or gateway 1 to 10; module V1 to V9	Enter a VAL board to which you want to move announcement translations that are currently administered on the Announcements/Audio Sources screen.

Intercom Group

This screen assigns extensions to intercom groups.

Field descriptions for page 1

Figure 127: Intercom Group screen

```
change intercom-group n                                         Page 1 of x
                                  INTERCOM GROUP
                                                Group Number: n
                                         Length of Dial Code: _

GROUP MEMBER ASSIGNMENTS
              Ext      DC    Name
           1: ________ __
           2: ________ __
           3: ________ __
           4: ________ __
           5: ________ __
           6: ________ __
           7: ________ __
           8: ________ __
           9: ________ __
          10: ________ __
          11: ________ __
          12: ________ __
          13: ________ __
          14: ________ __
          15: ________ __
          16: ________ __
```

DC

This field assigns a dial code to an extension. The dial code is the code users must dial to make intercom calls to the corresponding extension.

Valid entries	Usage
1 or 2-digit code	The number of digits entered must exactly match the number assigned in the **Length of Dial Code** field. For example, if the **Length of Dial Code** field is set to 2, you must type 1 as 01 in the **DC** field. This field cannot be blank.

Ext

This field assigns an extension to the group.

Valid entries	Usage
an extension number	Enter a physical extension number. You cannot enter a VDN in this field.

Group Number

This display-only field shows the group's ID number.

Length of Dial Code

This field sets the number of digits that users must dial to access an extension in the group. (On Page 2, this is a display-only field.

Valid entries	Usage
1	Enter **1** if there are 9 or fewer members.
2	Enter **2** if there are 10 or more members.

Name

Display-only field. The server running Communication Manager fills in this field with the name from the Station screen.

Note:
Avaya BRI stations support ASCII characters only. They do not support non-ASCII characters, such as Eurofont or Kanafont. Therefore, if you use non-ASCII characters in any Communication Manager **Name** field, such characters do not display correctly on a BRI station.

Related topics

See Intercom in *Avaya Aura™ Communication Manager Feature Description and Implementation,* 555-245-205, for more information.

Inter-Exchange Carrier (IXC) Codes

This screen allows identification of the IXC in the CDR record.

Field descriptions for page 1

Figure 128: Inter-Exchange Carrier Codes screen

```
change ixc-codes                                                    Page 1 of x
                      INTER-EXCHANGE CARRIER CODES
       IXC Codes Assignments (Enter up to 15)
       CDR   IXC                              CDR   IXC
       IXC   Access                           IXC   Access
       Code  Number    IXC Name               Code  Number    IXC Name
       1:    _______   _______________         9:   _______   _______________
       2:    _______   _______________        10:   _______   _______________
       3:    _______   _______________        11:   _______   _______________
       4:    _______   _______________        12:   _______   _______________
       5:    _______   _______________        13:   _______   _______________
       6:    _______   _______________        14:   _______   _______________
       7:    _______   _______________        15:   _______   _______________
       8:    _______   _______________
```

IXC Access Number

Valid entries	Usage
2 to 11 digits, **0** to **9** and *****	Enter the digits dialed or inserted by AAR/ARS into the outpulsed digit string to access the interexchange carrier. No duplicate access numbers are allowed in the table.

IXC Name

Valid entries	Usage
0 to 15 characters	Description to identify the IXC

Field descriptions for page 2

Figure 129: Inter-Exchange Carrier Codes screen

```
change ixc-codes                                            Page 2 of x
                             IXC Prefix            IXC Code Format
                        1.   ___                   _____
                        2.   ___                   _____
                        3.   ___                   _____
                        4.   ___                   _____
                        5.   ___                   _____
```

IXC Code Format

Valid entries	Usage
1 to 4 digit code format	
*****	
x	
X	

Valid entries	Usage
xxxx	For line 1
xxx	For line 2

IXC Prefix

Valid entries	Usage
1 to 3 digit prefix	
*	
101	For line 1
10	For line 2

Intra-Switch CDR

This screen administers extensions for which Intra-Switch CDR is to be enabled.

Note:
Attendants are not allowed to be optioned for the Intra-Switch CDR feature.

When you enter the `add` command to add extensions, the system automatically begins after the last administered extensions. If you enter the `change` command, the system display begins with the first extension. If you enter the `change` command with an extension number, the system begins the display with that extension.

When you enter the command `list intra-switch-cdr <extension> count x`, the system lists "x" switch extensions administered for Intra-Switch CDR beginning with the extension specified by <extension>. For example, if you enter `list intra-switch-cdr 81000 count 500`, the system displays extension 81000 (if it is administered for Intra-Switch CDR) and the next 500 extensions that are administered for Intra-Switch CDR. The `display` command functions similarly to the `change` command.

Capacities

The Intra-Switch CDR extension capacities vary from server to server. For more information, see the *Avaya Aura™ Communication Manager Hardware Description and Reference*, 555-245-207.

Field descriptions for page 1

Figure 130: Intra-Switch CDR screen

```
change intra-switch-cdr                                          Page   1 of   3
                                INTRA-SWITCH CDR

                  Assigned Members: 0 of 5000 administered
      Extension           Extension          Extension           Extension
    _____________       _____________      _____________       _____________
    _____________       _____________      _____________       _____________
    _____________       _____________      _____________       _____________
    _____________       _____________      _____________       _____________
    _____________       _____________      _____________       _____________
    _____________       _____________      _____________       _____________
    _____________       _____________      _____________       _____________
    _____________       _____________      _____________       _____________
    _____________       _____________      _____________       _____________
    _____________       _____________      _____________       _____________
    _____________       _____________      _____________       _____________
    _____________       _____________      _____________       _____________
    _____________       _____________      _____________       _____________
    _____________       _____________      _____________       _____________
    _____________       _____________      _____________       _____________

Use 'list intra-switch-cdr' to see all members.
Use 'change intra-switch-cdr <ext>' to add/change/remove other members.
```

Assigned Members

Displays the number of extensions currently administered for Intra-switch CDR.

Extension

Valid entries	Usage
Any valid extension	Enter the local extensions you want to track with Intra-Switch CDR. The number of extensions you can track might vary from one system to the next.

IP Address Mapping

This screen defines feature characteristics that depend on the IP address.

Note:
Enter data in either the **To IP Address** field or the **Subnet Mask** field.

Field descriptions for page 1

Figure 131: IP Address Mapping screen

```
change ip-network-map                                           Page   1 of  x
                               IP ADDRESS MAPPING

                                               Subnet Network     Emergency
 IP Address                                    Bits   Region VLAN Location Ext
 --------------------------------------------- ------ ------ ---- -------------
 FROM:                                         /             n
   TO:
 FROM:                                         /             n
   TO:
 FROM:                                         /             n
   TO:
 FROM:                                         /             n
   TO:
 FROM:                                         /             n
   TO:
 FROM:                                         /             n
   TO:
 FROM:                                         /             n
   TO:
 FROM:                                         /             n
   TO:
```

Emergency Location Extension

This field allows the system to properly identify the location of a caller who dials a 911 emergency call from this station. An entry in this field must be of an extension type included in the dial plan, but does not have to be an extension on the local system. It can be a UDP extension. The entry defaults to blank. A blank entry typically would be used for an IP softphone dialing in through PPP from somewhere outside your network.

If you populate the IP Address Mapping screen with emergency numbers, the feature functions as follows:

- If the **Emergency Location Extension** field in the Station screen is the same as the **Emergency Location Extension** field in the IP Address Mapping screen, the feature sends the extension to the Public Safety Answering Point (PSAP).
- If the **Emergency Location Extension** field in the Station screen is different from the **Emergency Location Extension** field in the IP Address Mapping screen, the feature sends the extension in the IP Address Mapping screen to the Public Safety Answering Point (PSAP).

Valid entries	Usage
0 to **9**	Enter the emergency location extension for this station. Default is blank.

Note:
On the ARS Digit Analysis Table screen, you must administer 911 to be call type **emer** or **alrt** in order for the E911 Emergency feature to work properly.

From IP Address

Defines the starting IP address.

Valid entries	Usage
32-bit address (4 decimal numbers, each in the range 0 to 255) also supports IPv6 addressing format	See *Administering Network Connectivity on Avaya Aura™ Communication Manager*, 555-233-504, for more information.

To IP Address

Defines the termination of a range of IP addresses.

If this field and the **Subnet Bits** fields are blank when submitted, the address in the **From IP Address** field is copied into this field.

The **Subnet Bits** field data is applied to the **From** field, creating the converted **To IP Address** field information.

Valid entries	Usage
32-bit address (4 decimal numbers, each in the range 0 to 255) also supports IPv6 addressing format	See *Administering Network Connectivity on Avaya Aura™ Communication Manager*, 555-233-504, for more information.

To IP Address or Subnet Bits

The end of the IP address range can be specified by either entering the last IP address in the range or the **From IP Address** and the number of bits of the subnet mask.

If the **Subnet Bits** field is used, then:

- The mask is applied to the **From IP Address** field, placing zeros in the non-masked rightmost bits. This becomes the stored "From" address.
- The mask is applied to the **To IP Address** field, placing 1's in the non-masked rightmost bits. This becomes the stored "To" address.

If this field and the **To IP Address** fields are blank when submitted, the address in the **From IP Address** field is copied into the **To IP Address** field.

Valid entries	Usage
0 to **64** or blank	Enter the last IP address in the range or the **From IP Address** and the number of bits of the subnet mask.

Network Region

Identifies the network region for the IP address range. For SIP, the value for this field must correlate with the configured network region for this range of addresses.

Valid entries	Usage
1 to **250**	Enter the network region number for this interface. This field must contain a non-blank value if the **From IP Address** field on the same row contains a non-blank value.

VLAN

Sends VLAN instructions to IP endpoints such as IP telephones and softphones. This field does not send VLAN instructions to the PROCR (S8300/S87XX Servers), CLAN, and Media Processor boards.

Valid entries	Usage
0 to **4094**	Specifies the virtual LAN value.
n	Disabled

IP Codec Set

The IP Codec Set screen allows you to specify the type of codec used for voice encoding and companding (compression/decompression). The main difference between codecs is in the compression algorithm used; some codecs compress the voice data more than others. A greater degree of compression results in lower bandwidth requirements on the network, but might also introduce transmission delays and lower voice quality. Codecs are used for VoIP links between any two VoIP resources or endpoints, for example, IP telephone to IP telephone, IP telephone to Media Gateway, Media Gateway to Media Gateway, etc. The order in which the codecs are listed on this screen is the order of your preference of usage. A trunk call between any two VoIP resources or endpoints is set up to use the first common codec listed.

The default codec is set for G711MU. The G711MU provides the highest voice quality because it does the least amount of compression, but it uses the most bandwidth. The G711MU default setting can be changed to one of two other codecs (and their “flavors”) if the G711MU does not meet your desired voice-quality/bandwidth trade-off specification. For example, if a far-end server is not running Communication Manager, you might need to change the codec to match one that is supported by that server’s software.

Field descriptions for page 1

This screen allows you to define the allowed codecs and packet sizes used between VoIP resources. You can also enable silence suppression on a per-codec basis. This screen dynamically displays the packet size in milliseconds for each codec in the set, based on the number of frames you administer per packet.

Figure 132: IP Codec Set screen - page 1

```
change ip-codec-set n                                          Page 1 of x

                              IP Codec Set

      Codec Set: 1

     Audio         Silence        Frames    Packet
     Codec         Suppression    Per Pkt   Size (ms)
1: G.711MU_______     y              3         30
2:_______________     _              _
3:_______________     _              _
4:_______________     _              _
5:_______________     _              _
6:_______________     _              _
7:_______________     _              _

Media Encryption:
1: aes
2: aea
3: srtp-aescm128-hmac80
```

Audio Codec

Specify the audio codec used for this codec set.

Valid entries	Usage
G.711A (a-law) **G.711MU** (mu-law) **G.722-64k** **G.722.1-24k** **G.722.1-32k** **G.723-5.3** **G.723-6.3** **G.726A-32K** **G.729** **G.729A** **G.729B** **G.729AB** **SIREN14-24k** **SIREN14-32k** **SIREN14-48k** **SIREN14-S48k** **SIREN14-S56k** **SIREN14-S64k** **SIREN14-S96k**	Enter the codec to be used for this codec set.

Important:
Avaya recommends that you include at least two codecs for every telephone in order to avoid incompatible codecs. Use the codecs specified in the following table for the telephones shown.

Telephone	Codec to use
All Avaya IP Telephones	**G.711**, **G.729B**
4601 4602 4602SW 4620SW 4621SW 4622SW	add **G.726A** (requires firmware R2.2)

Codec Set

Display only. Shows the number assigned to this Codec Set.

Frames Per Pkt

Specify the number of frames per packet up to a packet size of 60 milliseconds (ms).

Valid entries	Usage
1 to **6** or blank	Default frame sizes for codecs: ● G.711 and G.729: 2 frames (20 ms) ● G.723: 3 frames (30 ms) ● G.726A: 1 frame (10 ms)

Media Encryption

This field appears only if the **Media Encryption over IP** feature is enabled in the license file. Use this field to specify a priority listing of the three possible options for the negotiation of encryption. Communication Manager attempts to provide bearer encryption per this administered priority order.The selected option for an IP codec set applies to all codecs defined in that set.

Valid entries	Usage
aes	Advanced Encryption Standard (AES), a standard cryptographic algorithm for use by U.S. government organizations to protect sensitive (unclassified) information. Use this option to encrypt these links: ● Server-to-gateway (H.248) ● Gateway-to-endpoint (H.323)
aea	Avaya Encryption Algorithm. Use this option as an alternative to AES encryption when: ● All endpoints within a network region using this codec set must be encrypted. ● All endpoints communicating between two network regions and administered to use this codec set must be encrypted.
	1 of 2

Valid entries	Usage
	SRTP is a media encryption standard defined in RFC 3711 as a profile of RTP. Communication Manager 4.0 supports the following functionality as given in RFC 3711: ● Encryption of RTP (optional but recommended) ● Authentication of RTCP streams (mandatory) ● Authentication of RTP streams (optional but recommended) ● Protection against replay **Note:** In Communication Manager 4.0, SRTP encryption is supported by 96xx telephones only.
1-srtp-aescm128-hmac80	1-Encrypted/Authenticated RTP with 80-bit authentication tag
2-srtp-aescm128-hmac32	2-Encrypted/Authenticated RTP with 32-bit authentication tag
3-srtp-aescm128-hmac80-unauth	3-Encrypted RTP but not authenticated
4-srtp-aescm128-hmac32-unauth	4-Encrypted RTP but not authenticated
5-srtp-aescm128-hmac80-unenc	5-Authenticated RTP with 80-bit authentication tag but not encrypted
6-srtp-aescm128-hmac32-unenc	6-Authenticated RTP with 32-bit authentication tag but not encrypted
7-srtp-aescm128-hmac80-unenc-unauth	7-Unencrypted/Unauthenticated RTP
8-srtp-aescm128-hmac32-unenc-unauth	8-Unencrypted/Unauthenticated RTP **Note:** For stations, the only value supported is **srtp-aescm128-hmac80**. H.323 IP trunks support all eight of the listed algorithms.
none	Media stream is unencrypted. This is the default.
	2 of 2

Packet Size (ms)

A display-only field showing the packet size in milliseconds.

Silence Suppression

Enables RTP-level silence suppression on the audio stream.

Valid entries	Usage
y/n	Enter **y** to enable RTP-level silence suppression on the audio stream.

Field descriptions for page 2

Use this screen to assign the following characteristics to a codec set:

- Whether or not Direct-IP Multimedia is enabled for videophone transmissions
- Whether or not endpoints in the assigned network region can route fax, modem, or TTY calls over IP trunks

Note:
For more information on modem/fax/TTY over IP, see *Administering Network Connectivity for Avaya Aura™ Communication Manager*, 555-233-504.

- Which mode the system uses to route the fax, modem, or TTY calls
- Whether or not redundant packets is added to the transmission for higher reliability and quality

These characteristics must be assigned to the codec set, and the codec set must be assigned to a network region for endpoints in that region to be able to use the capabilities established on this screen.

CAUTION:
If users are using Super G3 fax machines as well as modems, do *not* assign these fax machines to a network region with an IP Codec set that is modem-enabled as well as fax-enabled. If its Codec set is enabled for both modem and fax signaling, a Super G3 fax machine incorrectly tries to use the modem transmission instead of the fax transmission.

Therefore, assign modem endpoints to a network region that uses a modem-enabled IP Codec set, and assign the Super G3 fax machines to a network region that uses a fax-enabled IP Codec set.

Note:
Transporting modem tones over IP between Communication Manager systems is a proprietary implementation. Also, FAX transport implementations, other than T.38 are proprietary implementations.

Figure 133: IP Codec Set screen page 2

```
change ip-codec-set n                                        Page   2 of   x

                             IP Codec Set

                                 Allow Direct-IP Multimedia? y
        Maximum Bandwidth Per Call for Direct-IP Multimedia: 256:Kbits

                    Mode        Redundancy

            FAX     relay          0

          Modem     off            0

        TDD/TTY     us             0

  Clear-channel     n              0
```

Allow Direct-IP Multimedia

Valid entries	Usage
y/n	Enter **y** to allow direct multimedia via the following codecs: ● H.261 ● H.263 ● H.264 (video) ● H.224 ● H.224.1 (data, far-end camera control).

Clear-channel

For more information on Clear Channel, see *Administering Network Connectivity on Avaya Aura™ Communication Manager*, 555-233-504.

Valid entries	Usage
y/n	Enter **y** to indicate that this codec set supports BRI data calls. **Note:** Clear Channel data transmission is supported on the TN2602AP IP Media Resource 320 circuit pack and the TN2302AP circuit pack.

FAX Mode

Valid entries	Usage
off	Turn off special fax handling when using this codec set. In this case, the fax is treated like an ordinary voice call. With a codec set that uses G.711, this setting is required to send faxes to non-Avaya systems that do not support T.38 fax.
relay	For users in regions using this codec, use Avaya relay mode for fax transmissions over IP network facilities. This is the default for new installations and upgrades to Communication Manager R2.1.
pass-through	For users in regions using this codec, use pass-through mode for fax transmissions over IP network facilities. This mode uses G.711-like encoding.
t.38-standard	For users in regions using this codec, use T.38 standard signaling for fax transmissions over IP network facilities.

Note:
If you have a telephone that is on an IP trunk too close to a fax machine, the handset can pick up the tones from the fax machine and change itself into the fax mode. To prevent this, set the **FAX** field to **off**, and put the FAX machines in an ARS partition that uses only circuit switched trunks, even for IGW FAX calls.

Maximum Bandwidth Per Call for Direct-IP Multimedia (value)

This field appears only when **Allow Direct-IP Multimedia** is **y**.

Valid entries	Usage
1 to **9999**	Enter the bandwidth limit for Direct-IP Multimedia transmissions on this codec set. Default is **256**.

Maximum Bandwidth Per Call for Direct-IP Multimedia (units)

This field displays only when **Allow Direct-IP Multimedia** is **y**.

Valid entries	Usage
kbits mbits	Enter the unit of measure corresponding to the value entered for bandwidth limitation. Default is **kbits**.

Modem Mode

Valid entries	Usage
off	Turn off special modem handling when using this codec set. In this case, the modem transmission is treated like an ordinary voice call. This is the default for new installations and upgrades to Communication Manager R2.1. With a codec set that uses G.711, this setting is required to send modem calls to non-Avaya systems.
relay	For users in regions using this codec, use relay mode for modem transmissions over IP network facilities. Avaya V.32/FNBDT Modem Relay is supported when using modem **relay** mode. **Note:** Modem over VoIP in relay mode is currently available only for use by specific analog telephones that serve as Secure Telephone Units (STUs). Contact your Avaya technical support representative for more information.
pass-through	For users in regions using this codec, use pass-through mode for modem transmissions over IP network facilities. Avaya V.8 Modem Pass-Thru is supported when using modem **pass-through** mode.

Redundancy

Valid entries	Usage
0 to **3**	Enter the number of duplicate or redundant packets that are sent in addition to the primary packet for all Modes except **pass-through** and **Clear-channel**. The default is **0**.

TDD/TTY Mode

Valid entries	Usage
off	Turn off special TTY handling when using this codec set. In this case, the TTY transmission is treated like an ordinary voice call. With a codec set that uses G.711, this setting is required to send TTY calls to non-Avaya systems. However, there might be errors in character transmissions.
US	For users in regions using this codec, use U.S. Baudot 45.45 mode for TTY transmissions over IP network facilities. This is the default for new installations and upgrades to Communication Manager R2.1.
	1 of 2

Valid entries	Usage
UK	For users in regions using this codec, use U.K. Baudot 50 mode for TTY transmissions over IP network facilities.
pass-through	For users in regions using this codec, use pass-through mode for TTY transmissions over IP network facilities.
	2 of 2

IP Interfaces

Use the IP Interfaces screen to assign a network region to an IP interface device, or to administer Ethernet options. The fields shown appear when the `add`, `change`, `display`, or `remove` command is used.

Note:
For information about Processor Ethernet interfaces, see *Administering Avaya Aura™ Communication Manager*, 03-300509.

The appearance of the IP Interfaces screen can vary according to the interface type you are administering, and your system's configuration. The figures shown are examples, intended to show most of the fields that might appear on this screen. Your own screen might vary from these examples. The list of field descriptions that follows the figures is intended to be comprehensive, and to include information on all fields that might appear. The field descriptions are in alphabetical order for quick reference.

Note:
When you start the process of administering the IP interface for the TN2602AP circuit pack, any active calls continue to use the TN2602AP circuit pack's physical IP address for the connection, not the virtual IP address you are setting in this procedure. Therefore, any of these calls, if they continue after you complete this procedure, will drop in the event of an interchange.

Field descriptions for page 1

Figure 134: IP Interfaces - Type board location screen

```
change ip-interface 01a03                                       Page   1 of   x
                                  IP INTERFACES


                  Type: MEDPRO
                  Slot: 01A03
           Code/Suffix: TN2302
      Enable Interface? y
                  VLAN: n
        Network Region: 1


                                 IPV4 PARAMETERS
             Node Name: medpro
           Subnet Mask: /24
     Gateway Node Name: Gateway001
```

Figure 135: IP Interfaces - Type procr screen

```
add ip-interface procr                                          Page   1 of   x
                                  IP INTERFACES


                  Type: PROCR
                                                     Target socket load: 19200

      Enable Interface? n                         Allow H.323 Endpoints? y
                                                   Allow H.248 Gateways? y
        Network Region:                             Gatekeeper Priority: 5



                                 IPV4 PARAMETERS
             Node Name: procr
           Subnet Mask: /24
```

Figure 136: IP Interfaces - Type VAL screen

```
change ip-interface 01a09                                       Page   1 of   x
                                  IP INTERFACES


                  Type: VAL
                  Slot: 01A09
           Code/Suffix: TN2501
      Enable Interface? y





                                 IPV4 PARAMETERS
             Node Name: TofuVAL172-1
           Subnet Mask: /19
     Gateway Node Name: Gateway001

         Ethernet Link: 13
         Network uses 1's for Broadcast Addresses? y
```

Figure 137: IP Interfaces - Type C-LAN screen

```
change ip-interface 01a02                                       Page   1 of   x
                                  IP INTERFACES


                  Type: C-LAN
                  Slot: 01A02        Target socket load and Warning level: 400
           Code/Suffix: TN799  D           Receive Buffer TCP Window Size: 8320
      Enable Interface? y                           Allow H.323 Endpoints? y
                  VLAN: n                            Allow H.248 Gateways? y
        Network Region: 1                             Gatekeeper Priority: 5



                                 IPV4 PARAMETERS
             Node Name: clan
           Subnet Mask: /24
     Gateway Node Name: Gateway001

         Ethernet Link: 1
         Network uses 1's for Broadcast Addresses? y
```

Allow H.248 Gateways

This field controls whether or not H.248 media gateways (G7000, G350, G250) can register on the interface.

Valid entries	Usage
y/n	On a single main server, enter **y** to allow H.248 endpoint connectivity to the PE interface. Enter **n** if you do not want H.248 endpoint connectivity to the PE interface. **Note:** For an Enterprise Survivable Server (ESS), this field is display-only and is set to **n**. H.248 endpoint connectivity using the PE interface on an ESS server is not supported. For a Local Survivable Processor (LSP), this field is display-only and is set to **y**.

Allow H.323 Endpoints

This field controls whether or not IP endpoints can register on the interface.

Valid entries	Usage
y/n	On a single main server, enter **y** to allow H.323 endpoint connectivity to the PE interface. Enter **n** if you do not want H.323 endpoint connectivity to the PE interface. **Note:** For an Enterprise Survivable Server (ESS), this field is display-only and is set to **n**. H.323 endpoint connectivity using the PE interface on an ESS server is not supported. For a Local Survivable Processor (LSP), this field is display-only and is set to **y**.

Code/Suffix

Valid entries	Usage
y/n	Circuit pack TN code and suffix. Display-only for TN2602AP when **Critical Reliable Bearer** is **n**. The second (right-side) **Code/Sfx** field is automatically populated based on the corresponding **Slot** field information, when **Critical Reliable Bearer** is **y**. **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.

Critical Reliable Bearer

Appears when the board code of slot location is TN2602.

Valid entries	Usage
y/n	A **y** entry indicates that two TN2602AP circuit packs are duplicated in a port. If **y**, a second column of information appears, for administering the second shared circuit pack. Default is **n**. **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.

Enable Interface?

Allows use of the Ethernet port.

Valid entries	Usage
y/n	Enter **y** to indicate that the Ethernet port associated with the TN2602AP circuit pack is in service. If this is an active board, set to **n** only when there is no standby, or when the standby has been disabled. **Note:** Enter **n** in this field before you make changes to the screen.

Ethernet Link

This display-only field shows the administered link number for an Ethernet link.

Valid entries	Usage
y/n	Shows the unique number for the Ethernet link assigned on the **Data Module** screen.

Gateway Node Name

Valid entries	Usage
Character string (up to 15 characters max.)	Enter the gateway node name associated with the IP address of the LAN gateway associated with the TN2602AP. This entry also applies to the second TN2602AP circuit pack when **Critical Reliable Bearer** is **y**. **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.

Gatekeeper Priority

Appears only if **Allow H.323 Endpoints** is **y** and the Communication Manager server is a main server or an LSP. This field does not display on an ESS server. This field allows a priority to be set on the interface. This affects where the interface appears on the gatekeeper list.

Valid entries	Usage
1 to **9**	Enter the desired priority number. The value in this field is used on the alternate gatekeeper list. The lower the number, the higher the priority. Default is **5**.

Network Region

Identifies the network region for the specified interface.

Valid entries	Usage
1 to **250**	Enter the value of the Network Region where the TN2602AP resides. This entry also applies to the second TN2602AP circuit pack when **Critical Reliable Bearer** is **y**. **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.

Network uses 1's for Broadnet Addresses

Valid entries	Usage
y/n	Enter **y** to indicate that a broadcast address is used to send the same message to all systems/clients on a local area network.

Node Name

The unique node name for the IP interface administered on the **Node Names** screen.

Valid entries	Usage
Character string (up to 15 characters max.)	Enter the node name associated with the IP address of the TN2602AP circuit pack.

Receive Buffer TCP Window Size

Valid entries	Usage
512 to **8320**	The number of bytes allotted for the buffer that receives TCP data for a TN799 (CLAN) circuit pack. The default is **512**.

Slot

Displays the slot location entered in the command line. Enter the location of the second TN2602AP circuit pack for a non-duplicated board. The second (right-side) **Slot** field is automatically populated when **Critical Reliable Bearer** is **y**.

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.

Valid entries	Usage
01 to **03** (DEFINITY CSI) or **1** to **64** (S87XX/S8300 Servers)	First and second characters are the cabinet number.

Valid entries	Usage
A to **E**	Third character is the carrier.
0 to **20**	Fourth and fifth character are the slot number.

Subnet Mask

The subnet mask is a 32-bit binary number that divides the network ID and the host ID in an IP address.

Valid entries	Usage
characters	Enter the Subnet Mask for TN2602AP. This entry also applies to the second TN2602AP circuit pack when **Critical Reliable Bearer** is **y**. **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.

Target socket load

This field appears when **Type** is **procr**. Use this field for load balancing endpoint traffic across multiple IP interfaces. The value that you enter in the **Target socket load** field controls the percentage of sockets allocated to each IP interface within the same Gatekeeper Priority. When all the IP interfaces within the same Gatekeeper Priority exceeds the target number that you allocate, the system continues to add sockets until the interface is at its maximum capacity.

Note:
The 4606, 4612, and 4624 telephones do not support the load balancing feature of the TN2602AP circuit pack.

Valid entries	Usage
1 to platform maximum as follows: ● S87XX series: **3500** ● S8500: **3500** ● S8400: **2500** ● CHAWK/BOXTER: **2000** ● VM/BLADE: **1700**	Enter the maximum number of sockets targeted for this interface. The default is 80% of the platform maximum.

Target socket load and Warning level

This field appears when **Type** is **clan**. The value that you enter in the **Target socket load and Warning level** field controls the percentage of sockets allocated to each IP interface within the same Gatekeeper Priority. When all the IP interfaces within the same Gatekeeper Priority exceeds the target number that you allocate, the system continues to add sockets until the interface is at its maximum capacity. If the targeted percentage is exceeded on a CLAN, a warning alarm is generated.

If there is only one IP interface within a priority, the **Target socket load and Warning level** field is no longer used for load balancing. You can still enter a value in this field to receive an error or a warning alarm if the targeted value is exceeded.

Note:
The 4606, 4612, and 4624 telephones do not support the load balancing feature of the TN2602AP circuit pack.

Valid entries	Usage
1 to **499**	Enter the maximum number of sockets targeted for this interface. If the number of sockets exceeds the targeted number, a warning alarm is generated.The default is **400**.

Type

Identifies the type of IP interface.

Valid entries	Usage
C-LAN **VAL** **MEDPRO** **procr**	This field is auto-populated based on the slot location specified in the command line.

VLAN

This field sends VLAN instructions to the PROCR (S8300/S87XX Servers), C-LAN, and Media Processor boards. It does not send VLAN instructions to IP endpoints such as IP telephones and softphones. This field cannot be administered for VAL boards.

Valid entries	Usage
0 to **4095**	Specifies the virtual LAN value.
n	Disabled. This is the default.

Field descriptions for page 2

Figure 138: IP Interfaces - Type board location screen

```
change ip-interface 01a03                                       Page   2 of   x
                                  IP INTERFACES

                                 ETHERNET OPTIONS
                  Slot: 01A03
                  Auto? n
                 Speed: 10Mbps
                Duplex: Half
```

Figure 139: IP Interface - Type VAL screen

```
change ip-interface 01a09                                       Page   2 of   x
                                  IP INTERFACES

                                 ETHERNET OPTIONS
                  Slot: 01A09
                  Auto? n
                 Speed: 10Mbps
                Duplex: Half
```

Figure 140: IP Interfaces - Type C-LAN screen

```
change ip-interface 01a02                                       Page   2 of   x
                                  IP INTERFACES

                                 ETHERNET OPTIONS
                  Slot: 01A02
                  Auto? n
                 Speed: 10Mbps
                Duplex: Half

                                 IPV6 PARAMETERS
             Node Name:
           Subnet Mask: /64
     Gateway Node Name:
      Enable Interface? n


         Ethernet Link:
```

ETHERNET OPTIONS

With each new system or IP board installation, one standard procedure should be to apply matching speed/duplex settings to each IP board and its corresponding Ethernet switch port. Then these fields can be used to verify the configured settings.

Auto?

Valid entries	Usage
y/n	Enter **y** for auto-negotiation or **n** for manual speed and duplex settings. Default is **y**. You must set the **Auto?** field to **n** to enable the **Duplex** or the **Speed** fields.

Duplex

Valid entries	Usage
Full **Half**	Enter the duplex settings for this IP board. When **Speed** is set to 100Mbps, this field defaults to **Full**. You still have the option of changing the value to **Half**. Default is **Half**.

Speed

Valid entries	Usage
10Mbps **100Mbps**	Enter the speed of the Ethernet connection. When **Auto** is set to **n**, the only speed option available for the TN2602AP circuit pack is 100Mbps. This is the default and cannot be changed.

IPV6 PARAMETERS

Enable Interface?

Allows use of the Ethernet port.

Valid entries	Usage
y/n	Enter **y** to indicate that the Ethernet port associated with the TN2602AP circuit pack is in service. If this is an active board, set to **n** only when there is no standby, or when the standby has been disabled. **Note:** Enter **n** in this field before you make changes to the screen.

Ethernet Link

This display-only field shows the administered link number for an Ethernet link.

Valid entries	Usage
y/n	Shows the unique number for the Ethernet link assigned on the **Data Module** screen.

Gateway Node Name

Valid entries	Usage
Character string (up to 15 characters max.) or blank	Enter the gateway node name associated with the IP address of the LAN gateway associated with the TN2602AP. This entry also applies to the second TN2602AP circuit pack when **Critical Reliable Bearer** is **y**. **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.

Node Name

The unique node name for the IP interface administered on the **Node Names** screen.

Valid entries	Usage
Character string (up to 15 characters max.)	Enter the node name associated with the IP address of the TN2602AP circuit pack.

Subnet Mask

The subnet mask is a 64-bit binary number that divides the network ID and the host ID in an IP address.

Valid entries	Usage
characters	Enter the Subnet Mask for TN2602AP. This entry also applies to the second TN2602AP circuit pack when **Critical Reliable Bearer** is **y**. **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.

Field descriptions for page 3

Figure 141: IP Interfaces - Type board location screen

```
change ip-interface 01a03                                        Page   3 of   x
                                  IP INTERFACES

  VOIP/NETWORK THRESHOLDS
  Enable VoIP/Network Thresholds? y
                 Packet loss (%): 5
                     Jitter (ms): 50
                   RT Delay (ms): 500
```

VOIP/NETWORK THRESHOLDS

Enable VoIP/Network Thresholds?

Valid entries	Usage
y/**n**	Enable or disable the recording of Voice/Network Statistics at a system level for a single media processor board (applies to both TN2602 boards, if duplicated). If you change the value of this field, an updated message is sent to the media processor board. Default is **n**. If **Enable VoIP/Network Thresholds** field is set to **n**, **Packet loss**, **Jitter**, and **RT Delay** fields do not appear on the IP Server Interface screen.

Packet loss (%)

Valid entries	Usage
0 to **100**	The **Packet loss (%)** field displays if **Enable VoIP/Network Thresholds?** filed is set to **y** on the screen. Enter the percentage of the unacceptable packet loss coming into the administered media processor board. Default is **5**. **Note:** "xxx" indicates 100% packet loss. This field appears on the screen if the board type is media processor board.

Jitter (ms)

Valid entries	Usage
0 to **9999**	The **Jitter (ms)** field displays if **Enable VoIP/Network Thresholds?** filed is set to **y** on the screen. Enter the unacceptable jitter coming into the media processor board at which point data is captured to send up to Communication Manager. Default is **50** milliseconds. This field appears on the screen if the board type is media processor board.

RT Delay (ms)

Valid entries	Usage
1 to **9999**	The **RT Delay (ms)** (Round Trip Delay) field displays if **Enable VoIP /Network Thresholds?** field is set to **y** on the screen. Enter the unacceptable elapsed time for a packet to reach remote location and revert. Default is **500** milliseconds. This field appears on the screen if the board type is media processor board.

IP Network Region

Use this screen to configure within-region and between-region connectivity settings for all VoIP resources and endpoints within a given IP region. The first page is used to modify the audio and QoS settings. The **Codec Set** field on this page reflects the CODEC set that must be used for connections between telephones within this region or between telephones and MedPro/Prowler boards and media gateways within this region. The ability to do NAT shuffling for direct IP-to-IP audio connections is also supported. Use the **IP Address Mapping** screen to administer network regions.

Field descriptions for page 1

Figure 142: IP Network Region screen

```
change ip-network-region n                                        Page 1 of x
                              IP NETWORK REGION
  Region: n
Location:                  Authoritative Domain:
    Name:
                                Intra-region IP-IP Direct Audio: n
MEDIA PARAMETERS                Inter-region IP-IP Direct Audio: n
   Codec Set: 1                            IP Audio Hairpinning? n
UDP Port Min: 2048
UDP Port Max: 3028                       RTCP Reporting Enabled? y
                                  RTCP MONITOR SERVER PARAMETERS
DIFFSERV/TOS PARAMETERS           Use Default Server Parameters? y
 Call Control PHB Value:                      Server IP Address:   .   .   .
        Audio PHB Value:                            Server Port: 5005
        Video PHB Value:

802.1P/Q PARAMETERS                    RTCP Report Period(secs): 5
 Call Control 802.1p Priority: 7
        Audio 802.1p Priority: 6
        Video 802.1p Priority: 7
                                       AUDIO RESOURCE RESERVATION PARAMETERS
H.323 IP ENDPOINTS                                       RSVP Enabled? y
  H.323 Link Bounce Recovery? y               RSVP Refresh Rate(secs): 15
 Idle Traffic Interval (sec): 20      Retry upon RSVP Failure Enabled? y
   Keep-Alive Interval (sec): 6                          RSVP Profile:
            Keep-Alive Count: 5       RSVP unreserved (BBE) PHB Value: 40
```

Note:
The **`display ip-network-region`** command displays the values that you assign on this screen.

Authoritative Domain

The name or IP address of the domain for which this network region is responsible (that is, authoritative).

Valid entries	Usage
Up to 20 characters or blank.	Enter the name or IP address of the domain for which this network region is responsible. Note that this appears in the “From” header of any SIP Enablement Services (SES) messages.

Name

Description of the region.

Valid entries	Usage
Up to 20 characters	Describes the region. **NOTE:** Avaya BRI stations support ASCII characters only. They do not support non-ASCII characters, such as Eurofont or Kanafont. Therefore, if you use non-ASCII characters in any Communication Manager **Name** field, such characters do not display correctly on a BRI station.

Region

A display-only field indicating the number of the network region being administered. Network regions are defined on the **IP Address Mapping** screen.

MEDIA PARAMETERS

Codec Set

Specifies the codec set assigned to the region

Valid entries	Usage
1 to **7**	Enter the number for the codec set for the region.

Intra-region IP-IP Direct Audio

Allows direct audio connections between IP endpoints within a network region.

Valid entries	Usage
y/n	Enter **y** to save on bandwidth resources and improve sound quality of voice over IP transmissions. An **n** entry might be used if, for example, the IP telephones within the region are behind two or more firewalls.

Valid entries	Usage
native(NAT)	Enter **native(NAT)** if the IP address from which audio is to be received for direct IP-to-IP connections within the region is that of the telephone/ softphone itself (without being translated by NAT). IP telephones must be configured behind a NAT device *before* this entry is enabled.
translated(NAT)	Enter **translated(NAT)** if the IP address from which audio is to be received for direct IP-to-IP connections within the region is to be the one with which a NAT device replaces the native address. IP telephones must be configured behind a NAT device *before* this entry is enabled.

Inter-region IP-IP Direct Audio

Allows direct audio connections between IP endpoints in different regions.

Valid entries	Usage
y/n	Enter **y** to save on bandwidth resources and improve sound quality of voice over IP transmissions. An **n** entry might be used if, for example, the IP telephones within the region are behind two or more firewalls.
translated(NAT)	Enter **translated(NAT)** if the IP address from which audio is to be received for direct IP-to-IP connections between regions is to be the one with which a NAT device replaces the native address. IP telephones must be configured behind a NAT device *before* this entry is enabled.
native(NAT)	Enter **native(NAT)** if the IP address from which audio is to be received for direct IP-to-IP connections between regions is that of the telephone itself (without being translated by NAT). IP telephones must be configured behind a NAT device *before* this entry is enabled.

IP Audio Hairpinning

Allows IP endpoints to be connected through the IP circuit pack in the server.

Valid entries	Usage
y/n	Enter **y** to allow IP endpoints to be connected through the Avaya S8XXX Server's IP circuit pack in IP format, without first going through the Avaya DEFINITY TDM bus. Default is **n**.

Location

Specifies the location of the IP network region, allowing correct date and time information, and trunk routing based on IP network region.

Note:
If the Multinational Locations feature is enabled, and IP telephones derive their network region from the IP Network Map, you must administer this field with a valid value (1 to 250). This allows the IP endpoint to obtain a VoIP resource.

Valid entries	Usage
1 to **44**	(For CSI only.) Enter the number for the location for the IP network region. The IP endpoint uses this as its location number. This applies to IP telephones and softphones.
1 to **250**	(For Avaya S8300/S87XX Servers) Enter the number for the location for the IP network region. The IP endpoint uses this as its location number. This applies to IP telephones and softphones.
blank	The location is obtained from the cabinet containing the CLAN or the media gateway that the endpoint registered with.

RTCP Reporting Enabled

Valid entries	Usage
y/n	Enter **y** to send RTCP Reports to a special server, such as for the VMON tool. If this field is set to **y**, then the **RTCP Monitor Server Parameters** fields appear. **Note:** Regardless of how this field is administered, RTCP packets are always sent peer-to-peer.

UDP Port Range

UDP Port Range Min

Specifies the minimum range of the UDP port number used for audio transport.

Valid entries	Usage
1024 to **65534** defaults: **2048** to **3028**	Enter the lowest UDP port number to be used for audio transport.

UDP Port Range Max

Specifies the maximum range of the UDP port number used for audio transport.

Valid entries	Usage
1025 to **65535** defaults: **2048** to **3028**	Enter the highest UDP port number to be used for audio transport.

RTCP MONITOR SERVER PARAMETERS

RTCP Report Period (secs)

This field only appears when the **Use Default Server Parameters** field is set to **n** and the **RTCP Reporting Enabled** field is set to **y**.

Valid entries	Usage
5 to **30**	Enter the report period for the RTCP Monitor server in seconds.

Server IP Address

This field only appears when the **Use Default Server Parameters** field is set to **n** and the and the **RTCP Enabled** field is set to **y**.

Valid entries	Usage
0 to **255** in *nnn.nnn.nnn.nnn* format	Enter the IP address for the RTCP Monitor server.

Server Port

This field only appears when the **Use Default Server Parameters** field is set to **n** and the and the **RTCP Enabled** field is set to **y**.

Valid entries	Usage
1 to **65535**	Enter the port for the RTCP Monitor server. Default is **5005**.

Use Default Server Parameters

This field only appears when the **RTCP Reporting Enabled** field is set to **y**.

Valid entries	Usage
y	Enter **y** to use the default RTCP Monitor server parameters as defined on the **IP-Options System Parameters** screen. If set to **y**, you must complete the **Default Server IP Address** field on the IP Options System Parameters screen.
n	If you enter **n**, you need to complete the **Server IP Address**, **Server Port**, and **RTCP Report Period** fields that appear.

DIFFSERV/TOS PARAMETERS

Audio PHB Value

Provides scalable service discrimination in the Internet without per-flow state and signaling at every hop. Use the **IP TOS** field to support the Audio PHB codepoint.

Valid entries	Usage
0 to **63**	Enter the decimal equivalent of the DiffServ Audio PHB value. Default is **46**.

Call Control PHB Value

Provides scalable service discrimination in the Internet without per-flow state and signaling at every hop. Use the **IP TOS** field to support the DiffServ codepoint.

Valid entries	Usage
0 to **63**	Enter the decimal equivalent of the Call Control PHB value. Default is **34**.

Video PHB Value

Valid entries	Usage
0 to **63**	Enter the decimal equivalent of the DiffServ Video PHB value. Default is **26**.

802.1P/Q PARAMETERS

Audio 802.1p Priority

Provides Lay 2 priority for Layer 2 switches.

Valid entries	Usage
0 to **7**	Specifies the Audio 802.1p priority value. Changes take effect after circuit pack reset, telephone reboot, or system reset.

Call Control 802.1p Priority

Provides Layer 2 priority for Layer 2 switches.

Valid entries	Usage
0 to **7**	Specifies the 802.1p priority value. Changes take effect after circuit pack reset, telephone reboot, or system reset.

Video 802.1p Priority

Valid entries	Usage
0 to **7**	Specifies the Video 802.1p priority value. Changes take effect after circuit pack reset, telephone reboot, or system reset.

AUDIO RESOURCE RESERVATION PARAMETERS

Retry upon RSVP Failure Enabled

This field only appears if the **RSVP Enabled** field is set to **y**.

Valid entries	Usage
y/n	Specifies whether to enable retries when RSVP fails.

RSVP Enabled

The entry in this field controls the appearance of the other fields in this section.

Valid entries	Usage
y/n	Specifies whether or not you want to enable RSVP.

RSVP Profile

This field only appears if the **RSVP Enabled** field is set to **y**. You set this field to what you have configured on your network.

Valid entries	Usage
guaranteed-service	This limits end-to-end queuing delay from sender to receiver. This setting is best for VoIP applications.
controlled-load	This subset of **guaranteed-service** provides for a traffic specifier, but not end-to-end queuing delay.

RSVP Refresh Rate (secs)

This field only appears if the **RSVP Enabled** field is set to **y**.

Valid entries	Usage
1 to **99**	Enter the RSVP refresh rate in seconds.

RSVP unreserved (BBE) PHB Value

Valid entries	Usage
0 to **63**	The BBE codepoint is used whenever an RSVP reservation is being obtained (pending), or has failed in some way, to provide better-than-best service to the voice stream.

H.323 IP ENDPOINTS

H.323 Link Bounce Recovery

A **y** entry in this field enables the H.323 Link Bounce Recovery feature for this network region. An **n** disables the feature.

Valid entries	Usage
y/n	Specifies whether to enable H.323 Link Bounce Recovery feature for this network region. Default is **y**.

Idle Traffic Interval (seconds)

This field represents the maximum traffic idle time after which a TCP Keep-Alive (KA) signal is sent from the endpoint.

Valid entries	Usage
5 to **7200**	Enter the maximum traffic idle time in seconds. Default is **20**.

Keep-Alive Interval (seconds)

Use this field to set the interval between TCP Keep-Alive re-transmissions. When no ACK is received for all retry attempts, the local TCP stack ends the TCP session and the associated socket is closed.

Valid entries	Usage
1 to **120**	Specify the interval between KA retransmissions in seconds. Default is **5**.

Keep-Alive Count

Use this field to set the number of times the Keep-Alive message is transmitted if no ACK is received from the peer.

Valid entries	Usage
1 to **20**	Specify the number of retries when if no ACK is received. Default is **5**.

Field descriptions for Page 2

This page covers the information for Inter-Gateway Alternate Routing (IGAR), backup server names in priority order, and security procedures.

Figure 143: IP Network Region screen

```
 change ip-network-region n                                   Page 2 of x

                                IP NETWORK REGION

INTER-GATEWAY ALTERNATE ROUTING/DIAL PLAN TRANSPARENCY
Incoming LDN Extension:
Conversion to Full Public Number - Delete: ___Insert: ___
 Maximum Number of Trunks to Use for IGAR:
Dial Plan Transparency in Survivable Mode? n


BACKUP SERVERS IN PRIORITY ORDER            H.323 SECURITY PROCEDURES
1                                           1
2                                           2
3                                           3
4                                           4
5
6                                          Allow SIP URI Conversion? y

TCP SIGNALING LINK ESTABLISHMENT FOR AVAYA H.323 ENDPOINTS
   Near End Establishes TCP Signaling Socket? y
                       Near End TCP Port Min: 61440
                       Near End TCP Port Max: 61444
```

INTER-GATEWAY ALTERNATE ROUTING/DIAL PLAN TRANSPARENCY

If Inter-Gateway Alternate Routing (IGAR) is enabled for any row on pages 3 through 19, you must complete the following fields for each network region in order to route the bearer portion of an IGAR call. For more information on Inter-Gateway Alternate Routing, see *Administering Network Connectivity for Avaya Aura™ Communication Manager*, 555-233-504.

Conversion to Full Public Number - Delete

Valid entries	Usage
0 to **7**	Enter the digits to delete.

Conversion to Full Public Number - Insert

Valid entries	Usage
up to 13 digits	Enter up to 13 digits to insert, or blank. International numbers should begin with ’+’. **Note:** The optional “+” at the beginning of the inserted digits is an international convention indicating that the local international access code (for example, 011 in North America and 00 in Europe) must be dialed before the number.

Dial Plan Transparency in Survivable Mode

Valid entries	Usage
y/n	Enter **y** to enable the Dial Plan Transparency feature when a media gateway registers with a local survivable processor (LSP), or when a port network registers with an Enterprise Survivable Server (ESS). Default is **n**.

Incoming LDN Extension

Valid entries	Usage
Valid unused extension	Assign an unused Listed Directory Number for incoming IGAR calls.

Maximum Number of Trunks to Use for IGAR

It is necessary to impose a limit on the trunk usage in a particular pot network in a network region when IGAR is active. The limit is required because if there is a major IP WAN network failure, it is possible to use all trunks in the network region(s) for IGAR calls.

Valid entries	Usage
1 to **999**, or blank	Enter the maximum number of trunks to be used for Inter-gateway alternate routing (IGAR).

Note:
The S8500 supports up to 800 IP trunks (via license file limitations), which is less than the S87XX limit, but the overall maximum number of trunk members is the same as on the S87XX: 8000.

BACKUP SERVERS IN PRIORITY ORDER

This section lists the backup server names in priority order. Backup server names should include LSP server names, but should not include ESS server names. The six fields under this label allow any valid node name as an entry. Valid node names can include names of Customer LANs, ICCs, and LSPs.

H.323 SECURITY PROCEDURES

Use this field to select the permitted security profile(s) for endpoint registration in this network region. At least one security procedure entry must be present when this screen is submitted; otherwise, no endpoint is permitted to register from the region.

Valid entries	Usage
challenge	Includes the various methods of PIN-based challenge/response schemes in current use; relatively weak.
pin-eke	The H.235 Annex H SP1
strong	Permits use of any strong security profile; at present, only the **pin-eke** profile fits in this category.
all	Includes all of the above security profiles.
none	No security profile is required; permits use of an endpoint without user authentication (use with caution).

Allow SIP URI Conversion

Use this field to administer whether or not a SIP URI should be permitted to change. Degrading the URI from sips//: to sip//: may result in a less secure call. This is required when SIP SRTP endpoints are allowed to make and receive calls from endpoints that do not support SRTP.

Note:
If you enter **n** for no URI conversion, then calls from SIP endpoints that support SRTP made to other SIP endpoints that do *not* support SRTP fails.

Valid entries	Usage
y/n	Enter **y** to allow conversion of SIP URIs. Default is **y**.

TCP SIGNALING LINK ESTABLISHMENT FOR AVAYA H.323 ENDPOINTS

Near End Establishes TCP Signaling Socket

Use this field to indicate whether Communication Manager (the near end) can establish the TCP socket for H.323 IP endpoints in this network region.

Valid entries	Usage
y	When set to **y**, Communication Manager determines when to establish the TCP socket with the IP endpoints, assuming the endpoints support this capability. This is the default.
n	When set to **n**, the IP endpoints always attempt to set up the TCP socket immediately after registration. This field should be set to **n** only in network regions where a non-standard H.323 proxy device or a non-supported network address translation (NAT) device would prevent the server from establishing TCP sockets with H.323 IP endpoints.

Near End TCP Port Min

Use the **Near End TCP Port Min** and **Near End TCP Port Max** fields to specify a range of port numbers to be used by the Control Lan (C-LAN) circuit pack or processor Ethernet when establishing the TCP signaling socket to the H.323 IP endpoint. The range of port number must be at least 5 (Max-Min+1).

.

Valid entries	Usage
1024 to **65531**	Set the minimum port value to be used by the Control Lan (C-LAN) circuit pack or processor Ethernet when establishing the TCP signaling socket to the H.323 IP endpoint. Default is **61440**.

Near End TCP Port Max

Use the **Near End TCP Port Min** and **Near End TCP Port Max** fields to specify a range of port numbers to be used by the Control Lan (C-LAN) or processor Ethernet when establishing the TCP signaling socket to the H.323 IP endpoint. The range of port number must be at least 5 (Max-Min+1).

.

Valid entries	Usage
1028 to **65535**	Set the maximum port value to be used by the Control Lan (C-LAN) circuit pack or processor Ethernet when establishing the TCP signaling socket to the H.323 IP endpoint. Default is **61444**.

Field descriptions for Page 3

Each page from page 3 on shows the inter-region connectivity for 15 region pairs. To accommodate the maximum of 250 regions for Linux platforms, up to 17 pages are available for this purpose.

Figure 144: Inter Network Region Connection Management screen

```
change ip-network-region 1                                     Page   3 of  x

 Source Region: 1     Inter Network Region Connection Management     I        M
                                                                     G  A     e
 dst codec direct   WAN-BW-limits   Video      Intervening      Dyn  A  G     a
 rgn  set   WAN  Units    Total Norm  Prio Shr Regions          CAC  R  L     s
 1    1                                                               all
 2
 3
 4
 5
 6
 7
 8
 9
 10
 11
 12
 13
 14
 15
```

AGL

Use this field to administer the maximum number of destination region IP interfaces to be included in alternate gatekeeper lists (AGL).

Valid entries	Usage
0 to **72,** or blank	Default values are as follows: ● 72 for all direct WAN connected regions (including the same region itself) ● 0 for regions indirectly connected, ie., connected via intervening regions ● 0 or blank for regions that are not connected.

Audio WAN-BW limits (units)

The entry in this field the unit of measure corresponding to the value entered for bandwidth limitation.

Valid entries	Usage
Calls **Dynamic** **Kbits/sec** **Mbits/sec** **NoLimit**	This field allows you to limit bandwidth by number of connections, bandwidth in Kbits/sec, bandwidth in Mbits/sec, or it can be left blank. Default is blank.

codec-set

Indicates which codec set is to be used between the two regions.

Valid entries	Usage
1 to **7**, **pstn**	If the two regions are not connected at all, this field should be blank. When the codec set is blank, the **direct-WAN**, **WAN-BW-limits**, and **Intervening-regions** entry fields are not displayed. This field cannot be blank if this route through two regions is being used by some non-adjacent pair of regions.

direct-WAN

The entry in this field indicates whether the two regions (source and destination) are directly connected by a WAN link.

Valid entries	Usage
y/n	The default value is **y(es)** if the **codec-set** field is not blank. If so, the **WAN-BW-limits** field displays, but the **Intervening-regions** fields do not. If the **direct-WAN** field is set to **n(o)**, then the **WAN-BW-limits** field does not display, but the **Intervening-regions** fields are displayed.

dst rgn

The entry in this field identifies the destination region for this inter-network connection.

Valid entries	Usage
1 to **250**	Display-only. Shows the destination region for this inter-network connection.

Dynamic CAC Gateway

This field only appears if the **Audio WAN-BW- limit** field is set to **dynamic**. The gateway must be configured to be a CAC (Call Admission Control) gateway.

.

Valid entries	Usage
1 to **250**	Set the gateway that reports the bandwidth-limit for this link. Default is blank.

IGAR

This field allows pair-wise configuration of Inter-Gateway Alternate Routing between network regions. If the field is set to **y**, the IGAR capability is enabled between the specific network region pair. If it is set to **n**, the IGAR capability is disabled between the network region pair. The (**f**)orced option moves all traffic onto the PSTN.

For more information on Inter-Gateway Alternate Routing, see *Administering Network Connectivity for Avaya Aura™ Communication Manager*, 555-233-504.

Valid entries	Usage
y/	Enter **y** to enable IGAR capability between this network region pair.
n	IGAR capability between this network region pair is disabled. The default is **n**, except when **codec set** is **pstn**. When **codec set** is **pstn**, this field defaults to **y**.
f	Forced. This option can be used during initial installation to verify the alternative PSTN facility selected for a network region pair. This option can also be used to temporarily move traffic off of the IP WAN if an edge router is having problems or an edge router needs to be replaced between a network region pair.

Intervening-regions

The entry in this field allows entry of intervening region numbers between the two indirectly-connected regions.

Valid entries	Usage
1 to **250**	Enter up to four intervening region numbers between the two indirectly-connected regions. **Note**: Entry is not allowed for indirect region paths until all direct region paths have been entered. In addition, the order of the path through the regions must be specified starting from the source region to the destination region.

src rgn

The entry in this field identifies the source region for this inter-network connection.

Valid entries	Usage
1 to **250**	Display-only. Shows the source region for this inter-network connection.

Video (Norm)

Valid entries	Usage
0 to **9999** for Kbits, **0** to **65** for Mbits, or blank for **NoLimit**	Set the amount of bandwidth that you want to allocate for the normal video pool to each IP network region.

Video (Prio)

Valid entries	Usage
0 to **9999** for Kbits, **0** to **65** for Mbits, or blank for **NoLimit**	Set the amount of bandwidth that you want to allocate for the priority video pool to each IP network region.

Video (Shr)

Valid entries	Usage
y/n	Specify whether the normal video pool can be shared for each link between IP network regions.

WAN-BW limits (value)

This field is used for entry of the bandwidth limits for direct WAN links.

Valid entries	Usage
1 to **9999**	Values for this field can be entered in the number of connections, bandwidth in Kbits/sec, bandwidth in Mbits/sec, or left blank. Default is blank. **Note**: For Release 2.0, the number must be less than or equal to 65 when the units part of the field is set to Mbits/sec.

WAN-BW limits (units)

The entry in this field the unit of measure corresponding to the value entered for bandwidth limitation.

Valid entries	Usage
Calls **Kbits/sec** **Mbits/sec** **NoLimit**	This field allows you to limit bandwidth by number of connections, bandwidth in Kbits/sec, bandwidth in Mbits/sec, or **NoLimit**. Default is **NoLimit**.

IP Node Names

Use this screen to administer node names and IP addresses for the switch and the terminal server media processors administered on the **IP Interfaces** screen.

Note:
The Processor Ethernet interface node name (**procr**) automatically appears on the IP Node Names screen. The PE interface node name cannot be added to the IP Node Names screen. The line containing the keyword **procr** displays the IP address. For more information on Processor Ethernet, see *Administering Avaya Aura™ Communication Manager*, 03-300509.

Field descriptions for page 1

Figure 145: IP Node Names screen

```
change node-names ip                                              Page   1 of   2
                                  IP NODE NAMES
    Name              IP Address
Gateway001          135.27.153.254
S8300_10            135.27.162.236
S8300_9             135.27.153.208
SE12                135.27.162.242
SES-1               135.27.153.169
SIPp-1              135.27.153.135
SIPp-2              135.27.153.227
clan                135.27.153.165
default             0.0.0.0
medpro              135.27.153.166
procr               135.27.153.163
sipp69              135.27.162.69




( 12 of 12   administered node-names were displayed )
Use 'list node-names' command to see all the administered node-names
Use 'change node-names ip xxx' to change a node-name 'xxx' or add a node-name
```

Name

Identifies the name of the adjunct or server/switch node.

Valid entries	Usage
1 to 15 alpha-numeric characters	Used as a label for the associated IP address. The node names must be unique for each server/switch. **NOTE:** Avaya BRI stations support ASCII characters only. They do not support non-ASCII characters, such as Eurofont or Kanafont. Therefore, if you use non-ASCII characters in any Communication Manager **Name** field, such characters do not display correctly on a BRI station.

IP Address

The IP address for the node named in the previous field.

Note:
If you are using the Converged Communications Server for SIP Enablement Services (SES) Instant Messaging, enter the IP address for the SIP Enablement Services (SES) Proxy Server for your network.

Valid entries	Usage
32-bit address (4 decimal numbers, each in the range 0 to 255) also supports IPv6 addressing format	A unique IP address is assigned to each port on any IP device that is used for a connection. See *Administering Network Connectivity on Avaya Aura™ Communication Manager*, 555-233-504, for more information.

IP-Options System Parameters

Field descriptions for page 1

Figure 146: IP-Options System Parameters screen

```
change system-parameters ip-options                             Page   1 of   x
                          IP-OPTIONS SYSTEM PARAMETERS

 IP MEDIA PACKET PERFORMANCE THRESHOLDS
    Roundtrip Propagation Delay (ms)    High: 800      Low: 400
                    Packet Loss (%)     High: 40       Low: 15
                    Ping Test Interval (sec): 20
    Number of Pings Per Measurement Interval: 10
                  Enable Voice/Network Stats? n
 RTCP MONITOR SERVER
         Default Server IP Address:    .   .   .
               Default Server Port: 5005
  Default RTCP Report Period(secs): 5

AUTOMATIC TRACE ROUTE ON
           Link Failure? y


 H.248 MEDIA GATEWAY                   H.323 IP ENDPOINT
  Link Loss Delay Timer (min): 5        Link Loss Delay Timer (min): 5
                                          Primary Search Time (sec): 75
                                Periodic Registration Timer (min): 20
```

IP MEDIA PACKET PERFORMANCE THRESHOLDS

Enable Voice/Network Stats

Valid entries	Usage
y/n	Enable or disable the recording of voice/network statistics at a system level for all TN2302/TN2602 media processor boards in your network. The default value for **Enable Voice/Network Stats** field is **n**.

Number of Pings Per Measurement Interval

Specifies the number of test pings that comprise a measurement from which the performance values (delay and loss) are calculated.

Valid entries	Usage
10 to **100**	Enter the number. Default is **10**.

Packet Loss (%)

Specifies thresholds to be applied to packet loss rates (as measured by ping) for determining activation/deactivation of signaling group bypass.

High

Valid entries	Usage
0 to **100**	This value cannot be less than the minimum value. Default is **40**.

Low

Valid entries	Usage
0 to **100**	This value cannot be more than the maximum value. Default is **15**.

Ping Test Interval (sec)

Specifies the time between performance test pings for each testable signaling group.

Valid entries	Usage
10 to **999**	Enter the time. Default is **20**.

Roundtrip Propagation Delay (ms)

Specifies thresholds to be applied to roundtrip packet propagation delays (as measured by ping) for use in activating or clearing signaling group bypass.

High

Valid entries	Usage
10 to **9999**	This value cannot be less than the minimum value. Default is **800**.

Low

Valid entries	Usage
10 to **9999**	This value cannot be more than the maximum value. Default is **400**.

RTCP MONITOR SERVER

Default RTCP Report Period (secs)

In conjunction with the IP address and server port, this value tells the IP telephones, IP softphones and VoIP media modules how often to send the information (RTCP packets) to the RTCP server

.

Valid entries	Usage
5 to **99**	Enter the desired number of seconds.

Default Server IP Address

The default server IP address that can be utilized by the IP Network Region screen for each administered region.

Valid entries	Usage
0 to **255** in *nnn.nnn.nnn.nnn* format	A unique IP address is assigned to each port on any IP device that is used for a connection.

Default Server Port

The RTCP monitor is a separate computer that receives RTCP packets from many devices. Communication Manager pushes these values to IP telephones, IP softphones and VoIP media modules, such that they know where to send the data. The IP address is that of the RTCP server. The server port is the TCP/IP port of that RTCP server where the information should be sent.

Valid entries	Usage
1 to **65535**	Enter the port being used as the RTCP monitor. Default is **5005**. **Note:** You can also change the RTCP monitor server port setting from the default of **5005** for an individual network region by entering **n** in the **Use Default Server Parameters** field in the RTCP Monitor Server section of the **IP Network Region** screen. When you enter **n**, additional fields appear for entering alternative server parameters.

Automatic Trace Route on Link Failure

In order to diagnose network problems, especially to determine where a network outage exists, Communication Manager initiates an automatic trace-route command when the connectivity between a server and its port networks, media gateways, or IP trunks is lost.

Valid entries	Usage
y	Enter **y** to turn the automatic trace route command feature on.
n	Enter **n** to turn the automatic trace route command feature off.

Note:
If you disable the feature, any automatic trace-route currently in progress finishes, and no subsequent trace-route commands are launched or logged (the link failure buffer is cleared).

MEDIA GATEWAY ANNOUNCEMENT SERVER PARAMETERS

Announcement Server IP Address

Identifies the IP address of the Announcement Server.

Valid entries	Usage
0 to **255**	A unique IP address is assigned to each port on any IP device that is used for a connection.

Announcement Storage Path Name

Indicates the path name on the Announcement Server where the announcements are stored.

Valid entries	Usage
Up to 40 characters or blank	Enter the directory path name where announcements are stored.

Login

Indicates the login to be used by the Media Gateway to access the Announcement Server.

Valid entries	Usage
1 to 10 characters or blank	Enter a login up to 10 characters.

Password

Indicates the password to be used by the Media Gateway to access the Announcement Server.

Valid entries	Usage
1 to 10 characters or blank	Enter a password up to 10 characters.

H.248 MEDIA GATEWAY

Link Loss Delay Timeout (minutes)

This field is to assist with the H.248 link bounce recovery mechanism of the Avaya G700 Media Gateway; specifically, to prevent the call controller from removing all boards and ports prematurely in response to a link bounce.

Valid entries	Usage
1 through **30**	Enter the number of minutes to delay the reaction of the call controller to a link bounce. Default is **5**

H.323 IP ENDPOINT

Link Loss Delay Timer (minutes)

This timer specifies how long the Communication Manager server preserves registration and any stable calls that might exist on the endpoint after it has lost the call signaling channel to the endpoint. If the endpoint does not re-establish connection within this period, Communication Manager tears down the registration and calls (if any) of the endpoint. This timer does not apply to soft IP endpoints operating in telecommuter mode.

.

Valid entries	Usage
1 to **60**	Enter the number of minutes to delay the reaction of the call controller to a link bounce. Default is **5**.

Periodic Registration Timer (min)

This timer is started when an IP telephone registration is taken over by another IP endpoint. When the timer expires, the telephone tries to reregister with the server. Default timer value is dependent on the number of unsuccessful periodic registration attempts. As long as the RRJ error message continues to be “Extension in Use,” the endpoint continues to attempt registration with the current gatekeeper address. Sample field values apply unless the endpoint is interrupted, such as by power loss, or the user takes manual action to override this automatic process:

- 20 means once every 20 minutes for two hours, then once an hour for 24 hours, then once every 24 hours continually.

- 60 means once an hour for two hours, then once an hour for 24 hours, then once every 24 hours continually.

Valid entries	Usage
1 to **60**	Enter the number of minutes before an IP telephone registration is taken over by another IP endpoint attempts to reregister with the server. Default is **60**.

Primary Search Time (seconds)

While the telephone is hung-up, this is the maximum time period that the IP endpoint expends attempting to register with its current Communication Manager server. The need for this timer arises in situations where the current Communication Manager server might have a large number of Control Lan (C-LAN) circuit packs. this timer allows the customer to specify the maximum time that an IP endpoint spends on trying to connect to the Control Lan (C-LAN) circuit packs before going to an LSP.

While the IP telephone's receiver is lifted, the endpoint continues trying to re-establish connection with the current server until the call ends.

Valid entries	Usage
5 to **3600**	Enter the number of seconds an IP endpoint spends on trying to connect to the C-LAN circuit packs before going to an LSP. Default is **75**.

Field descriptions for page 2

Figure 147: IP-Options System Parameters screen

```
change system-parameters ip-options                             Page   2 of   4
                          IP-OPTIONS SYSTEM PARAMETERS

 Always use G.711 (30ms, no SS) for intra-switch Music-On-Hold? n
                      Force Phones and Gateways to Active LSPs? n

 IP DTMF TRANSMISSION MODE
   Intra-System IP DTMF Transmission Mode: rtp-payload
                     Inter-System IP DTMF: See Signaling Group Forms

 HYPERACTIVE MEDIA GATEWAY REGISTRATIONS
   Enable Detection and Alarms? n
```

Always use G.711 (30ms, no SS) for intra-switch Music-On-Hold

Valid entries	Usage
y/n	A **y** entry indicates that G.711 is used for intra-switch Music-On-Hold. Default is **n**.

Force Phones and Gateways to Active LSPs?

This field indicates whether telephones and media gateways are forced to active LSPs.

Valid entries	Usage
y/n	The default value is **n**. Set this field to **y**, to force all the telephones and media gateways backed-up by an LSP to register to the LSP if it becomes active.

IP DTMF TRANSMISSION MODE

Intra-System IP DTMF Transmission Mode

Enter the appropriate IP transmission mode.

Valid entries	Usage
in-band	DTMF digits encoded within existing RTP media stream for G.711/G.729 calls. G.723 is sent out-of-band.
rtp-payload	Initially, support for SIP Enablement Services (SES) trunks requires the entry of **rtp-payload**.

Inter-System IP DTMF Transmission Mode

See the **DTMF Over IP** field on the Signaling Group screen.

HYPERACTIVE MEDIA GATEWAY REGISTRATIONS

Enable Detection and Alarms

Enables or disables the hyperactive media gateway registration feature. Default is **n**.

Valid entries	Usage
y/n	Enter **y** to enable the hyperactive media gateway registration feature.

Parameters for Media Gateway Alarms: Hyperactive Registration Window (minutes)

This field appears when, in the DETECTION AND ALARMING OF HYPERACTIVE MEDIA GATEWAY REGISTRATIONS section of the IP-Options System Parameters screen, **Feature Enabled** is **y**.

Valid entries	Usage
1 to **15**	Time in minutes for checking hyperactive media gateway registrations. Default is 4 minutes.

Number of Registrations within the Window

This field appears when, in the DETECTION AND ALARMING OF HYPERACTIVE MEDIA GATEWAY REGISTRATIONS section of the IP-Options System Parameters screen, **Feature Enabled** is **y**.

Valid entries	Usage
1 to **19**	Number of registrations that occur within the hyperactivity window for generating a Gateway alarm. Default is 3.

Parameters for Network Region Registration (NR-REG) Alarms: % of Gateways in Network Region with Hyperactive Registration Alarms

This field appears when, in the DETECTION AND ALARMING OF HYPERACTIVE MEDIA GATEWAY REGISTRATIONS section of the IP-Options System Parameters screen, **Feature Enabled** is **y**.

Valid entries	Usage
1 to **99**	Percent of Gateways within an ip-network region that should be alarmed before an IP-Registration alarm is generated. Default is 80%.

Field descriptions for page 3

Use this screen to administer SNMP station parameters and services dialpad parameters. Applicable terminal types include: 4601, 4602, 4610, 4620, 4621, 4622, 4625, 96xx, or 16xx.

Figure 148: IP-Options System Parameters screen

```
change system-parameters ip-options                             Page   3 of   4
                          IP-OPTIONS SYSTEM PARAMETERS

 SNMP PARAMETERS
    Download Flag? n
    Community String:

 SOURCE ADDRESSES
        1.                                  4.
        2.                                  5.
        3.                                  6.


 SERVICES DIAL PAD PARAMETERS

        Download Flag? n
        Password: 27238
```

SNMP STATION PARAMETERS

Community String

The SNMP Community String is used by IP endpoints to determine whether the terminal allows receipt of SNMP queries, and if so, with what "password." If the SNMP community string is null, the terminal ignores all incoming SNMP messages. Otherwise, the community string must be present in the incoming SNMP message for the Terminal to act on that message (subject to other considerations, such as the SNMP Source Address).

Valid entries	Usage
1 to 32 ASCII characters, or blank	Default is NULL (string of zero length). If Community String is null, the terminal ignores all incoming 20 SNMP messages.

Download Flag

Valid entries	Usage
y/n	Determines whether the SNMP parameters are downloaded to the terminals or not. If set to **n**, the Community String and associated IP Addresses are NOT downloaded to terminals. If set to **y**, Community String and associated IP Addresses are downloaded to terminals.Default is **n**. To disable SNMP parameters, set the **Download** Flag to **y** and leave the **Community String** field blank.

Source Addresses

The SNMP Source IP Address(es) are used to validate the source of an SNMP message. If the SNMP Source Address list is null, the Terminal responds to any valid SNMP message (where “valid” means the appropriate SNMP community string is properly included). Otherwise, the Terminal responds to valid SNMP messages only if the IP Source Address of the query matches an address in the SNMP Source Address list.

Valid entries	Usage
Valid Node Name	Enter up to 6 Node Names. Node Names map to proper IP Addresses on the **IP Node Names** screen. An IP address of 0.0.0.0 is a valid address. If you want to remove a node name from the list, you must make sure that the node name is not being used any place in the administration system.

SERVICES DIALPAD PARAMETERS

Download Flag

Valid entries	Usage
y/n	Determines whether the administered Password is downloaded to the terminals or not. If set to **n**, the administered Password is NOT downloaded to terminals. If set to **y**, the administered Password is downloaded to terminals. Default is **n**. To disable SERVICES DIAL PAD parameters, set the **Download Flag** to **y** and leave the **Password** field blank.

Password

Valid entries	Usage
up to 7 digits (1 to 9), or blank	Enter a password. The Craft Procedures Password is used as part of the Craft Procedures (also called "Local Procedures") that allow a technician to go to an IP Terminal, and modify individual parameters on that specific Terminal (such as the Terminal's IP address, Ethernet interface speed, etc.). The Craft Procedures Password must be entered on the dialpad in the applicable manner, for the technician to have access to the Craft Procedures. Default is 27238 (craft).

Field descriptions for page 4

Figure 149: Syslog From TN Boards screen

```
change system-parameters ip-options                          Page   4 of   x
                                SYSLOG FROM TN BOARDS


   Local Facility #: local4

  Dest #1 IP address:                                        Port #: 514
  Dest #2 IP address:                                        Port #: 514
  Dest #3 IP address:                                        Port #: 514
```

Dest # 1, 2 or 3 IP address

Valid entries	Usage
32-bit address (4 decimal numbers, each in the range 0 to 255)	Enter the valid destination IPv4 address format. Provides support for future IPv6 address format. The default destination address is **0.0.0.0**

Local Facility

Valid entries	Usage
local0 to **local7** (for local use)	Displays the help message upon acceptable values. The default value is **local4**.

Port

Valid entries	Usage
1 to **65535**	Enter the valid port number associated with the **Dest # 1, 2 or 3 IP address** field. The default port number is **514**.

IP Routing

Figure 150: IP Routing screen

```
add ip-route next                                                  Page   1 of   x
                              IP ROUTING


     Route Number: 1
 Destination Node:
     Network Bits: /
          Gateway:
            Board:
           Metric:
```

Field descriptions for page 1

Board

Enter the necessary characters.

Valid entries	Usage
01 to **03** (DEFINITY CSI) or **1** to **64** (S87XX/S8300 Servers)	First and second characters are the cabinet number
A to **E**	Third character is the carrier
0 to **20**	Fourth and fifth character are the slot number
01 to **04** (Analog TIE trunks) **01** to **31**	Six and seventh characters are the circuit number
1 to **80** (DEFINITY CSI) or **1** to **250** (S87XX/S8300 Servers)	Gateway
V1 to **V9**	Module

Destination Node

The node name of the final destination for this connection.

Valid entries	Usage
The name previously entered on the **IP Node Names** screen.	Enter the name of the final destination node of the IP route for this connection.

Gateway

The node name of the first intermediate node.

Valid entries	Usage
A name previously entered on the **IP Node Names** screen and is either a port on the CLAN circuit pack or is identified as a Destination Node on another IP route.	If there are one or more intermediate nodes, the first intermediate node is the Gateway. If there are no intermediate nodes between the local and remote CLAN ports for this connection, the Gateway is the local CLAN port.

Metric

Valid entries	Usage
0 or **1**	Enter **1** on a server that has more than one CLAN circuit pack installed.

Network Bits

This field is a 32-bit binary number that divides the network ID and the host ID in an IP address.

Valid entries	Usage
0 to **32**	Enter the number of Network Bits associated with this IP route.

Route Number

Identifies the IP route.

Valid entries	Usage
1 to **400**	Enter the number of the IP route you want to add or change, or enter **n** for the next available number.

IP Server Interface (IPSI) Administration

Use this screen to add a TN2312 IPSI (IP Server Interface) circuit pack.The Avaya S8XXX Server uses the IP Server Interface (IPSI) to control port networks and provide tone, clock, and call classification services. The IPSI board connects to the control network by way of Ethernet.

In Communication Manager Release 5.2, the IP server interface administration for the TN2312 IPSI or the TN8412 SIPI provides support for Communication Manager based SAT administration of IPSI Quality of Service (QoS) and Ethernet interface settings parameters. All further references to IPSI in this document also apply to the TN8412 SIPI.

Note:
Initial IPSI settings must be done using the IPSI CLI interface.

For more information, see *Administering Network Connectivity for Avaya Aura™ Communication Manager*, 555-233-504.

Field descriptions for page 1

Figure 151: IP Server Interface (IPSI) Administration screen

```
change ipserver-interface n                                 Page   1 of   2
          IP SERVER INTERFACE (IPSI) ADMINISTRATION - PORT NETWORK 1

   IP Control? y             Ignore Connectivity in Server Arbitration? n
   Encryption? y

 PRIMARY IPSI                                    QoS AND ETHERNET SETTINGS
         DHCP? n                     Use System Level Parameter Values? y
                                                              802.1p: 6
     Location:  1A01                                        DiffServ: 46
  Subnet Mask: /24                                              Auto? n
    IP Address: 135.9.181.25                                    Speed: 100Mbps
     Gateway:                                                  Duplex: Full
```

Administer secondary ip server interface board

Valid entries	Usage
y/n	Enter **y** to assign a secondary IPSI board.

Ignore Connectivity in Server Arbitration

Valid entries	Usage
y/n	Default is **n**.

IP Control

Use this field to administer IP control of port networks.

Note:
In Phase 1 of the S8400, this field is display-only and is set to **y**. This is because, in phase 1, the S8400 is a single port network. The IPSI functionality must therefore be turned on to support the port network. In phase 2 of the S8400, when duplication is supported, this restriction is removed.

Valid entries	Usage
y	All port networks have an IPSI that provides control. ● display-only, if **IP-PNC** is **y** on the System Parameters Customer-Options **(Optional Features)** screen ● A DS1 Converter (DS1C) circuit pack cannot be added to a port network when **IP Control** is **y**
n	This IPSI is used only for Tone Clock/Tone Detector functions ● remaining fields on this screen do not appear when **IP Control** is **n** and **IP-PNC** is **n** on the System Parameters Customer-Options **(Optional Features)** screen ● **n** when the port network contains a DS1 Converter (DS1C) circuit pack

Encryption

Valid entries	Usage
y/n	Enter **y** to turn on socket encryption for the Avaya S8XXX Server and IPSI link.

PRIMARY IPSI

DHCP?

Valid entries	Usage
y/**n**	Displays whether IPSI is currently set up for DHCP addressing, or static addressing. If DHCP is not enabled in the System Management Interface, the value of **DHCP?** field is set to **n** (read-only). If you attempt to change the **DHCP** value from **y** to **n** on the IP Server Interface screen, the following event occurs: ● If IPSI is in-service, it disallows the service and displays the `ipserver must be busied out` message. ● If IPSI is busied out, the static equivalent to the DHCP address automatically populates the **Host** field. You must populate the **Subnet Mask** and **Gateway** fields manually. You can optionally overwrite the pre-populated values. If you attempt to change the **DHCP** value from **n** to **y** on the IP Server Interface screen, the following event occurs: ● If the IPSI is in-service, it disallows and displays the `ipserver must be busied out` message. ● If the IPSI is busied out, it accepts the changes and re-populates the DHCP field.

Gateway

Valid entries	Usage
32-bit address (4 decimal numbers, each in the range 0 to 255)	Enter the valid gateway IPv4 address. Provides support for future IPv6 address format. If the **DHCP** field is **y** on the IP Server Interface screen, the **Gateway** field is read-only. You can view the **Gateway** field based on the access. If the **DHCP** field is **n** on the IP Server Interface screen and if you attempt to change the gateway address, after screen validation, the system checks if IPSI is busied out. If not, the IPSI does not accept the change, and displays the `ipserver must be busiedout` message.

Host

Valid entries	Usage
characters/digits	Enter the name of the DHCP client identifier. If DHCP is enabled on the System Management Interface, the **Host** field is displayed on the IP Server Interface screen and the field is read-only. If DHCP is not enabled, **Host** field is not displayed.

IP Address

Valid entries	Usage
32-bit address (4 decimal numbers, each in the range 0 to 255)	Enter the valid IPv4 IP address. Provides support for future IPv6 address format. If DHCP is **y** on the IP Server Interface screen, the **IP Address** field is read-only. If DHCP is **n** and if you attempt to change the IP address, after screen validation, the system checks if IPSI is busied out. If not, the IPSI does not accept change, and displays the `ipserver must be busiedout` message.

Location

Valid entries	Usage
cabinet (**1** to **64**); carrier (**A** to **E**), slot(**1** to **20**);OR gateway(**1** to **250**), module(**V1** to **V9**)	Enter the IPSI board location.

Subnet Mask

Valid entries	Usage
/xx	The field value is represented as subnet bits. If DHCP is **y** on the IP Server Interface screen, the Subnet Mask field is read-only. You can view the **Subnet Mask** field based on the access. If DHCP is **n** and if you attempt to change the mask, after screen validation, the system checks if IPSI is busied out. If not, the IPSI does not accept change, and displays the `ipserver must be busiedout` message.

QoS AND ETHERNET SETTINGS

802.1p

Valid entries	Usage
0 to **7** (whole numbers)	If **Use System Level Parameter Values?** field is set to **y** on the change IP Server Interface screen, this field is read-only. If **Use System Level Parameter Values?** field is set to **n** on the change IP Server Interface screen, you can change the value of the **802.1p** field. This value takes effect when the IPSI is busied out or released. The default value is **6**.

Auto?

Valid entries	Usage
y/**n**	If **Auto?** field is set to **y**, **Speed** and **Duplex** fields do not appear on the IP Server Interface screen. If **Auto?** field is set to **n**, **Speed** and **Duplex** fields appear on the IP Server Interface screen. The default value is **y**. If the IPSI is not busied out, the following happens: ● This field is read-only. ● If you attempt to change the value, IPSI displays the `"y" Must busyout IPSI before changing this field` message. If the IPSI is busied out, you can change the value of the field.

DiffServ

Valid entries	Usage
0 to **63** (whole numbers)	If **Use System Level Parameter Values?** field is set to **y** on the change IP Server Interface screen, this field is read-only. If **Use System Level Parameter Values?** field is set to **n** on the change IP Server Interface screen, you can change the value of the **DiffServ** field. This value takes effect when the IPSI is busied out or released. The default value is **46**.

Duplex

Valid entries	Usage
Half/**Full**	Enter the duplex settings for this IP board. If the IPSI is not busied out, the following happens: ● This field is read-only. ● If you attempt to change the value, displays the `ipserver must be busied out` message. If the IPSI is busied out, you can change the value. The default value is **Full**.

Speed

Valid entries	Usage
10Mbps / **100Mbps**	Enter the speed of the Ethernet connection. If the IPSI is not busied out, the following happens: ● This field is read-only. ● If you attempt to change the value, display the `ipserver must be busied out` message. If the IPSI is busied out, you can change the value. The default value is **100 Mbps**.

Use System Level Parameter Values?

Valid entries	Usage
y/**n**	If **Use System Level Parameter Values?** field is set to **y**, the following happens: ● Both the **802.1p** and the **DiffServ** fields is read-only, as set on the System Parameters IP Server Interface screen. ● If you attempt to change the value, displays the `value set in system-parameters ipserver-interface` message. If **Use System Level Parameter Values?** field is set to **n**, you can set the values of **802.1p** and **DiffServ** fields. The default value is **y**.

SECONDARY IPSI

DHCP?

Valid entries	Usage
y/**n**	Displays whether IPSI is currently set up for DHCP addressing, or static addressing. If DHCP is not enabled in the ecs.conf file (through the IPSI Web page), the value of **DHCP?** field is set to **n** (read-only). If you attempt to change the **DHCP** value from **y** to **n** on the IP Server Interface screen, the following event occurs: ● If IPSI is in-service, it disallows the service and displays the `ipserver must be busied out` message. ● If IPSI is busied out, the static equivalent to the DHCP address automatically populates the **Host** field. You must populate the **Subnet Mask** and **Gateway** fields manually. You can optionally overwrite the pre-populated values. If you attempt to change the **DHCP** value from **n** to **y** on the IP Server Interface screen, the following event occurs: ● If the IPSI is in-service, it disallows and displays the `ipserver must be busied out` message. ● If the IPSI is busied out, it accepts the changes and re-populates the DHCP field.

Host

Valid entries	Usage
characters/digits	Enter the name of the DHCP client identifier. If DHCP is enabled on the System Management Interface, the **Host** field is displayed on the IP Server Interface screen and the field is read-only. If DHCP is not enabled, **Host** field is not displayed.

Gateway

Valid entries	Usage
32-bit address (4 decimal numbers, each in the range 0 to 255)	Enter the valid gateway IPv4 address. Provides support for future IPv6 address format. If **DHCP** field is set to **y** on the IP Server Interface screen, the **Gateway** field is read-only. You can view the **Gateway** field based on the access. If **DHCP** field is set to **n** on the IP Server Interface screen and if you attempt to change the gateway address, after screen validation, the system checks if IPSI is busied out. If not, the IPSI does not accept the change, and displays the `ipserver must be busiedout` message.

IP Address

Valid entries	Usage
32-bit address (4 decimal numbers, each in the range 0 to 255)	Enter the valid IPv4 IP address. Provides support for future IPv6 address format. If **DHCP** field is set to **y** on the IP Server Interface screen, the **IP Address** field is read-only. If **DHCP** field is set to **n** on the IP Server Interface screen and if you attempt to change the IP address, after screen validation, the system checks if IPSI is busied out. If not, the IPSI does not accept the change, and displays the `ipserver must be busiedout` message.

Location

Valid entries	Usage
cabinet (**1** to **64**); carrier (**A** to **E**), slot (**1** to **20**):OR gateway (**1** to **250**), module (**V1** to **V9**)	Enter the board location for the secondary IPSI.

Subnet Mask

Valid entries	Usage
/xx	The field value is represented as subnet bits. If **DHCP** field is set to **y** on the IP Server Interface screen, the Subnet Mask field is read-only. You can view the **Subnet Mask** field based on the access. If **DHCP** field is set to **n** and if you attempt to change the mask, after screen validation, the system checks if IPSI is busied out. If not, the IPSI does not accept change, and displays the `ipserver must be busiedout` message.

Field descriptions for page 2

Figure 152: IP Server Interface (IPSI) - Syslog Settings screen

```
change ipserver-interface 1                                    Page   2 of   x
                               SYSLOG SETTINGS

      Enable Syslog: y
   Local Facility #: local4                  Use System Syslog Values: n

 Dest #1 IP address:                                           Port #: 514
 Dest #2 IP address:                                           Port #: 514
 Dest #3 IP address:                                           Port #: 514

 DEBUG FILTER VALUES
          Object          Level
 1)                       65535
 2)                       65535
 3)                       65535
 4)                       65535
```

Dest # 1, 2 or 3 IP address

Valid entries	Usage
32-bit address (4 decimal numbers, each in the range 0 to 255)	Enter the valid destination IPv4 address format. Provides support for future IPv6 address format. The default destination address is unspecified.

Enable Syslog

Valid entries	Usage

y/**n**	This field is enabled only for init and inads login access.
	If you attempt to set to **y**, the following event occurs:
	Checks the firmware support syslog by capabilities exchange information as described:
	● If it does not get the support, displays the `unsupported board code or vintage` error message and does not enable the syslog value to **y**.
	● Else the following event occurs:
	  ● Displays the rest of the syslog fields.
	  ● When IP Interfaces screen is submitted, checks if at least one severity level and at least one facility level is selected. Else revert back the syslog value to **n**.
	If you set to **n**, does not display the rest of the syslog option fields.
	For duplicated TN2602 boards, this field is displayed for each board.
	The default value is **n**.

Local Facility

Valid entries	Usage
local0 to **local7** (for local use)	Displays the help message on acceptable values. The default value is **local4**.

Port

Valid entries	Usage
1 to **65535**	Enter the valid port number associated with the **Dest # 1, 2 or 3 IP address** field. The default port number is **514**.

Use System Syslog Values

Valid entries	Usage
y/**n**	If set to **y**, populates the address, port and facility information as read-only. The Default value is **n**.

DEBUG FILTER VALUES

Object

Valid entries	Usage
0 to **255**	Enter the category of a log event. For example, angel and archangel. **Note:** Maximum value of 255 trigger a file dump.

Level

Valid entries	Usage
0 to **65535**	Enter the level of logging for the given object. **Note:** Maximum value of 65535 trigger a file dump.

IP Services

The IP Services screen is used to administer the connectivity for various adjuncts.

Note:
You cannot remove a service from this screen if that service has overrides defined on the **Survivable Processor** screen. For more information, see *Administering Avaya Aura™ Communication Manager*, 03-300509.

Field descriptions for page 1

Figure 153: IP Services screen

```
change ip-services                                              Page 1 of X

                               IP SERVICES
Service      Enabled     Local      Local         Remote        Remote
  Type                   Node       Port          Node          Port
___________     _      __________  ____   ___________________ _____
___________     _      __________  ____   ___________________ _____
___________     _      __________  ____   ___________________ _____
___________     _      __________  ____   ___________________ _____
___________     _      __________  ____   ___________________ _____
___________     _      __________  ____   ___________________ _____
___________     _      __________  ____   ___________________ _____
___________     _      __________  ____   ___________________ _____
___________     _      __________  ____   ___________________ _____
___________     _      __________  ____   ___________________ _____
___________     _      __________  ____   ___________________ _____
___________     _      __________  ____   ___________________ _____
___________     _      __________  ____   ___________________ _____
___________     _      __________  ____   ___________________ _____
___________     _      __________  ____   ___________________ _____
___________     _      __________  ____   ___________________ _____
___________     _      __________  ____   ___________________ _____
```

Enabled

This field appears when **Service Type** is **AESVCS** or **SAT**. Controls whether the IP Service specified under **Service Type** is enabled.

Valid entries	Usage
y	Enter **y** to enable this IP service.
n	This IP service is disabled.

Local Node

Specify the node name for the port.

Valid entries	Usage
Node names as defined on the IP Node Names screen.	If the link is administered for services over the Control Lan (C-LAN) circuit pack, enter a node name defined on the **IP Node Names** screen.
procr	Enter **procr** to use the Communication Manager's Processor Ethernet interface for adjunct connectivity.

Local Port

Specify the originating port number.

Valid entries	Usage
5000 to **9999**	**5111** to **5117** for SAT applications **5678** for ASAI
0	For client applications, defaults to **0**.

Remote Node

Specify the server/switch at the far end of the link for SAT. The remote node should not be defined as a link on the IP Interface or Data Module screens.

Valid entries	Usage
Node name as defined on the **IP Node Names** screen	For SAT, use a node name to provide added security.
any	Use any available node.

Remote Port

Specify the port number of the destination.

Valid entries	Usage
5000 to **64500**	Use if this service is a client application, such as CDR or PMS. This must match the port administered on the adjunct, PC or terminal server that is at the remote end of this connection.
0	Default for System Management applications.

Service Type

Defines the service provided.

Valid entries	Usage
AESVCS	AE Services.
CBC	Reserves the trunk for outgoing use only to enhance Network Call Redirection.
CDR1, CDR2	Use this to connect either the primary or secondary CDR device over a TCP/IP link.
PMS	Property Management System.
PMS_JOURNAL	Use this to connect the PMS journal printer over a TCP/IP link.
PMS_LOG	Use this to connect the PMS log printer over a TCP/IP link.
	1 of 2

Valid entries	Usage
SAT	System administration terminal.
SYS_PRINT	Use this to connect the system printer over a TCP/IP link.
	2 of 2

IP Services screen - Session Layer Timers page

Use this screen to enable reliable protocol for TCP/IP links, and to establish other session-layer parameters. This screen only appears if you enter CDR1, CDR2, PMS_JOURNAL, or PMS_LOG in the **Service Type** field on page 1 or 2.

Figure 154: IP Services screen - Session Layer Timer page

```
change ip-services                                          Page   3 of   x

                               SESSION LAYER TIMERS
Service    Reliable   Packet Resp    Session Connect    SPDU    Connectivity
 Type      Protocol      Timer        Message Cntr      Cntr       Timer
CDR1          y          3                  1             1          1
```

Connectivity Timer

Valid entries	Usage
1 to **255**	Indicates the amount of time (in seconds) that the link can be idle before Communication Manager sends a connectivity message to ensure the link is still up.

Packet Resp Timer

Valid entries	Usage
1 to **255**	Determines the number of seconds to wait from the time a packet is sent until a response (acknowledgement) is received from the far-end, before trying to resend the packet.

Reliable Protocol

Indicates whether you want to use reliable protocol over this link.

Valid entries	Usage
y/n	Use reliable protocol if the adjunct on the far end of the link supports it.

Service Type

A display-only field that identifies the service type for which you are establishing parameters.

Valid entries	Usage
CDR1, CDR2	Used to connect either the primary or secondary CDR device over a TCP/IP link.
PMS_JOURNAL	Used to connect the PMS journal printer over a TCP/IP link.
PMS_LOG	Used to connect the PMS log printer over a TCP/IP link.

Session Connect Message Cntr

Valid entries	Usage
1 to **5**	The Session Connect Message counter indicates the number of times Communication Manager tries to establish a connection with the far-end adjunct.

SPDU Cntr

Valid entries	Usage
1 to **5**	The Session Protocol Data Unit counter indicates the number of times Communication Manager transmits a unit of protocol data before generating an error.

IP Services screen - AE Services Administration page

This screen appears when the **Service Type** is **AESVCS**. Use this screen to create symbolic name and password pairs for all AE Services servers that are allowed to connect to Communication Manager.

Figure 155: IP Services screen - AE Services Administration page

```
change ip-services                                            Page   4 of   x

                          AE Services Administration

Server ID        AE Services       Password      Enabled      Status
                   Server

___________      ____________   ______________    _____
___________      ____________   ______________    _____
___________      ____________   ______________    _____
___________      ____________   ______________    _____
___________      ____________   ______________    _____
___________      ____________   ______________    _____
___________      ____________   ______________    _____
___________      ____________   ______________    _____
___________      ____________   ______________    _____
___________      ____________   ______________    _____
___________      ____________   ______________    _____
___________      ____________   ______________    _____
___________      ____________   ______________    _____
```

AE Services Server

Valid entries	Usage
characters	Enter a valid AE Services Server name. The name must match the AE Services server machine name. Each name must be unique on this screen.

Enabled

Valid entries	Usage
y/n	Enter **y** to enable the AE Services server

Password

Valid entries	Usage
12-16 alphanumeric characters; must contain at least one alpha character and one numeric character	Enter a password for future access to this screen.

Server ID

This field is display only.

Valid entries	Usage
1 to **16**	Displays the number assigned to this server.

Status

This field is display only.

Valid entries	Usage
idle	The AE Services server is connected to Communication Manager.
in-use	The AE Services server is not connected to Communication Manager.
blank	No AE Server is administered.

ISDN Network Facilities

See **Network Facilities** screen.

ISDN Numbering Calling Party Number Conversion for Tandem Calls

Tandem calls that route to the public network cannot always provide the correct calling party information, resulting in loss of caller ID information. Communication Manager provides a way of modifying the calling party number on a tandem call that lands in the public network.

Use the Calling Party Number Conversion for Tandem Calls screen to administer calling party number formats for tandem calls. To generate a calling party number for the public network, the system compares the incoming calling party number to the sets of calling party lengths, calling party prefixes, and trunk groups. When a match is found, the calling party number is constructed by deleting digits identified in the **Delete** field on the Calling Party Number Conversion for Tandem Calls screen, and then inserting the digits specified in the **Insert** field. The numbering format specified in the **Format** field is used to determine the encoding of the NPI and TON fields for the calling party number.

Entries on this screen are only exercised if the **Modify Tandem Calling Number** field on the ISDN Trunk Group screen is set to **y**. To access the Calling Party Number Conversion for Tandem Calls screen, type `change tandem-calling-party-number`. Press **Enter**.

Field descriptions for page 1

Figure 156: Calling Party Number Conversion for Tandem Calls screen

```
change tandem-calling-party-number                              Page   1 of   x

                    CALLING PARTY NUMBER CONVERSION
                         FOR TANDEM CALLS

         CPN              Trk
   Len   Prefix           Grp(s)   Delete    Insert             Format

    5    22               12-99    1         732852             natl-pub
    7    5381234          2        0         303                lev0-pvt
    __   _______________  _____    __        _______________    ________
    __   _______________  _____    __        _______________    ________
    __   _______________  _____    __        _______________    ________
    __   _______________  _____    __        _______________    ________
    __   _______________  _____    __        _______________    ________
    __   _______________  _____    __        _______________    ________
    __   _______________  _____    __        _______________    ________
    __   _______________  _____    __        _______________    ________
    __   _______________  _____    __        _______________    ________
    __   _______________  _____    __        _______________    ________
    __   _______________  _____    __        _______________    ________
    __   _______________  _____    __        _______________    ________
    __   _______________  _____    __        _______________    ________
```

CPN Len

Use the **Calling Party Length** field to enter the total number of digits in the calling party number.

Valid entries	Usage
1 to **15**	Enter a number between 1 and 15 to indicate calling party number length.
blank	This is the default. Leave blank when deleting an entry.

CPN Prefix

Use the **Calling Party Prefix** field to enter the prefix of the tandem calling party number.

Valid entries	Usage
any combination of digits 0 to 9, up to 15 digits	Enter up to 15 digits to indicate the calling party prefix.
blank	Indicates that all trunk groups are valid provided the **Modify Tandem Calling Party** field on the ISDN Trunk Group screen is set to **y**. A specific calling party number digit string match is not required, provided other matching criteria for tandem calling party number modification are met. This is the default.

Trk Grp(s)

Use the **Trunk Groups** field to enter the ISDN trunk group number.

Valid entries	Usage
Valid trunk group or range of group numbers	Enter an ISDN trunk group number, or a range (x to y) of group numbers.
blank	Indicates that all trunk groups are valid provided the **Modify Tandem Calling Number** field on the ISDN Trunk Group screen is set to **y**. This is the default.

Delete

Use the **Delete Digits** field to enter the digits to delete in modifying the tandem calling party number.

Valid entries	Usage
1 to **15**	Enter a valid number of deleted digits up to 15.
all	Enter all to indicate that all digits are deleted.
blank	Indicates that all trunk groups are valid provided the **Modify Tandem Calling Party** field on the ISDN Trunk Group screen is set to **y**. No digits are deleted from the received calling party number. This is the default.

Insert

Use the **Insert Digits** field to enter the digits to insert in modifying the tandem calling party number.

Valid entries	Usage
any combination of digits 0 to 9, up to 15 digits	Enter a valid number of between 1 and 15 to indicate the number of digits to insert.
blank	Indicates that all trunk groups are valid provided the **Modify Tandem Calling Party** field on the ISDN Trunk Group screen is set to **y**. The received calling party number is not prefixed with any digits. This is the default.

Number Format

Use the **Number Format** field to enter the numbering format to use in modifying the tandem calling party number.

Valid entries	Usage
intl-pub, lev0-pvt, lev1-pvt, lev2-pvt, locl-pub, natl-pub, pub-unk, unk-unk	Enter the appropriate format for the tandem calling number.
blank	Indicates that all trunk groups are valid provided the **Modify Tandem Calling Party** field on the ISDN Trunk Group screen is set to **y**. The numbering format information is not modified.

Note:
The following end validation checks should be performed for this screen:

- The length of the calling party number (the combination of CPN length, deleted digits and inserted digits) cannot exceed 15 digits.
- The number of deleted digits cannot be greater than the CPN length.
- The number of digits entered for the CPN prefix cannot be greater than the CPN length.

If any of the above are true, an error message displays.

ISDN Trunk Group

This screen assigns an Integrated Services Digital Network (ISDN) trunk group that supports the ISDN and Call-by-Call Service Selection service selection features. The trunk group provides end-to-end digital connectivity and supports a wide range of services including voice and non-voice services, to which users have access by a limited set of CCITT-defined, standard multipurpose interfaces.

The ISDN trunk group can contain ISDN-PRI or ISDN-BRI interfaces. However, it is not possible to use the two types of interfaces in the same trunk groups. The type of interface is chosen when the trunk members are assigned to the trunk group.

When ISDN-PRI interfaces are used on ISDN trunk groups, they can also be used to support the Wideband Switching feature. This is intended to work with the H0 (384 Kbps), H11 (1536 Kbps), H12 (1920 Kbps), and NXDS0 (128 to 1984 Kbps) data services, and to support high-speed video conferencing and data applications.

For descriptions of the screens and fields used with non-ISDN trunks, see Trunk Group on page 971.

Field descriptions for page 1

The field descriptions which follow are for fields that are unique to the ISDN Trunk Group screen. For descriptions of other Trunk Group fields, see Trunk Group on page 971.

Figure 157: ISDN Trunk Group screen

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

Group Number: 1                    Group Type: isdn          CDR Reports: y
  Group Name: OUTSIDE CALL                COR: 1        TN: 1         TAC:
   Direction: outgoing       Outgoing Display? n        Carrier Medium:
 Dial Access? n                Busy Threshold: 255
Queue Length: 0
Service Type:                       Auth Code:            TestCall ITC: rest
                         Far End Test Line No:
TestCall BCC:                                Member Assignment Method:
                                                      Signaling Group:
                                                    Number of Members:
```

Auth Code

The Auth Code field is available only for incoming or two-way trunk groups if you enable the **Authorization Codes** feature. If you enable the **Auth Code** field, Communication Manager performs an auth code check for the incoming trunk call that is routed over another trunk. For more infomation on the **Authorization Codes** feature, see Authorization Code - PIN Checking for Private Calls **on page 76**.

Note: The **Auth Code** field is unavailable if:

- the **Group Type** field is **tandem**
- the **Group Type** field is **ISDN** and the **Service Type** field is **tandem**.

In these situations, permissions of the caller are transmitted using Traveling Class Mark.

The following table provides the situations when the caller must enter an auth code.

Calling party	Called party	Facility Restriction Level (FRL) check?	Auth Code, required if
Station	Station	No	No
	Trunk	Yes, if the trunk was accessed by Route Pattern and not by Trunk Access Code	the FRL of the calling station is less than the FRL of the outgoing route pattern
Incoming Trunk	Station	No	the Auth Code field is enabled on the incoming trunk group
	Trunk	Yes, if the trunk was accessed by Route Pattern and not by Trunk Access Code	the FRL of the incoming trunk is less than the FRL of the outgoing route pattern or the Auth Code field is enabled on the incoming trunk group

Carrier Medium

This field lets you to specify the type of transport medium interface used for the ISDN trunk group. Appears only when the **Group Type** field is **isdn** and, on the System Parameters Customer-Options (Optional Features) screen, either the **Async. Transfer Mode (ATM) Trunking** field or the **H.323** field is set to **y**.

Valid entries	Usage
ATM	The trunk is implemented via the ATM Interface circuit pack.
H.323	The trunk is implemented as an H.323 trunk group.
PRI/BRI	The trunk is implemented as a standard DS1 or BRI interface.

Charge Advice

Use this field to accumulate and access charge information about a call. You already must have set the **CDR Reports** field to **y** or **r** (ring-intvl) before changing this field from its default of **none**. Remember that receiving Advice of Charge during the call (administered as "automatic" or "during-on-request") affects system performance because of the increased ISDN message activity on the signaling channel, which might reduce the maximum call capacity.

Valid entries	Usage
none	Enter **none** if you do not want the system to collect Advice of Charge information for this trunk group.
automatic	Enter **automatic** only if your public network sends Advice of Charge information automatically.
end-on-request	Enter **end-on-request** if Communication Manager must request charge information with each call, and you want to receive only the final call charge.
during-on-request	Enter **during-on-request** if Communication Manager must request charge information with each call, and you want charges to display during and at the end of a call.

Far End Test Line No.

Specifies the number sent to the far-end's ISDN test line extension. When the `test trunk long` command is issued, this exact number is sent to the far-end to establish a call that tests the integrity of the trunk member under test. The number does not pass through routing or undergo digit manipulation. The digits entered must be what the far-end expects. For example, for an ISDN tandem trunk, the far-end test number should be a 7-digit ETN (Electronic Tandem Network) number.

Valid entries	Usage
Up to 15 digits	Enter a code to test signaling channel
blank	

Incoming Calling Number Insert

Valid entries	Usage
Enter up to 15 characters (**0** to **9**), **all**, or blank	Enter the number of digits to insert in the calling party number for all incoming calls on this trunk group.

Incoming Calling Number Format

This field indicates the TON/NPI encoding applied to CPN information modified by the CLI Prefix feature. This encoding does not apply to calls originating locally. The **Numbering Format** field on page 2 of this screen applies to calls originated from this server running Communication Manager.

If this field is blank, Communication Manager passes on the encoding received in the incoming setup message. If the incoming setup message did not contain CPN information and digits are added, the outgoing message will contain these digits. If the **Format** field is blank in this case, the value defaults to **pub-unk**.

If the **Format** field on page 2 of this screen is also administered as **unknown**, the trunk group is modified to **unk-unk** encoding of the TON/NPI. Therefore, this field also must contain a value other than **unknown**.

Note:
The values for this field map to the **Type of Numbering (TON)** and **Numbering Plan Identifier (NPI)** values shown below.

Valid entries	Type of numbering (TON)	Numbering plan identifier (NPI)
blank	incoming TON unmodified	incoming NPI unmodified
natl-pub	national(2)	E.164(1)
intl-pub	international(1)	E.164(1)
locl-pub	local/subscriber(4)	E.164(1)
pub-unk	unknown(0)	E.164(1)
lev0-pvt	local(4)	Private Numbering Plan - PNP(9)
lev1-pvt	Regional Level 1(2)	Private Numbering Plan - PNP(9)
lev2-pvt	Regional Level 2(1)	Private Numbering Plan - PNP(9)
unk-unk	unknown(0)	unknown(0)

Member Assignment Method

Appears when **Carrier Medium** on the Trunk Group screen is **H.323**.

Valid entries	Usage
manual	Default. Users manually assign trunk members to a signaling group.
auto	The system automatically generates members to a specific signaling group. Entering **Auto** causes the **Signaling Group** and **Number of Members** fields to appear.

Number of Members

Appears when **Carrier Medium** on the Trunk Group screen is **H.323** and **Member Assignment Method** is **auto**. Indicates the number of virtual trunk members to be automatically assigned to the signaling group number entered in the Signaling Group field.

Valid entries	Usage
0 to **255**	Enter the number of trunks assigned to this signaling group. Default is **0**.

Service Type

Indicates the service for which this trunk group is dedicated. The following table provides a listing of predefined entries. In addition to the Services/Features listed in this table, any user-defined **Facility Type** of **0** (feature) or **1** (service) on the **Network Facilities** screen is allowed.

As many as 10 (for Avaya DEFINITY Server CSI) ISDN trunk groups can have this field administered as **cbc**.

Valid entries	Usage
access	A tie trunk giving access to an Electronic Tandem Network.
accunet	ACCUNET Switched Digital Service — part of ACI (AT&T Communications ISDN) phase 2.
cbc	Call-by-Call service — provides different dial plans for different services on an ISDN trunk group. Indicates this trunk group is used by the Call-By-Call Service Selection feature.
dmi-mos	Digital multiplexed interface — message oriented signaling.
i800	International 800 Service — allows a subscriber to receive international calls without a charge to the call originating party.
inwats	INWATS — provides OUTWATS-like pricing and service for incoming calls.
lds	Long-Distance Service — part of ACI (AT&T Communications ISDN) phase 2.
megacom	MEGACOM Service — an AT&T communications service that provides unbanded long-distance services using special access (switch to 4ESS switch) from an AT&T communications node.
mega800	MEGACOM 800 Service — an AT&T communications service that provides unbanded 800 service using special access (4ESS switch to switch) from an AT&T communications node.
multiquest	AT&T MULTIQUEST Telecommunications Service — dial 700 service. A terminating-user's service that supports interactive voice service between callers at switched-access locations and service provides directly connected to the AT&T Switched Network (ASN).
operator	Network Operator — provides access to the network operator.
outwats-bnd	OUTWATS Band — WATS is a voice-grade service providing both voice and low speed data transmission capabilities from the user location to defined service areas referred to as bands; the widest band is 5.
public-ntwrk	Public network calls — It is the equivalent of CO (outgoing), DID, or DIOD trunk groups. If Service Type is **public-ntwrk**, Dial Access can be set to **y**.
	1 of 2

Valid entries	Usage
sddn	Software Defined Data Network — provides a virtual private line connectivity via the AT&T switched network (4ESS switches). Services include voice, data, and video applications. These services complement the SDN service. Do not use for DCS with Rerouting.
sdn	Software Defined Network (SDN) — an AT&T communications offering that provides a virtual private network using the public switched network. SDN can carry voice and data between customer locations as well as off-net locations.
sub-operator	Presubscribed Common Carrier Operator — provides access to the presubscribed common carrier operator.
tandem	Tandem tie trunks integral to an ETN
tie	Tie trunks — general purpose
wats-max-bnd	Maximum Banded Wats — a WATS-like offering for which a user's calls are billed at the highest WATS band subscribed to by users.
	2 of 2

Signaling Group

Appears when **Carrier Medium** on the Trunk Group screen is **H.323** and **Member Assignment Method** is **auto**.

Valid entries	Usage
1 to **650** or blank	Enter assigned h.323 or SIP Enablement Services (SES) signaling group number between **1** and **650**, or blank.

TestCall BCC

Indicates the Bearer Capability Code (BCC) used for the ISDN test call.

Valid entries	Usage
0	Voice
1	Mode 1
2	Mode 2 Asynchronous
4	Mode 0

Testcall ITC

Controls the encoding of the Information Transfer Capability (ITC) codepoint of the bearer capability Information Element (IE) in the SETUP message when generating an ISDN test call. Allowed values are **rest** (restricted) and **unre** (unrestricted).

Note:
ISDN Testcall feature has no routing, so a testcall is never blocked due to an incompatible ITC.

Testcall Service

Specifies the call-by-call selection for an ISDN test call. Only appears if the **Service Type** field is **cbc**. Valid entries are all of the services listed in Service Type on page 544, excluding **sddn** or any new **Facility Type** of **0** (feature), **1** (service), or **3** (outgoing) that is defined by users on the Network Specific Facility Encoding screen.

Usage Alloc

Appears when the **Service Type** field is **cbc**.

Valid entries	Usage
y/n	Enter **y** to allocate service provided by the trunk group. Use **y** to enhance Network Call Redirection. When you enter **y**, the CBC Trunk Group Usage Allocation Plans screen and the **CBC Trunk Group Usage Allocation Plan Assignment Schedule** appear.

Field descriptions for page 2

The field descriptions which follow are for fields that are unique to the ISDN Trunk Group screen. For descriptions of other Trunk Group fields, see Trunk Group on page 971.

Figure 158: ISDN Trunk Group screen

```
add trunk-group next                                            Page   2 of  x
      Group Type: isdn                 Trunk Type:

TRUNK PARAMETERS
         Codeset to Send Display: 6     Codeset to Send National IEs: 6
        Max Message Size to Send: 260   Charge Advice: none
  Supplementary Service Protocol: a     Digit Handling (in/out): enbloc/enbloc

            Trunk Hunt: cyclical
                                                   Digital Loss Group: 13

              Bit Rate: 1200         Synchronization: async    Duplex: full
 Disconnect Supervision -        Out? n
 Answer Supervision Timeout: 0
          Administer Timers? n      CONNECT Reliable When Call Leaves ISDN? n
```

Administer Timers

This field is displayed for all trunk group types except **cpe**, **h.323**, and **sip**. When this field is **y**, the Administrable Timers page is available to administer timer values.

Valid entries	Usage
y/n	Enter **y** to allow administration of timers on this trunk group. For **Group Type isdn**, the default value is **n**. For all other trunk group types, the default is **y**.

Codeset to Send Display

This field defines the codeset for sending the information element for display. The value depends on the type of server/switch to which the user is connected.

Valid entries	Usage
0	CCITT (non-Communication Manager equipment).
6	Any other than CCITT or System 85 R2V4, 4E11.
7	System 85 R2V4, 4E11.

Codeset to Send National IEs

This field defines the codeset for sending the information element (IE) for national IEs. National IEs include all IEs previously sent only in code set 6 (such as DCS IE). Now these national IEs, including Traveling Class Marks (TCMs) and Lookahead Interflow (LAI), can be sent in code set 6 or 7. The value depends on the type of server/switch the user is connected to.

Valid entries	Usage
6	Other types.
7	System 85 R2V4, 4E11, or newer Avaya S8XXX Server types.

Note:
A Traveling Class Mark (that is, the user's FRL or the user's trunk group FRL) is passed between tandem nodes in an ETN in the setup message only when the **Service Type** field is **tandem**. It then is used by the distant tandem switch to permit access to facilities consistent with the originating user's privileges.

CONNECT Reliable When Call Leaves ISDN

This field appears when **Group Type** is **ISDN**. The value tells the Communication Manager server whether a CONNECT received on an outgoing call that is not end-to-end ISDN is a reliable indication that the far end has answered the call.

Valid entries	Usage
y	CONNECT is considered as a reliable answer.
n	This is the default (works as before). If a call is not end-to-end ISDN, the CONNECT message is considered unreliable (that is, it may be the result of a timer expiring). If the call was originated by a Call Center adjunct, a Call Classifier may be used instead to determine whether the call has been answered.

Digit Handling (in/out)

This field defines whether overlap receiving and overlap sending features are enabled.

Valid entries	Usage
enbloc/enbloc	Set the field to **overlap** when you want overlap receiving or overlap sending. Set to **enbloc** when you do not want these features enabled. The first field value indicates digit receiving and the second value indicates digit sending.
enbloc/overlap	
overlap/enbloc	
overlap/overlap	

Without overlap receiving or sending enabled, the digits on incoming and outgoing calls are sent enbloc. If the **Digit Handling** field is **overlap/enbloc** or **overlap/overlap**, the following results:

- Incoming Call Handling Treatment table does not appear
- The **Digit Treatment** and **Digits** fields appear
- Warning message indicates that all **Incoming Call Handling** entries are removed when screen is submitted
- When screen is submitted with these values, all **Incoming Call Handling** entries are removed

Group Type

Displays the type of trunk group selected for this field on page 1 of the Trunk Group screen. For details, see the field description for the page 1 **Group Type** field.

Max Message Size to Send

Defines Communication Manager's maximum ISDN message size. Currently, the system can receive 260-byte messages. Valid entries are **128**, **244**, **256**, and **260**.

The following table indicates the expected ISDN-PRI message size from several Lucent Technologies and Avaya Inc. products.

Products	Message Length (octets) Received
4ESS (4E11)	256
4ESS (4E13)	256
4ESS (4E14)	256
5ESS (5E4)	244

Products	Message Length (octets) Received
5ESS (5E5)	244
5ESS (5E6)	244
System 75 (all)	260
System 85 (R2V4)	128
System 85 (R2V5)	260
System 85 (R2V6)	260

Supplementary Service Protocol

Indicates which supplementary service protocol to use for services over this trunk group. Supplementary Service protocols are mutually exclusive.

Valid entries	Usage
a	National
b	ISO/ETSI QSIG Private Network. Also used for SBS signaling trunks.
c	ETSI public network
d	European Computer Manufacturer's Association (ECMA) QSIG private network (supports only Name Identification and Additional Network Feature Transit Counter (ANF-TC))
e	DCS with Rerouting ● Do not use the **Service Type** field entry of **dmi-mos** or **sddn** with this option. ● Set the Used for DCS field (on page 2) to **y**.
f	ISDN Feature Plus Public network feature plus signaling.
g	ANSI. Available only if, on the System Parameters Customer-Options (Optional Features) screen, the **ISDN-PRI** or **ISDN-BRI** field is **y** or the **Used for DCS** field is **y**.

Trunk Hunt

Communication Manager performs a trunk hunt when searching for available channels within a facility in an ISDN trunk group. With both **ascend** and **descend**, all trunks within an ISDN trunk group are selected based on this field and without regard to the order in which trunks are administered within the trunk group. When using ISDN-BRI interfaces, only **cyclical** is allowed.

Valid entries	Usage
ascend	Enter to enable a linear trunk hunt search from the lowest to highest numbered channels.
cyclical	Enter to enable a circular trunk hunt based on the sequence the trunks were administered within the trunk group.
descend	Enter for a linear trunk hunt search from the highest to lowest numbered channels.

Note:
The cyclical option cannot be set if the trunk group using ISDN-PRI interfaces is to be used for Wideband operations (the **Wideband Support** field set to **y**).

The search can be administered per ISDN-PRI trunk group, but it infers the direction of search within all ISDN-PRI facilities (or portions of those facilities) administered within the trunk group.

Trunk Type

Displays the type of trunk selected for this field on page 1 of the Trunk Group screen. For details, see the field description for the page 1 **Trunk Type (in/out)** field.

Field descriptions for page 3

The field descriptions which follow are for fields that are unique to the ISDN Trunk Group screen. For descriptions of other Trunk Group fields, see Trunk Group on page 971.

Figure 159: ISDN Trunk Group screen

```
add trunk-group next                                          Page 3 of x
                                 TRUNK FEATURES
           ACA Assignment? n           Measured: none____
                                                        Maintenance Tests? y
                                Data Restriction? n
                                       Send Name:

    Abandoned Call Search? n
    Suppress # Outpulsing? n

        Charge Conversion: 1
            Decimal Point: none
          Currency Symbol:
              Charge Type: units
                                       Per Call CPN Blocking Code:
                                     Per Call CPN Unblocking Code:

              Outgoing ANI:                      DS1 Echo Cancellation? n

Apply Local Ringback? n                     US NI Delayed Calling Name Update? _
Show ANSWERED BY on Display? y
                              Network (Japan) Needs Connect Before Disconnect? _
DSN Term?
```

Apply Local Ringback

This field appears for ISDN and H.323 trunk groups when the **Carrier Medium** field is **PRI_BRI**.

Valid entries	Usage
y/n	Enter **y** to provide a local ringback tone to the caller. The local ringback is removed when the call is connected. Default is **n**.

BSR Reply-best DISC Cause Value

Servers running Communication Manager that are polled as resources in a Best Service Routing application return data to the polling serve in the ISDN DISC message. Since some cause values do not work over some networks, this field sets the cause value that your server return in response to a BSR status poll. If this field is set incorrectly, incoming status poll calls over this trunk group is dropped before any data is returned to the polling server or switch. This field only appears if the **UUI IE Treatment** field is set to **shared**.

Valid entries	Usage
31 (normal-unspecified)	Enter **31** unless otherwise instructed by Avaya or your network service provider.
17 (user-busy)	
16 (normal-call-clearing)	

CAUTION:
In most cases, this field is set to the appropriate value during installation. If you need to change it, your network service provider should be able to help you choose an appropriate value. Don't change this field without the assistance of Avaya or your network service provider.

DCS Signaling

Specifies the means used to send the DCS message. This field only appears if the **Used for DCS** field entry is **y** and the **Service Type** field is anything except **dmi-mos** or **sddn**.

Valid entries	Usage
d-chan	Enter for the DCS over ISDN-PRI D-channel feature.

DCS over D-channel is not supported on trunk groups containing ISDN-BRI interfaces.

- Hop Dgt — The Tandem Hop Limitation and QSIG Additional Network Feature Transit Counter (ANF-TC) features provide a counter that reflects the number of switches (that is, the number of hops) that a call has gone through. The counter increments as a call leaves Communication Manager using tandem facilities. Valid values are **y** and **n**. One or both of the features can be applied to the trunk group depending on the following:
  - If you enter **y** and the **Group Type** field is **tandem** or the **Group Type** field is **isdn** and the **Service Type** field is **tandem**, the Tandem Hop Limitation feature is applied to the trunk group.

– If you enter **y** and you set the **Group Type** field to **isdn**, set the **Service Type** field to **access**, **dmi-mos**, **public-ntwrk**, **tandem**, **tie**, or any of the craft-defined services allowed in the field. Set the **Supplementary Service Protocol** field to **b** or **d**, then the ANF-TC feature is applied to calls on the trunk group.

Note:
The above conditions overlap. If the **Group Type** field is **isdn**, the **Service Type** field is **tandem**, and the **Supplementary Service Protocol** field is **b** or **d**, then both the Tandem Hop Limitation and ANF-TC features are applied to calls on the trunk group.

– If both features are applied to calls on the trunk group, ANF-TC takes precedence. In situations where Communication Manager serves as an Incoming or Outgoing Gateway, either feature uses the hop count and transit information provided by the other.

Decimal Point

This field appears for CO, DIOD, FX, and WATS trunk groups when the **Direction** field is **outgoing** or **two-way**. For ISDN trunk groups, it appears when the **Charge Advice** field is *not* **none**. Chose the appropriate representation for a decimal point as it appears on telephone displays. Entering **comma** or **period** in this field divides the charge value by 100.

Note:
On a QSIG trunk group, unlike other trunk groups, the **Decimal Point** field does not drive whether a decimal point appears on the calling display. Instead, it tells what symbol should be displayed if the QSIG AOC received has a 1/10 or 1/100 or 1/1000 Multiplier. If the received charge contains no decimals, no decimal point is displayed (that is, the administered decimal point is ignored for charge information received with no decimals). On an upgrade from a QSIG trunk group with the **Decimal Point** field administered as **none**, the field defaults to **period**.

Valid entries	Usage
comma	If the received charge contains decimals, the charge is displayed at the calling endpoint’s display with a comma as the decimal point.
period	This is the default. If the received charge contains decimals, the charge is displayed at the calling endpoint’s display with a period as the decimal point.
none	No decimal point is displayed.

Maximum Size of UUI IE Contents

This field appears when the **UUI IE Treatment** field is **shared**.

Valid entries	Usage
32 to **128**	Enter the maximum number of bytes of user information that the network supports.

Modify Tandem Calling Number

This field appears when **Trunk Group Type** is **ISDN**, Direction is either **Outgoing** or **Two-way**, **Carrier Medium** is **PRI/BRI** or **IP**, and **Send Calling Number** is either **y** or **r**. It is used to control whether call processing processes the entries on the ISDN -Tandem Calling Party Number screen.

Valid entries	Usage
y/n	Enter **y** to modify the calling party number IE in the format specified on the ISDN-Tandem Calling Party Number screen. Default is **n**.

NCA-TSC Trunk Member

Identifies the trunk member whose D-channel is used to route tandem NCA-TSCs or QSIG CISCs. Value range for this field is from 1 to the maximum number of members per trunk group supported on the server/switch, or blank.

Network Call Redirection

This field is administrable if, on the System Parameters Customer-Options (Optional Features) screen, the **ISDN-PRI** field is **y**, the **ISDN Network Call Redirection** field is **y**, and on the ISDN Trunk Group screen, the **Supplementary Service Protocol** field is **a, c,** or **g**. Whenever the **Supplementary Service Protocol** field is changed, this field resets to **none** to prevent an inadvertent incorrect value.

Following are the allowed settings for TBCT or MCI/Verizon NCT:

	Telcordia TBCT	ANSI-1998 ECT
G3 Version	V11 or later	V11 or later
Customer Options	ISDN-PRI ISDN Network Call Redirection	ISDN PRI ISDN Network Call Redirection
DS1 Country Protocol	1b or 1d	Any, but typically 1a

	Telcordia TBCT	ANSI-1998 ECT
Trunk Group Supplementary Service Protocol	a	g
Network Call Redirection keyword	Telcordia-TBCT	ANSI Transfer (MCI/Verizon) Enhanced ANSI Transfer Nortel-Transfer (Nortel DMS)
Trunk Group member's signaling group is Network Call Transfer	y	y

Following are the allowed settings for ETSI protocol:

G3 Version	V12 or later
Customer Options	ISDN PRI ISDN Network Call Redirection
Trunk Group Supplementary Service Protocol	c
Network Call Redirection keyword	implicit-etsi-ect explicit-etsi-ect deflect
Trunk Group member's signaling group is Network Call Transfer	y
Call Center Release	8.3 or later

Network (Japan) Needs Connect Before Disconnect

Sends an ISDN Connect message just prior to the Disconnect message.

Numbering Format

This field appears if the **Send Calling Number** field is **y** or **r** or the **Send Connected Number** field is **y** or **r**. This specifies the encoding of Numbering Plan Indicator for identification purposes in the Calling Number and/or Connected Number IEs, and in the QSIG Party Number. Valid entries are **public**, **unknown**, **private**, and **unk-pvt**. **Public** indicates that the number plan according to CCITT Recommendation E.164 is used and that the Type of Number is national. **Unknown** indicates the Numbering Plan Indicator is unknown and that the Type of Number is unknown. **Private** indicates the Numbering Plan Indicator is PNP and the Type of Number is determined from the Numbering - Private Format screen. An entry of **unk-pvt** also determines the **Type of Number** from the Numbering - Private Format screen, but the **Numbering Plan Indicator** is unknown.

Outgoing Channel ID Encoding

Appears only if the **Group Type** field is **isdn** and the **Service Type** field is anything except **dmi-mos** or **sddn**. Determines whether to encode the Channel ID IE as preferred or exclusive. Blank is not a valid entry. Defaults are determined as follows:

If the **Group Type** field is **isdn** and the **Used for DCS** field is **y**, default is **exclusive**.

If the **Group Type** field is **isdn** and the **Used for DCS** field is **n**, default is **preferred**.

If the **Group Type** field is not **isdn** or it is **isdn**, but the **Used for DCS** field does not appear, default is **preferred**.

Path Replacement Method

Appears when either the **ISDN-PRI trunk** or the **ISDN-BRI trunk** fields and the **Basic Call Setup** and **Supplementary Services with Rerouting** fields are set to **y** on the System Parameters Customer-Options (Optional Features) screen and when the **Supplementary Service Protocol** is either **b** or **e** and the **Group Type** field is **isdn** on the ISDN Trunk Group screen.

Valid entries	Usage
better-route	Uses the most economical route, for example, the reconfigured call does not use the same trunk group as the original call.
always	Always reconfigures the call regardless of the trunk group used.

Replace Restricted Numbers

Appears when the **Group Type** field is **isdn**. Indicates whether to replace restricted numbers with administrable strings for incoming and outgoing calls assigned to the specified trunk group. This field applies to BRI, PRI, H.323 and SIP trunks.

Valid entries	Usage
y/n	Enter **y** for the display to be replaced regardless of the service type of the trunk.

Replace Unavailable Numbers

Appears when the **Group Type** field is **isdn** or **sip**. Indicates whether to replace unavailable numbers with administrable strings for incoming and outgoing calls assigned to the specified trunk group. This field applies to BRI/PRI, H.323, and SIP Enablement Services (SES) trunks. This field also applies to analog trunks if, on the System Parameters Customer Options screen, **Analog Trunk Incoming Call ID** is **y**, and on the Trunk Group screen, **Receive Analog Incoming Call ID** is set to any value except **disabled**.

Valid entries	Usage
y/n	Enter **y** for the display to be replaced regardless of the service type of the trunk.

SBS

Appears when the **Local Country Code and International Access Code** fields are administered on the Feature-Related System-Parameters screen and when the **Supplementary Service Protocol** is **b** and the **Group Type** field is **isdn** and **Carrier Medium** is **IP** and **Dial Access** is **n** on page 1 of the ISDN Trunk Group screen.

Valid entries	Usage
y/n	Enter **y** to enable Separation of Bearer and Signaling (SBS) for the trunk group. The default is **n** (SBS is not enabled).

Send Called/Busy/Connected Number

Appears if the **QSIG Value-Added** field on the Trunk Group screen is **y**. Specifies if the dialed number, whether called (ringing), busy (busy tone), or connected (answered) is sent on incoming or tandemed ISDN calls. Valid entries are **y**, **n**, or **r** (restricted). If **y** is entered, the ISDN Numbering - Public/Unknown Format screen is accessed to construct the actual number sent, or the ISDN Numbering-Private screen (based on the **Numbering Format** field) is used. If the value is **r**, the connected number is sent "presentation restricted." The **Send Called/Busy/Connected Number** field must be set to **y** in order for the Calling Party Number of an incoming ISDN call to display at the transferred-to station after a QSIG transfer operation.

Send Calling Number

Specifies whether the calling party's number is sent on outgoing or tandemed ISDN calls. Valid entries are y, **n**, or **r** (restricted). If **y** is entered, the ISDN Numbering - Public/Unknown Format screen is accessed to construct the actual number to be sent, or the ISDN Numbering-Private screen (based on the **Numbering Format** field) is used. If the value is **r**, the calling number is sent "presentation restricted."

When **Send Calling Number** is **n**, an incoming number is not tandemed out again. Similarly, when **Send Calling Number** is **r** (restricted), an incoming number is marked restricted when it is tandemed out again. This applies to all Supplementary Service Protocols.

Note:
The ISDN Numbering - Public/Unknown Format screen can override the **Send Calling Number** field entry for any administrable block of extensions.

Send Codeset 6/7 LAI IE

Specifies whether the ISDN trunk should transmit information in Codeset 6/7. If the UUI IE Treatment field is **shared**, then this field should be **n**. Otherwise, the same information will be sent twice and might exceed the message size. Default is **y** for pre-DEFINITY 6.3 compatibility.

Send Connected Number

Appears if the **QSIG Value-Added** field on the Trunk Group screen is **n**. Specifies if the connected party's number is sent on incoming or tandemed ISDN calls. Valid entries are **y**, **n**, or **r** (restricted). If **y** is entered, the ISDN Numbering - Public/Unknown Format screen is accessed to construct the actual number sent, or the ISDN Numbering-Private screen (based on the **Numbering Format** field) is used. If the value is **r**, the connected number is sent "presentation restricted." The **Send Connected Number** field must be set to **y** in order for the Calling Party Number of an incoming ISDN call to display at the transferred-to station after a QSIG transfer operation.

When **Send Connected Number** is **n**, an incoming number is not tandemed out again. Similarly, when **Send Connected Number** is **r** (restricted), an incoming number is marked restricted when it is tandemed out again. This applies to all Supplementary Service Protocols.

Note:
The AT&T Switched Network Protocol does not support restricted displays of connected numbers. Therefore, if you administer the 1a country-protocol/ protocol-version combination on the DS1 Circuit Pack screen, you should not administer the **Send Connected Number** field to **r** (restricted) on the ISDN Trunk Group screen, as this causes display problems.

The ISDN Numbering - Public/Unknown Format screen overrides the **Send Connected Number** field entry for any administrable block of extensions.

Send Name

Specifies whether the calling/connected/called/busy party's administered name, or the name on a redirected call, is sent to the network on outgoing/incoming calls. Valid entries are **y**, **n**, or **r** (restricted). The value **r** indicates that the calling/connected name will be sent by Communication Manager, but will be marked "presentation restricted." This value is valid only if the **Supplementary Service Protocol** field is **a** (national supplementary service), **b** (for called/busy only) or **d** for the QSIG Global Networking Supplementary Service Protocol. When the **Supplementary Service Protocol** field is **e** (DCS with Rerouting), only values of **y** and **n** are permitted. For redirected calls, the value **y** indicates that the name is displayed, while for **n** and **r**, the redirected caller name is not displayed.

When the **Send Name** field in **n**, an incoming name is not tandemed out again if the **Supplementary Service Protocol** field is any value other than **b** (QSIG). Similarly, when **Send Name** is **r** (restricted), an incoming name is marked restricted when it is tandemed out again. However, if the **Supplementary Service Protocol** field is **b** (QSIG), then an incoming name is passed on unchanged and the **Send Name** field is ignored.

Note:
If name information is not administered for the calling station or the connected/called/busy station, the system sends the extension number in place of the name.

Send UCID

Specifies whether or not the trunk should transmit Universal Call IDs. Valid entries are **y** and **n**. Send UCID field does not appear unless the UUI IE Treatment field is set to Shared.

Send UUI IE

Specifies whether to block sending UUI information on a per trunk group basis. The valid entries are y and **n**.

Show ANSWERED BY on Display

This field appears when the **Group Type** field is **isdn pri/bri** or **sip**. Use this field to administer whether or not the words "ANSWERED BY" are displayed in addition to the connected telephone number on calls over this trunk.

Note:
Based on display language settings for stations, "ANSWERED BY" is translated into and displayed in the appropriate language.

Valid entries	Usage
y	When set to **y**, the words "ANSWERED BY" are displayed in addition to the connected telephone number. This is the default.
n	When set to **n**, only the connected telephone number is displayed. This might be preferred when outgoing calls are over a trunk that might be redirected.

US NI Delayed Calling Name Update

Administrable if, on the System Parameters Customer-Options (Optional Features) screen, the **ISDN-PRI** field is **y**, and on the Trunk Group screen, the **Carrier Medium** field is either **PRI/BRI** or **ATM**, and the **Supplementary Service Protocol** field is **a**. This field provides display updates to the terminating telephone for delayed calling party name provided by the network.

Valid entries	Usage
y	If calling name information is received after the incoming call has been delivered to the terminating telephone, there is a display update. **Note:** BRI trunks do not support this value.
n	If calling name information is received after the incoming call has been delivered to the called telephone, there is no display update to the terminating telephone.

UUI IE Treatment

Specifies whether the user Information Element (IE) is shared.

Valid entries	Usage
shared	If the trunk is connected to an Avaya DEFINITY 6.3 (or later) server, or an Avaya S8XXX Server.
service-provider	If the trunk is connected to a pre-DEFINITY 6.3 switch, or service provider functionality is desired.

Wideband Support

Note:
This feature is not supported on the DS1 interfaces on H.248 gateways.

Specifies whether Wideband Switching is supported by this trunk group. Valid entries are **y** or **n**. For ISDN trunk groups containing ISDN-BRI interfaces, the only valid entry is **n**. Otherwise you can administer this field only if the **Wideband Switching** field is **y** on the System Parameters Customer-Options (Optional Features) screen. If set to **y**, the Wideband Support Options page appears. All trunk members must be from TN464C (or later) circuit packs. The trunk members that are supported using DS1 modules on the H.248 gateways (G700/G350) do not provide for "n" X 64kbps wideband channel support. Those interfaces only provide for call connections based on a single B-channel.

Note:
Wideband trunk calls are treated as a single trunk call when Automatic Circuit Assurance (ACA) measurements are taken. This way, if an ACA referral call is generated (for short or long holding time), the wideband call only triggers a single referral call using the lowest B-channel trunk member associated with the wideband channel.

Field descriptions for QSIG Trunk Group Options page

This fields on this screen appear only when **Group Type** is **isdn** and **Supplementary Service Protocol** is **b**. The field descriptions which follow are for fields that are unique to the ISDN Trunk Group screen. For descriptions of other Trunk Group fields, see Trunk Group on page 971.

Figure 160: QSIG Trunk Group Options screen

```
add trunk-group next                                              Page x of y

                        QSIG Trunk Group Options

   TSC Method for Auto Callback? n
           Diversion by Reroute? y
               Path Replacement? y
Path Replacement with Retention? n
        Path Replacement Method: better-route
                           SBS? n
  Display Forwarding Party Name? y
    Character Set for QSIG Name: iso8859-1
               QSIG Value-Added? n
   QSIG-Value Coverage Encoding: proprietary
```

Character Set for QSIG Name

Use this field to set the character set for transmission of QSIG name data for display. This field appears only when **Group Type** is **isdn**, **Supplementary Service Protocol** is **b**, and **Display Character Set** on the [System Parameters Country-Options](#) screen is **Roman**.

Valid entries	Usage
eurofont	The Roman Eurofont character set. This is the default. **NOTE:** Avaya BRI stations support ASCII characters only. They do not support non-ASCII characters, such as Eurofont, Kanafont, or Optrex. Therefore, if you use non-ASCII characters in any Communication Manager **Name** field, such characters do not display correctly on a BRI station.
iso-8859-1	All data (that is, characters) in the Name value transmitted over QSIG are converted from Eurofont (Avaya proprietary encoding) to ISO 8859-1. **Note:** ISO 8859-1, more formally known as ISO/IEC 8859-1, or less formally as Latin-1, is part 1 of ISO/IEC 8859, a standard character encoding defined by ISO. It encodes what it refers to as Latin alphabet no. 1, consisting of 191 characters from the Latin script, each encoded as a single 8-bit code value.

Diversion by Reroute

This field appears only when **Group Type** is **isdn** and **Supplementary Service Protocol** is **b**.

Valid entries	Usage
y	The Diversion by Reroute feature is enabled. Default is **y**.
n	The Diversion by Reroute feature is disabled. Communication Manager does not originate a Diversion/Reroute request over that trunk group, and rejects any Diversion/Reroute request it receives over that trunk group.

Display Forwarding Party Name

This field appears only when **Group Type** is **isdn** and **Supplementary Service Protocol** is **b**.

Valid entries	Usage
y/n	Enter **y** to display the name of the party who is forwarding the call. Default is **y**.

QSIG Value-Added

Valid entries are **y** and **n**. Provides QSIG-VALU services. This field appears only if the **Value-Added (VALU)** field on the System Parameters Customer-Options (Optional Features) screen is **y**. This field can be set to **y** only if the **Supplementary Service Protocol** field on the System Parameters Customer-Options (Optional Features) screen is **b**.

QSIG-Value Coverage Encoding

Use this field to indicate the encoding method to use to encode DL1, DL2, and DL3 extensions. This field appears only when **Group Type** is **isdn**, **Supplementary Service Protocol** is **b**, and **QSIG Value-Added** is **y**.

Valid entries	Usage
proprietary	Communication Manager sends extension information in the normal manner. This is the default.
standard	In addition to normal extension information, Communication Manager also sends the data part (as null) of the extension.

Path Replacement

This field appears only when **Group Type** is **isdn** and **Supplementary Service Protocol** is **b**.

Valid entries	Usage
y	The Path Replacement feature is enabled. Default is **y**.
n	The Path Replacement feature is disabled. Communication Manager does not originate a Path Replacement request over that trunk group, and rejects any Path Replacement request it receives over that trunk group.

Path Replacement Method

Appears when the following fields are set on the Trunk Group screen: trunk **Group Type** is **ISDN**, **Supplementary Service Protocol** is **b** or **e**, the **Path Replacement with Retention** is **n**, and the **Supplementary Services with Rerouting** field or the **DCS with Rerouting** field on the System Parameters Customer-Options (Optional Features) screen is **y**.

Valid entries	Usage
always	Use any QSIG (SSB) trunk group as the replacement trunk group. A new call is always originated, even when the original trunk group is determined to be the replacement trunk group.
BR (better route)	Route pattern preferences help determine trunk group path replacement. The original trunk group is retained if the **Path Replacement with Retention** field is **y**. Path replacement fails (and the original trunk group is retained) if the **Path Replacement with Retention** field is **n**.

Path Replacement with Retention

Appears when the following fields are set on the Trunk Group screen: trunk **Group Type** is **ISDN**, **Supplementary Service Protocol** is **b** or **e**, and the **Supplementary Services with Rerouting** field or the **DCS with Rerouting** field on the System Parameters Customer-Options (Optional Features) screen is **y**.

Valid entries	Usage
y/n	Enter **y** to retain the original trunk group. Set to **n** to allow path replacement according to setting on the **Path Replacement Method** field.

TSC Method for Auto Callback

Use this field to control the signaling connection method for the QSIG-TSC when Communication Manager is the terminating or the outgoing gateway PINX. When this field is set to **always-retain**, QSIG Temporary Signaling Connections are always retained for successful call completion activation.

Valid entries	Usage
drop-if-possible	signaling connection is released
always-retain	signaling connection is retained

Field Descriptions for Administrable Timers page

This screen displays only when the **Administer Timers** field on page 2 of the Trunk Group screen is **y**. This screen does not display when for trunks of **Group Type cpe** or **sip**.

Note:
If the ISDN trunk group has a **Carrier Medium** value of **H.323**, or it the trunk group has BRI members, then the Administrable Timers page is not administrable. In these cases, an error message displays when you attempt to submit the screen.

Figure 161: Administrable Timers for ISDN Trunk Group screen

```
add trunk-group next                                                  Page 5 of x
                              ADMINISTRABLE TIMERS








 Programmed Dial Pause(msec): _____



END TO END SIGNALING
    Tone (msec): ____      Pause (msec): 150
```

CAUTION:
Customers: Do not change fields on this page without assistance from Avaya or your network service provider.

Programmed Dial Pause (msec)

This timer is administrable for all outgoing and two-way trunk groups. This timer works with the TN464B (or later), TN767, TN458, TN2140, and TN2242 tie circuit packs. All CO circuit packs that accept administrable timers accept this timer.

Valid entries	Usage
100 to **25500** in increments of 100	Set the exact duration of pauses used during abbreviated dialing, ARS outpulsing, and terminal dialing operations.

END TO END SIGNALING

Pause (msec)

THis field is administrable only if the **Trunk Type** field is blank. All CO, DIOD, and tie circuit packs that accept administrable timers accept this timer. However, this timer is sent only to the following circuit packs: TN464B (or later), TN767, TN436B, TN459B, TN2146, TN2199, and TN2242, and TN429 and TN2184 ports in a DID trunk group.

Valid entries	Usage
20 to **2550** in increments of 10	Enter the minimum acceptable interval (pause) between DTMF tones sent from a hybrid telephone.

Tone (msec)

This field appears only if the **Trunk Type** field is blank. All CO, DIOD, and Tie circuit packs that accept administrable timers accept this timer. This timer is also sent to the following circuit packs: TN464B (or later), TN767, TN436B, TN459B, TN2146, TN2199, TN429, TN2184 ports in a DID trunk group.

Valid entries	Usage
20 to **2550** in increments of 10	Enter the duration of a DTMF tone sent when a button on a hybrid telephone is pressed.

Field descriptions for the Shared UUI Feature Priorities page

The fields in this page show the priorities for each type of information to be forwarded in the Shared UUI. This page appears only on the ISDN trunk group screen when all of the following conditions are met:

- The **UUI IE Treatment** field is **shared**.
- The **Supplementary Service Protocol** field is set to anything except **b**.

The field descriptions which follow are for fields that are unique to the ISDN Trunk Group screen. For descriptions of other Trunk Group fields, see Trunk Group on page 971.

Figure 162: Shared UUI Feature Priorities screen

```
add trunk-group next                                              Page y of x
                       SHARED UUI FEATURE PRIORITIES

                     ASAI: 1

        Universal Call ID: 2

MULTI SITE ROUTING (MSR)

              In-VDN Time: 3
                 VDN Name: 4
         Collected Digits: 5
    Other LAI Information: 6
```

Changing the priorities in this screen might affect whether certain information will be sent.These fields are unique to the ISDN Trunk Group screen.

ASAI

User information from ASAI. Valid entries are **1** to **6** (**1** is high) and blank. If blank, that field's information is not forwarded.

Collected Digits

Digits collected from caller (not including dial-ahead digits). Valid entries are **1** to **6** (**1** is high) and blank. If blank, that field's information is not forwarded.

In-VDN TIme

Number of seconds the call has spent in vector processing. Valid entries are **1** to **6** (**1** is high) and blank. If blank, that field's information is not forwarded.

Other LAI Information

Includes the time stamp of when the call entered the current queue, the call's priority level in its current queue, and the type of interflow. Valid entries are **1** to **6** (**1** is high) and blank. If blank, that field's information is not forwarded.

Universal Call ID

Unique tag to identify each call. Valid entries are **1** to **6** (**1** is high) and blank. If blank, that field's information is not forwarded.

VDN Name

Name of the active VDN (also called LAI DNIS). Valid entries are **1** to **6** (**1** is high) and blank. If blank, that field's information is not forwarded.

Field Descriptions for the CBC Trunk Group Usage Allocation page

Appears when the **Service Type** field is **cbc** and the **Usage Alloc** field is **y**. The field descriptions which follow are for fields that are unique to the ISDN Trunk Group screen. For descriptions of other Trunk Group fields, see Trunk Group on page 971.

Figure 163: CBC Trunk Group Usage Allocation screen

```
add trunk-group next                                                 Page y of x
                         CBC TRUNK GROUP USAGE ALLOCATION
Usage Allocation Plan 1    Usage Allocation Plan 2    Usage Allocation Plan 3
                Min# Max#                  Min# Max#                  Min# Max#
Service/Feature Chan Chan  Service/Feature Chan Chan Service/Feature  Chan Chan

_______________  __   __   _______________  __   __   _______________  __   __
_______________  __   __   _______________  __   __   _______________  __   __
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_______________  __   __   _______________  __   __   _______________  __   __
```

The CBC Trunk Group Usage Allocation screen sets a minimum and maximum number of members for up to ten different Services/Features for up to three different Usage Allocation Plans (1 to 3). See Call-by-call Service Selection in *Avaya Aura™ Communication Manager Feature Description and Implementation,* 555-245-205, for a detailed description of Usage Allocation Plans.

Max# Chan

Indicates the maximum number of members of a ISDN trunk group with a **Service Type** field of **cbc** that a particular Service/Feature can use at any given time. This field must be completed if a Service/Feature has been entered in the Incoming Call Handling Treatment Table screen. Valid values are **0** to **99** or blank.

Min# Chan

Indicates the minimum number of members of an ISDN trunk group with a **Service Type** field of **cbc** that a particular Service/Feature can use at any given time. The sum of the minimum number of members for all Service/Features must not exceed the total number of members of the trunk group. Valid values are **0** to **99** or blank.

Service/Feature

Specifies the ISDN Services/Features that can be requested at call setup time when using this trunk group. See the Service Type field description for a list of predefined Services/Features that can be received on a call by call basis. In addition, user-defined service types can be used. Any user-defined **Facility Type** of **0** (feature) or **1** (service), **2** (incoming), or **3** (outgoing) on the **Network Facilities** screen is allowed.See the description of the **Network Facilities** screen for details. The identifier **other** is used for all Services/Features not explicitly specified.

Field descriptions for the CBC Service Trunk Group Allocation Plan Assignment Schedule page

Appears when the **Service Type** field is **cbc** and the **Usage Alloc** field is **y**. The field descriptions which follow are for fields that are unique to the ISDN Trunk Group screen. For descriptions of other Trunk Group fields, see Trunk Group on page 971.

Figure 164: CBC Service Trunk Group Allocation Plan Assignment Schedule screen

```
add trunk-group next                                           Page y of x
            CBC SERVICE TRUNK GROUP ALLOCATION PLAN ASSIGNMENT SCHEDULE

  Usage Method:

         Fixed? y      Allocation Plan Number: 1
     Scheduled? n


Usage Allocation Plan Activation Schedule:

       Act   Plan    Act   Plan    Act   Plan    Act   Plan    Act   Plan    Act   Plan
        Time  #       Time  #       Time  #       Time  #       Time  #       Time  #
Sun     __:__ __     __:__ __     __:__ __     __:__ __     __:__ __     __:__ __
Mon     __:__ __     __:__ __     __:__ __     __:__ __     __:__ __     __:__ __
Tue     __:__ __     __:__ __     __:__ __     __:__ __     __:__ __     __:__ __
Wed     __:__ __     __:__ __     __:__ __     __:__ __     __:__ __     __:__ __
Thu     __:__ __     __:__ __     __:__ __     __:__ __     __:__ __     __:__ __
Fri     __:__ __     __:__ __     __:__ __     __:__ __     __:__ __     __:__ __
Sat     __:__ __     __:__ __     __:__ __     __:__ __     __:__ __     __:__ __
```

The CBC Service Trunk Group Allocation Plan Assignment Schedule screen provides for administering a fixed schedule or administering a schedule that can change up to 6 times a day for each day of the week. This screen determines which CBC Service Trunk Group Allocation Plan is in use at any given time.

Act Time

Indicates the time the usage allocation plan administered in the next field (**Plan #**) will become effective. Enter the time in military time. There must be at least one entry per day. Valid entries are **00:00** through **23:59**.

Allocation Plan Number

Specifies the CBC Trunk Allocation Plan (1 through 3) that is in effect if a fixed usage method has been selected. This field must be assigned if the **Fixed** field is **y**. Valid entries are **1** to **3** or blank.

Fixed

Indicates whether the allocation plan is fixed. If **y** is entered, the plan number entered in the **Allocation Plan Number** field is enabled.

Plan

Specifies the number of the usage allocation plan that is in effect from the activation time until the activation time of the next scheduled plan change. Valid entries are **1** to **3** or blank.

Scheduled

Indicates whether or not the allocation plans is in effect according to the schedule found on this page. If **y** is entered in this field then there must be at least one entry in the schedule.

Field descriptions for the Wideband Support Options page

The Wideband Support Options screen appears immediately before the trunk group member pages. The actual page number varies.The field descriptions which follow are for fields that are unique to the ISDN Trunk Group screen. For descriptions of other Trunk Group fields, see Trunk Group on page 971.

Figure 165: Wideband Support Options screen

```
add trunk-group next                                          Page y of x
                      Wideband Support Options

                              H0? n
                             H11? n
                             H12? n
                           NxDS0? y     Contiguous? n
```

Note:
All B-channels that comprise the wideband call must reside on the same ISDN-PRI facility. Also, all trunk members in an ISDN trunk group with the **Wideband Support** field set to **y** must be from a TN464C (or later) circuit pack.

H0

Enter **y** to specify the ISDN information transfer rate for 384-kbps of data, which is comprised of six B-channels. When a trunk group is administered to support H0, the trunk/hunt algorithm to satisfy a call requiring 384-kbps of bandwidth uses a fixed allocation scheme.

H11

Enter **y** to specify the ISDN information transfer rate for 1536-kbps of data, which is comprised of 24 B-channels. When a trunk group is administered to support H11, the trunk/hunt algorithm to satisfy a call requiring 1536-kbps bandwidth uses a fixed allocation scheme.

H12

Enter **y** to specify the ISDN information transfer rate for 1920-kbps of data, which is comprised of 30 B-channels. When a trunk group is administered to support H12, the trunk/hunt algorithm to satisfy a call requiring 1920-kbps bandwidth uses a fixed allocation scheme.

Contiguous

Specifies whether or not to hunt contiguous NXDS0 channels. This field only appears if the **NxDS0** field is **y**.

The trunk/hunt algorithm to satisfy an NXDS0 call is as follows:

- Enter **y** to specify the "floating" scheme. NXDS0 calls are placed on a contiguous group of B-channels large enough to satisfy the requested bandwidth without constraint on the starting channel (no fixed starting point trunk).

Note:
H0 and NXDS0 "floating" scheme cannot both be **y**.

- Enter **n** to specify the "flexible" scheme. NXDS0 calls are placed on any set of B-channels on the same facility as long as the requested bandwidth is satisfied. There are no constraints such as contiguity of B-channels or fixed starting points.

NxDS0

Enter **y** to specify the "N by DS-zero" multi-rate service.

Field Descriptions for the Group Member Assignments page

The field descriptions which follow are for fields that are unique to the ISDN Trunk Group screen. For descriptions of other Trunk Group fields, see Trunk Group on page 971.

Figure 166: ISDN Group Member Assignments screen

```
add trunk-group next                                               Page y of x
                               TRUNK GROUP
                               Administered Members(min/max): xxx/yyy
GROUP MEMBER ASSIGNMENTS          Total Administered Members: xxx

       Port     Code     Sfx  Name        Night   Sig Grp
    1: _______  _____     _   __________  _____     ___
    2: _______  _____     _   __________  _____     ___
    3: _______  _____     _   __________  _____     ___
    4: _______  _____     _   __________  _____     ___
    5: _______  _____     _   __________  _____     ___
    6: _______  _____     _   __________  _____     ___
    7: _______  _____     _   __________  _____     ___
    8: _______  _____     _   __________  _____     ___
    9: _______  _____     _   __________  _____     ___
   10: _______  _____     _   __________  _____     ___
   11: _______  _____     _   __________  _____     ___
   12: _______  _____     _   __________  _____     ___
   13: _______  _____     _   __________  _____     ___
   14: _______  _____     _   __________  _____     ___
   15: _______  _____     _   __________  _____     ___
```

The total number of pages that make up the Trunk Group screen, and the first page of Group Member Assignments, vary depending on whether the CBC and Wideband Support pages display.

Note:
When supporting DCS, Member Number Assignments must be the same between nodes (Member #1 must be Member #1 at the far-end trunk group).

Port

When using ISDN-BRI interfaces, B-channel 1 is the port number while B channel 2 is the port number plus 16. For example, if B channel 1's port number is 01A1002, then B channel 2's port number is 01A1018.

When using ISDN-PRI interfaces, the port number is the one allied with the B-channel. For example, if the DS1 is located in 01A10, then B channel 1 will be 01A1001, B channel 2 will be 01A1002 and so forth.

Note:
When administering analog trunks connected to a TIM518, physical ports 17-24 are administered as ports 9 to 16 in Communication Manager.

Sig Grp

This field appears when the **Group Type** field is **isdn-pri**. Enter the signaling group of this trunk group member. Valid entries are from **1** to **650**, and must be configured for IP group members. If you administer a port that resides on a DS1 board and that DS1 board belongs to one and only one signaling group, you can leave the **Signaling Group** column blank. Then, when you submit the screen, the appropriate default signaling group number is inserted by Communication Manager. If a DS1 board is assigned to more than one signaling group, then you must enter a signaling group number. You must enter a signaling group if the port is entered as **IP**. A trunk group can contain members from different signaling groups.

Related topics

See ISDN Service in *Avaya Aura™ Communication Manager Feature Description and Implementation,* 555-245-205, for more information Integrated Services Digital Network trunks.

See DS1 Trunk Service in *Avaya Aura™ Communication Manager Feature Description and Implementation,* 555-245-205, for more information.

ISDN-BRI Trunk Circuit Pack

This screen administers an ISDN-BRI circuit pack. See *Avaya Aura™ Communication Manager Hardware Description and Reference*, 555-245-207, for information on the maximum number of ISDN-BRI circuit packs that you can administer.

Figure 167: BRI Trunk screen (using a TN2185 circuit pack)

```
change bri-trunk-board                                          Page 1 of x
                         ISDN-BRI TRUNK CIRCUIT PACK

              Location: 01A09                          Name: ______________
  Interface Companding: a-law_  DCP/Analog Bearer Capability: 3.1kHz
 T3 Timer Length (sec): 15_                 Termination Type: TE


Port  Interface   Side  Cntry/Peer TEI             Synch   Layer 1  Detect
                         Protocol                  Source? Stable?  Slips?
  1:  user_______         12__     0__               n       n        n
  2:  network____         etsi     0__               y       y        y
  3:  user_______         2___     auto              n       y        n
  4:  peer-slave_  b      QSIG     0__               y       y        n
  5:  peer-master  a      QSIG     auto              n       n        n
  6:  ___________         ____     0__               n       y        n
  7.  ___________         ____     0__               n       y        n
  8:  ___________         ____     0__               n       y        n
```

Field descriptions for page 1 (with a TN2185 circuit pack)

Cntry/Peer Protocol

Tells call processing software which ISDN protocol standard is applied.

Valid entries	Usage
1 to **25**	When this field is **10**, **12**, **13**, or **etsi**, the **Protocol Version** field is equivalent to **b** on the DS1 Circuit Pack screen. When the **Cntry/Peer Protocol** field is set **10**, **12**, **13**, or **etsi**, set the **Protocol Version** field to **b**. For all other administered values, the **Protocol Version** sets to **a**.
etsi	
QSIG	When the **Interface** field is **peer-slave** or **peer-master**, this field must be **QSIG**. The choice **QSIG** is valid only when the Interface field is **peer-slave**.
blank	You cannot leave this field blank if the **Interface** field is set to a valid, non-blank value

DCP/Analog Bearer Capability

Valid entries	Usage
3.1kHz	Indicates how to encode the Bearer Capability IE for an outgoing call originated by a DCP or analog endpoint.
speech	

Detect Slips

Valid entries	Usage
y/n	Tells maintenance software whether slips reported by the BRI port should be logged.

ETSI CCBS

This field appears when **Group Type** is **isdn-pri**, and **TSC SS Protocol** is set to **c** for ETSI. The contents of this new column are only administrable if the **TSC SS Protocol** in this row is set to **c**. If the **TSC SS Protocol** is **c**, the default value of the new column for this row is **both (directions)**. For any other TSC Supplementary Service protocol, the default is **none** and the field is read-only.

Valid entries	Usage
none	Interface supports neither incoming nor outgoing ETSI CCBS. This is the default, except when **TSC SS Protocol** is **c**.
inco(ming)	Interface supports only incoming ETSI CCBS.
outg(oing)	Interface supports only outgoing ETSI CCBS.
both (directions)	Interface supports incoming and outgoing ETSI CCBS. When **TSC SS Protocol** is **c**, this is the default. **Note**: When upgrading from a version of Communication Manager that is earlier than 5.1, this is the default.

Interface

Valid entries	Usage
network	Tells call processing software whether a particular port is connected to a user/network or a peer interface. These entries are valid for the TN2185. You can enter **peer-slave** only if the QSIG Basic Call Setup feature is enabled
user	
peer-master	
peer-slave	

Interface Companding

Valid entries	Usage
a-law	Indicates the companding algorithm expected by the system at the far end.
mu-law	

Layer 1 Stable

Valid entries	Usage
y	Tells call processing and maintenance software whether to expect the network to drop Layer 1 when the BRI port is idle. Only the TN2185 can be set to **n**.
n	

Location

This is a display-only field. It shows the TN2185 circuit pack location (PPCSS)

Name

Valid entries	Usage
1 to 15 alpha-numeric characters	This name is used to identify the circuit pack. **NOTE:** Avaya BRI stations support ASCII characters only. They do not support non-ASCII characters, such as Eurofont or Kanafont. Therefore, if you use non-ASCII characters in any Communication Manager **Name** field, such characters do not display correctly on a BRI station.

Port

This is a display-only field. It shows the port number to which parameters administered on the row apply.

Side

Valid entries	Usage
a	Determines how glare conditions are handled when Interface field is **peer-slave**.
b	

Synch Source

The **Synch Source** field applies only for TN2185 boards.

Note:
For MM720 and MM722 bri media modules, the **Synch Source** field does not appear. For the MM720 and MM722, this parameter is configured using the gateway CLI.

Valid entries	Usage
y	When set to **y**, allows a TN2185 board displayed on the Synchronization Plan screen to be entered as the Primary or Secondary synchronization source, if at least one of the ports on that board has **Synch Source?** enabled. Only those ports marked **y** are candidates for clock synchronization with the far-end network.
n	

T3 Timer Length (sec)

Valid entries	Usage
1 to **127**	Tells the TE side how long to wait, in seconds, for an inactive Layer 1 to become active.

TEI

Valid entries	Usage
auto	TEI is assigned automatically by the network.
0	TEI is fixed.

Termination Type

When a MM720 media module is used as a trunk interface, and the MM720 supports both Line side and Trunk side of BRI, use this field to indicate whether the media module is to operate in Terminal or Network termination mode.

Note:
On a MM720 that can function only as a BRI Trunk Media Module (that is, MM720 without the firmware upgrade that supports both line side and trunk side of BRI), this field defaults to **TE** and is display-only.

Valid entries	Usage
TE	Terminal Endpoint termination. The MM720 provides the TE side of the BRI interface. This is the default.
NT	Network Termination. The MM720 provides the NT side of the BRI interface.

Field descriptions for page 1 (TN556B or TN2198 circuit pack)

Figure 168: BRI Trunk screen (with a TN556B or TN2198 circuit pack)

```
change bri-trunk-board                                         Page 1 of x
                         ISDN-BRI TRUNK CIRCUIT PACK

              Location: 01A09                    Name:  ______________
   Interface Companding: a-law_  DCP/Analog Bearer Capability: 3.1kHz
                                            Termination Type: TE

Port  Interface   Side  Cntry/Peer TEI
                        Protocol
  1:  network__           12__      0__
  2:  network__           etsi      0__
  3:  network__           2___      auto
  4:  peer-master  b      QSIG      0__
  5:  peer-master  a      QSIG      auto
  6:  __________          ____      0__
  7.  __________          ____      0__
  8:  __________          ____      0__
  9:  __________          ____      0__
 10:  __________          ____      0__
 11:  __________          ____      0__
 12:  __________          ____      0__
```

The following field descriptions are unique to the ISDN-BRI Circuit Pack screen with a TN556B or TN2198 circuit pack. The following fields do not display with a TN556B or TN2198 circuit pack:

- **T3 Timer Length (sec)**
- **Synch Source**
- **Layer 1 Stable**
- **Detect Slips**

Cntry/Peer Protocol

Tells call processing software which ISDN protocol standard is applied.

Valid entries	Usage
1 to **25**	When this field is **10**, **12**, or **13**, the **Protocol Version** field is equivalent to **b** on the DS1 Circuit Pack screen.
etsi	When this field is **etsi**, the **Protocol Version** field is equivalent to **b** on the DS1 Circuit Pack screen.
QSIG	When the **Interface** field is **peer-master**, this field must be **QSIG**.
blank	You cannot leave this field blank if the **Interface** field is set to a valid, non-blank value.

Interface

Valid entries	Usage
network	Tells call processing software whether a particular port is connected to a user/network or a peer interface. These entries are valid for the TN556B. You can enter **peer-master** only if the QSIG Basic Call Setup feature is enabled
peer-master	

Side

Valid entries	Usage
a	Determines how glare conditions are handled when **Interface** field is peer-slave. This field is not administrable when the Interface field is **network**.
b	

Field descriptions for page 2

Note:
If administering a TN2185 circuit pack, 8 ports appear; otherwise, 12 ports appear.

Figure 169: BRI Trunk screen - Page 2 (using a TN2185 circuit pack)

```
change bri-trunk-board                                     Page 2 of x
                                 ISDN-BRI TRUNK CIRCUIT PACK

 Port  Interwork  XID  Endpt    SPID      Endpt    SPID      Endpt  Max
       Message   Test? Init?               ID                 ID   NCA TSC
   1:  PROGress   y     n    ____________  __   ____________  __     0___
   2:  ALERTing   y     y    908957200000  __   ____________  __     0___
   3:  PROGress   y     y    0001________  __   ____________  __     0___
   4:  PROGress   n     n    ____________  __   ____________  __     0___
   5:  PROGress   n     y    625761449___  01   ____________  __     0___
   6:  PROGress   n     n    ____________  __   ____________  __     0___
   7.  PROGress   n     n    ____________  __   ____________  __     0___
   8:  PROGress   n     n    ____________  __   ____________  __     0___

   Port Directory   Directory         Port Directory   Directory
          Number      Number                 Number      Number
     1:                                 5:
     2:                                 6:
     3:                                 7:
     4:                                 8:
```

Figure 170: BRI Trunk screen - Page 2 (using a TN2198/TN556B circuit pack)

```
change bri-trunk-board                                     Page 2 of x
                                 ISDN-BRI TRUNK CIRCUIT PACK

 Port  Interwork  XID  Endpt    SPID      Endpt                     Max
       Message   Test? Init?               ID                      NCA TSC
   1:  PROGress   n     y    ____________  __                        0___
   2:  ALERTing   n     y    ____________  __                        0___
   3:  PROGress   n     y    ____________  __                        0___
   4:  PROGress   n     y    ____________  __                        0___
   5:  PROGress   n     y    ____________  __                        0___
   6:  PROGress   n     y    ____________  __                        0___
   7.  PROGress   n     y    ____________  __                        0___
   8:  PROGress   n     y    ____________  __                        0___
   9:  PROGress   n     y    ____________  __                        0___
  10:  PROGress   n     y    ____________  __                        0___
  11:  PROGress   n     y    ____________  __                        0___
  12:  PROGress   n     y    ____________  __                        0___
```

Note:
You cannot change the **Endpt Init**, **SPID**, or **Endpt ID** port parameters unless that port is busied out or unadministered. It is possible to change all other fields on this page even if the corresponding port is active.

If the **Interface** field on page 1 contains a valid value when the screen is submitted, the contents of the fields on page 2 for that port are validated. If the **Interface** field is blank when the screen is submitted, the fields on this page for that port reset to their default values.

Directory Number

These 10-digit fields contain the directory numbers assigned to the interface, which it views as being allocated to 2 separate endpoints.

Valid entries	Usage
Any string of 1 to 10 digits	These fields must be administered in pairs. If you enter a value in one field, you must enter a value in the other.

Endpt ID

A 2-digit field containing the Endpoint Identifier expected by the far end. Communication Manager blocks you from changing this field unless the port is busied out or unadministered.

Valid entries	Usage
00 to **62**	Leading zeroes considered significant and not ignored.

Endpt Init

Indicates whether the far end supports endpoint initialization. Communication Manager blocks you from changing this field unless the port is busied out or unadministered.

Valid entries	Usage
y	If set to **y**, the **SPID** field must *not* be blank. Communication Manager blocks you from changing this field and the **SPID** field unless that port is busied out or unadministered.
n	If set to **n**, the **SPID** and **Endpt ID** fields must be blank.

Interworking Message

This field determines what message Communication Manager sends when an incoming ISDN trunk call interworks (is routed over a non-ISDN trunk group).

Valid entries	Usage
PROGress	Normally select this value. PROGress asks the public network to cut through the B-channel and let the caller hear tones such as ringback or busy tone provided over the non-ISDN trunk.
ALERTing	ALERTing causes the public network in many countries to play ringback tone to the caller. Select this value only if the DS1 is connected to the public network, and it is determined that callers hear silence (rather than ringback or busy tone) when a call incoming over the DS1 interworks to a non-ISDN trunk.

Max NCA TSC

Valid entries	Usage
0 to **63**	This 2-digit field gives the maximum number of Non-Call-Associated Temporary Signaling Connections allowed on this BRI D-channel. This field's function is the same as the field with the same name on the Signaling Group screen.

Port

This is a display-only field. It shows the port number to which parameters administered on the row apply.

SPID

A 12-digit field containing the SPID expected by the far end. Communication Manager blocks you from changing this field unless the port is busied out or unadministered. The only protocol supported for SPID initialization is Bellcore (Country Code 1). Trunks will not be put in service if SPID installation is not successful.

Valid entries	Usage
Any string of 1 to 12 digits	Leading zeroes considered significant and not ignored.

XID Test

Valid entries	Usage
y/n	Indicates whether the far end supports the Layer 2 XID test.

ISDN-BRI Trunk Member Administration

Administer BRI trunk members using the following scheme to address the individual B-channels:

- B-channel 1 uses the port address of the BRI Trunk Port.
- B-channel 2 uses the port address of B-channel 1 incremented by 16.

When adding a BRI trunk to an ISDN trunk-group, Communication Manager blocks you from administering a Signaling Group for that trunk member.

Communication Manager blocks you from administering a BRI trunk member if the port has not yet been administered on the BRI Trunk screen.

For example, administer the B-channels on a TN2185 circuit pack inserted in slot 01A10 as follows:

Port	B-channel 1	B-channel 2
1	01A1001	01A1017
2	01A1002	01A1018
3	01A1003	01A1019
4	01A1004	01A1020
5	01A1005	01A1021
6	01A1006	01A1022
7	01A1007	01A1023
8	01A1008	01A1024

Interactions

The **`add bri-trunk board PPCSS`** command is rejected if PPCSS identifies a TN556B circuit pack, and a port on that circuit pack has already been assigned to a station or data-module. If a TN556B circuit pack has been administered as a BRI trunk circuit pack, any port on that circuit pack is prevented from being assigned to a station or data-module.

Language Translations

Pre-translated messages are available in English, French, Italian, and Spanish to display on your system telephones. Translations for many Communication Manager messages can be assigned using the Language Translations screens. As of July 1, 2005, however, new messages are no longer added to the Language Translations screens, so these screens might not show all available Communication Manager messages.

As a preferred method for entering translations for user-defined telephone messages, Avaya recommends using the *Avaya Message Editing Tool (AMET)*. This tool is available for download from http://support.avaya.com/amet For more information, see *Avaya Message Editing Tool Job Aid*.

All button names can be assigned a user-defined name.

Note:
If **user-defined** is entered for the display language on the Station screen or Attendant Console screen, and no messages are defined on these screens, a string of asterisks appears on all display messages. For information on administering Unicode languages, see *Administering Avaya Aura™ Communication Manager*, 03-300509.

In this section, the field descriptions are listed before the screens.

Field descriptions for Language Translation pages

WARNING:
Do not use the translation pages if you have installed the file avaya_user-defined.txt.

English

This is a display-only field. It contains the English version of the message on the display.

Meaning of English term

This is a display-only field. It explains the abbreviated English message.

Translation

Enter the message you want to appear on the telephone display in place of the English message. Remember that a long message might be shortened on telephones that display fewer than 32 characters.

Figure 171: Language Translations screen — AD programming

```
change display-messages ad-programming                          Page  1 of  x
                              LANGUAGE TRANSLATIONS

     English                                   Translation

  1. Press button to program.         1. ****************************************
  2. Change program?                  2. ***********************
  3. Yes = 1 No = 2                   3. ****************
  4. Enter number:                    4. ****************************************
  5. Press # to save.                 5. ****************
  6. Number saved.                    6. ****************************************
  7. Change label?                    7. ***********************
  8. Enter label:                     8. ****************************************
  9. Press * to advance; # to save.   9. ******************************
 10. Press * to reenter; # to save.  10. ******************************
 11. Label saved. Hang up to update. 11. ****************************************
```

Figure 172: Language Translations screen — Auto-Wakeup-Dn-Dst (Page 1)

```
change display-messages auto-wakeup-dn-dst                        Page  1 of  x
                              LANGUAGE TRANSLATIONS

   1.     English:    AUTO WAKEUP - Ext:
      Translation: *******************:

   2.     English: WAKEUP ENTRY DENIED
      Translation: ************************

   3.     English: WAKEUP REQUEST CANCELED
      Translation: ****************************************

   4.     English: WAKEUP REQUEST CONFIRMED
      Translation: ****************************************

   5.     English: Wakeup Call
      Translation: **************************************

   6.     English:  Time:
      Translation: *****:
```

Figure 173: Language Translations screen — Auto-Wakeup-Dn-Dst (Page 2)

```
change display-messages auto-wakeup-dn-dst                        Page  2 of  x
                              LANGUAGE TRANSLATIONS

   7.     English:    DO NOT DIST - Ext:
      Translation: *******************:

   8.     English:     DO NOT DIST - Group:
      Translation: **********************:

   9.     English: DO NOT DIST ENTRY DENIED
      Translation: ************************

  10.     English: THANK YOU - DO NOT DIST ENTRY CONFIRMED
      Translation: ****************************************

  11.     English: THANK YOU - DO NOT DIST REQUEST CANCELED
      Translation: ****************************************
```

Figure 174: Language Translations screen — Auto-Wakeup-Dn-Dst (Page 3)

```
change display-messages auto-wakeup-dn-dst                        Page  3 of  x
                              LANGUAGE TRANSLATIONS

  12.     English: INTERVAL FULL
      Translation: *************

  13.     English: NO PERMISSION
      Translation: *************

  14.     English: SYSTEM FULL
      Translation: *************

  15.     English: TOO SOON
      Translation: *************

  16.     English: INVALID EXTENSION - TRY AGAIN
      Translation: ****************************************

  17.     English: INVALID GROUP - TRY AGAIN
      Translation: ****************************************
```

Figure 175: Language Translations screen — Button Labels (page 1)

```
change display-messages button-labels                             Page 1 of x
                              LANGUAGE TRANSLATIONS

 English                          Translation

  1. Abr Mark                  1. *************
  2. Abr Pause                 2. *************
  3. Abr Program               3. *************
  4. Abr Spec Char             4. *************
  5. Abr Stop                  5. *************
  6. Abr Suppress              6. *************
  7. AbRing                    7. *****
  8. Abr Wait                  8. *************
  9. Account                   9. *************
 10. AD                       10. **********
 11. AddBusyInd               11. *************
 12. AdmConnAlarm             12. *************
 13. AfterCall                13. **********
 14. Alert Agent              14. *************
```

Figure 176: Language Translations screen — Button Labels (page 2)

```
change display-messages button-labels                          Page 2 of x
                              LANGUAGE TRANSLATIONS

 English                          Translation

  1. Alternate FRL             1. *************
  2. ANI Request               2. *************
  3. Assist                    3. **********
  4. ASVN Halt                 4. *************
  5. AttQueueCall              5. *************
  6. AttQueueTime              6. *************
  7. Audix Record              7. *************
  8. Auto Callback             8. *************
  9. Auto Ckt Halt             9. *************
 10. AutoIC                   10. *******
 11. Auto In                  11. **********
 12. AutoWakeAlarm            12. *************
 13. Auto Wakeup              13. *************
 14. AuxWork                  14. ********
 15. Busy                     15. *****
```

Figure 177: Language Translations screen — Button Labels (page 3)

```
change display-messages button-labels                          Page 3 of x
                              LANGUAGE TRANSLATIONS

 English                          Translation

  1. Button Ring               1. *************
  2. Button View               2. *************
  3. Caller Info               3. *************
  4. CFrwd                     4. *****
  5. Call Park                 5. *************
  6. Call Pickup               6. *************
  7. Call Time                 7. *************
  8. CAS Backup                8. *************
  9. Cancel LWC                9. *************
 10. CDR1 Fail                10. *************
 11. CDR2 Fail                11. *************
 12. CFBDA                    12. *****
 13. Check In                 13. *************
 14. Check Out                14. *************
```

Figure 178: Language Translations screen — Button Labels (page 4)

```
change display-messages button-labels                          Page 4 of x
                              LANGUAGE TRANSLATIONS

 English                          Translation

  1. ClkOverride             1. ************
  2. CO Line                 2. ***********
  3. Conf Display            3. *************
  4. Consult                 4. *************
  5. Cover Msg Ret           5. *************
  6. CovrCallBack            6. *************
  7. CPN Block               7. *************
  8. CPN Unblock             8. *************
  9. Crisis Alert            9. *************
 10. Data                   10. *****
 11. Delete Msg             11. *************
 12. Dial Icom              12. **********
 13. DID Remove             13. *************
 14. DID View               14. *************
```

Figure 179: Language Translations screen — Button Labels (page 5)

```
change display-messages button-labels                           Page 5 of x
                             LANGUAGE TRANSLATIONS

 English                          Translation

  1. Directory               1. *************
  2. Dir Pickup              2. *************
  3. Disp Charges            3. *************
  4. DoNotDisturb            4. *************
  5. EC500                   5. *************
  6. Exclusion               6. *************
  7. ExtDoNotDstrb           7. *************
  8. Extend Call             8. *************
  9. Far End Mute            9. *************
 10. Flash                  10. *************
 11. FTC Alarm              11. *************
 12. Goto Cover             12. *************
 13. GrpPg                  13. *****
 14. GrpDoNotDstrb          14. *************
 15. Hunt NS                15. **********
 16. InCalID                16. ********
```

Figure 180: Language Translation screen - Button Labels (page 6)

```
change display-messages button-labels                              Page 6 of x
                              LANGUAGE TRANSLATIONS

 English                          Translation

  1. Inspect                   1. *************
  2. IntAutoAnswer             2. *************
  3. License Error             3. *************
  4. Link Fail                 4. ***********
  5. Lock LWC                  5. *************
  6. LSVN Halt                 6. *************
  7. Major Alarm               7. *************
  8. Make Call                 8. *************
  9. ManOverid                 9. ************
 10. Manual In                10. **********
 11. MCT Activate             11. *************
 12. MCT Control              12. *************
 13. Mj/Mn Alarm              13. *************
```

Figure 181: Language Translations screen — Button Labels (page 7)

```
change display-messages button-labels                             Page 7 of x
                              LANGUAGE TRANSLATIONS

 English                          Translation

  1. MM Basic                  1. *************
  2. MM Call                   2. *************
  3. MM Call Fwd               3. *************
  4. MM Data Conf              4. *************
  5. MM Mult Nbr               5. *************
  6. MM PC Audio               6. *************
  7. Msg                       7. *****
  8. Msg Retrieve              8. *************
  9. MsgW                      9. *****
 10. MsgWaitAct               10. *************
 11. MsgWaitDeact             11. *************
 12. MST Debug                12. *************
 13. Next                     13. *************
 14. Night Service            14. *************
```

Figure 182: Language Translations screen — Button Labels (page 8)

```
change display-messages button-labels                          Page 8 of x
                              LANGUAGE TRANSLATIONS

 English                                  Translation

  1. NoAnsAltr                    1. **********
  2. OffBoardAlarm                2. *************
  3. PAGE1 Alarm                  3. *************
  4. PAGE2 Alarm                  4. *************
  5. PMS Failure                  5. *************
  6. PMS Ptr Alarm                6. *************
  7. Posted MSGs                  7. *************
  8. Priority Call                8. *************
  9. QueueCall                    9. **********
 10. QueueTime                   10. **********
 11. Release                     11. *************
 12. RemBusyInd                  12. *************
 13. ResetAlert                  13. *************
 14. Ringer Off                  14. *************
 15. Ring Stat                   15. *************
```

Figure 183: Language Translations screen — Button Labels (page 9)

```
change display-messages button-labels                         Page 9 of x
                              LANGUAGE TRANSLATIONS

 English                                  Translation

  1. RSVN Halt                    1. *************
  2. SD                           2. **
  3. SendAllCalls                 3. *************
  4. Send TEG                     4. *************
  5. Service Obsrv                5. *****
  6. Sgnl                         6. *************
  7. SSVN Halt                    7. *************
  8. Start Bill                   8. *************
  9. Station Lock                 9. *************
 10. Stored Number               10. *************
 11. Store LWC                   11. *************
 12. Stroke Count                12. ************
 13. System Alarm                13. *************
 14. TermGroup                   14. **********
 15. Time/Date                   15. *************
 16. Timer                       16. *************
```

Figure 184: Language Translations screen — Button Labels (page 10)

```
change display-messages button-labels                          Page 10 of x
                              LANGUAGE TRANSLATIONS

 English                          Translation

  1. Toggle Swap  :            1. ************
  2. Trunk ID                  2. ***********
  3. Trunk Name                3. *************
  4. Trunk NS                  4. *************
  5. UUI Info                  5. *************
  6. Verify                    6. *************
  7. VIP Check In              7. *************
  8. VIP retry                 8. *************
  9. VIP Wakeup                9. *************
 10. VOA Repeat               10. *****
 11. VU Display               11. *************
 12. WhisperAct               12. **********
 13. WhisperAnbk              13. *************
 14. WhisperOff               14. *************
 15. Work Code                15. *************
```

Figure 185: Language Translations screen — Call-Identifiers (Page 1)

```
change display-messages call identifiers                        Page  1 of  x
                              LANGUAGE TRANSLATIONS

     English               Meaning of English term                 Translated
     Term                                                          Term

  1. sa           ACD Supervisor Assistance                     1: **
  2. ac           Attendant Assistance Call                     2: **
  3. tc           Attendant Control Of A Trunk Group            3: **
  4. an           Attendant No Answer                           4: **
  5. pc           Attendant Personal Call                       5: **
  6. rc           Attendant Recall Call                         6: **
  7. rt           Attendant Return Call                         7: **
  8. sc           Attendant Serial Call                         8: **
  9. co           Controlled Outward Restriction                9: **
 10. cs           Controlled Station To Station Restriction    10: **
 11. ct           Controlled Termination Restriction           11: **
 12. db           DID Find Busy Station With CO Tones          12: **
 13. da           DID Recall Go To Attendant                   13: **
 14. qf           Emergency Queue Full Redirection             14: **
 15. hc           Held Call Timed Reminder                     15: **
```

Figure 186: Language Translations screen — Call-Identifiers (Page 2)

```
change display-messages call identifiers                       Page  2 of  x
                              LANGUAGE TRANSLATIONS

     English                 Meaning of English term                Translated
     Term                                                           Term

 16. ic             Intercept                                   16: **
 17. ip             Interposition Call                          17: **
 18. ld             LDN Calls On DID Trunks                     18: **
 19. so             Service Observing                           19: **
 20. na             Unanswered Or Incomplete DID Call           20: **
 21. ACB            Automatic Callback                          21: ********
 22. callback       Callback Call                               22: ********
 23. park           Call Park                                   23: ********
 24. control        Control                                     24: ********
 25. ICOM           Intercom Call                               25: ********
 26. OTQ            Outgoing Trunk Queuing                      26: ********
 27. priority       Priority Call                               27: ********
 28. recall         Recall Call                                 28: ********
 29. return         Return Call                                 29: ********
 30. ARS            Automatic Route Selection                   30: ********
```

Figure 187: Language Translations screen — Call-Identifiers (Page 3)

```
change display-messages call identifiers                       Page  3 of  x
                              LANGUAGE TRANSLATIONS

     English                 Meaning of English term                Translated
     Term                                                           Term

 31. forward        Call Forwarding                             31: ********
 32. cover          Cover                                       32: ********
 33. DND            Do Not Disturb                              33: ********
 34. p              Call Pickup                                 34: *
 35. c              Cover All Calls                             35: *
 36. n              Night Station Service, Including No Answer  36: *
 37. B              All Calls Busy                              37: *
 38. f              Call Forwarding                             38: *
 39. b              Cover Busy                                  39: *
 40. d              Cover Don't Answer                          40: *
 41. s              Send All Calls                              41: *
 42. to             <calling party> to <called party>           42: **
 43. VDN            Vector Directory Number                     43: ***
 44. hunt           Station Hunting, Origination                44: ********
 45. h              Station Hunting, Termination                45: *
```

Figure 188: Language Translations screen — Call-Identifiers (Page 4)

```
change display-messages call identifiers                        Page  4 of  x
                              LANGUAGE TRANSLATIONS

     English         Meaning of English term                  Translated
     Term                                                     Term

 46. OPERATOR        Operator                               46: ***************
 47. EXT             Extension                              47: ***************
 48. OUTSIDE CALL    Outside Call                           48: ***************
 49. UNKNOWN NAME    Unknown Name                           49: ***************
 50. CONFERENCE      Conference                             50: ************
 51. ringing         Ringing                                51: ********
 52. busy            Busy                                   52: ********
 53. busy(I)         Busy With Intrusion Allowed            53: ********
 54. wait            Wait                                   54: *****
 55. (I)             Intrusion                              55: ***
 56. Sta             Station                                56: *********
 57. Trk             Trunk                                  57: *********
 58: offered         QSIG call offered to remote endpoint   58: ********
 59: cl              Controlled Toll Restriction            59: **
 60: vm              Call to Attendant Out of Voicemail     60: **
```

Figure 189: Language Translations screen — Date-Time (Page 1)

```
change display-messages date-time                               Page  1 of  x
                          LANGUAGE TRANSLATIONS

     English           Translation            English          Translation

  1. SUNDAY          1: ***********        11. APRIL         11: ***********
  2. MONDAY          2: ***********        12. MAY           12: ***********
  3. TUESDAY         3: ***********        13. JUNE          13: ***********
  4. WEDNESDAY       4: ***********        14. JULY          14: ***********
  5. THURSDAY        5: ***********        15. AUGUST        15: ***********
  6. FRIDAY          6: ***********        16. SEPTEMBER     16: ***********
  7. SATURDAY        7: ***********        17. OCTOBER       17: ***********
  8. JANUARY         8: ***********        18. NOVEMBER      18: ***********
  9. FEBRUARY        9: ***********        19. DECEMBER      19: ***********
 10. MARCH          10: ***********

 20.     English: SORRY, TIME UNAVAILABLE NOW
     Translation: ****************************************
```

Figure 190: Language Translations screen — Leave-Word-Calling (Page 1)

```
change display-messages leave-word-calling                     Page  1 of  x
                              LANGUAGE TRANSLATIONS

   1.     English:             MESSAGES FOR
      Translation: ************************

   2.     English: WHOSE MESSAGES? (DIAL EXTENSION NUMBER)
      Translation: ****************************************

   3.     English: END OF MESSAGES (NEXT TO REPEAT)
      Translation: ****************************************

   4.     English: MESSAGES UNAVAILABLE - TRY LATER
      Translation: ****************************************

   5.     English: MESSAGE RETRIEVAL DENIED
      Translation: ****************************************

   6.     English: MESSAGE RETRIEVAL LOCKED
      Translation: ****************************************
```

Figure 191: Language Translations screen — Leave-Word-Calling (Page 2)

```
change display-messages leave-word-calling                    Page  2 of  x
                              LANGUAGE TRANSLATIONS

   7.     English: NO MESSAGES
      Translation: ****************************************

   8.     English: IN PROGRESS
      Translation: ****************************************

   9.     English: DELETED
      Translation: ****************************************

  10.     English: GET DIAL TONE, PUSH Cover Msg Retrieval
      Translation: ****************************************

  11.     English: Message Center (AUDIX) CALL
      Translation: ****************************************

  12.     English: CANNOT BE DELETED - CALL MESSAGE CENTER
      Translation: ****************************************
```

Figure 192: Language Translations screen — Malicious-Call-Trace (Page 1)

```
change display-messages mailcious-call-trace                   Page  1 of  x
                              LANGUAGE TRANSLATIONS

   1.     English: MALICIOUS CALL TRACE REQUEST
      Translation: ****************************************

   2.     English: END OF TRACE INFORMATION
      Translation: ****************************************

  3.     English: original call redirected from:
     Translation: *****************************:

  4.     English:          voice recorder port:
     Translation: *****************************:

  5.     English:  MCT activated by:       for:
     Translation: *****************:      ****:
```

Figure 193: Language Translations screen — Malicious-Call-Trace (Page 2)

```
change display-messages mailcious-call-trace                    Page  2 of  x
                             LANGUAGE TRANSLATIONS


  6.     English:    party :                   (EXTENSION)
     Translation: ******** :       ***********************

  7.     English:    party :                (ISDN SID/CNI)
     Translation: ******** :              ****************

  8.     English:    party :                     (PORT ID)
     Translation: ******** :           *******************

  9.     English:    party :                (ISDN PORT ID)
     Translation: ******** :           *******************
```

Figure 194: Language Translations screen — Miscellaneous-Features (Page 1)

```
change display-messages miscellaneous-features                  Page  1 of x
                              LANGUAGE TRANSLATIONS

     English                                   Translation

  1. ALL MADE BUSY                         1: ***************
  2. BRIDGED                               2: ***************
  3. DENIED                                3: ***************
  4. INVALID                               4: ***************
  5. NO MEMBER                             5: ***************
  6. OUT OF SERVICE                        6: ***************
  7. RESTRICTED                            7: ***************
  8. TERMINATED                            8: ***************
  9. TRUNK SEIZED                          9: ***************
 10. VERIFIED                             10: ***************
 11. CDR OVERLOAD                         11: ***************
 12. ANSWERED BY                          12: ***************
 13. CALL FROM                            13: ***************
 14. Skills                               14: ***************
```

Figure 195: Language Translations screen — Miscellaneous Features (Page 2)

```
change display-messages miscellaneous-features                 Page  2 of x
                              LANGUAGE TRANSLATIONS
     English              Meaning of English term                Translated
     Term                                                        Term

 15. TOLL          Toll                                     15: ****
 16. FULL          Full                                     16: ****
 17. NONE          None                                     17: ****
 18. ORIG          Origination                              18: ****
 19. OTWD          Outward                                  19: ****
 20. CALL          <call> This Number                       20: ****
 21. INTL          International                            21: ****
 22. Info          Information                              22: *****
 23. p             Primary                                  23: *
 24. s             Secondary                                24: *
 25. m             Mark                                     25: *
 26. p             Pause                                    26: *
 27. s             Suppress                                 27: *
 28. w             Wait For A Specified Time                28: *
 29. W             Wait For Off-Premise Dial Tone           29: *
```

Figure 196: Language Translations screen — Miscellaneous-Features (Page 3)

```
change display-messages miscellaneous-features                  Page 3 of x

                              LANGUAGE TRANSLATIONS

  30.     English: You have adjunct messages
      Translation: ****************************************

  31.     English: Login Violation
      Translation: ****************************************

  32.     English: Barrier Code Violation
      Translation: ****************************************

  33.     English: Authorization Code Violation
      Translation: ****************************************

  34.     English: DIRECTORY - PLEASE ENTER NAME
      Translation: ****************************************

  35.     English: DIRECTORY UNAVAILABLE - TRY LATER
      Translation: ****************************************
```

Figure 197: Language Translations screen — Miscellaneous-Features (Page 4)

```
change display-messages miscellaneous-features                   Page 4 of x

                              LANGUAGE TRANSLATIONS

  36.     English: NO MATCH - TRY AGAIN
      Translation: ****************************************

  37.     English: NO NUMBER STORED
      Translation: ****************************************

  38.     English: TRY AGAIN
      Translation: ****************************************

  39.     English: Ext          in EMRG Q
      Translation: ***          **********

  40.     English:       HUNT GROUP   NOT ADMINISTERED
      Translation: ****************   *********************

  41.     English: Q-time       calls
      Translation: ******       *****
```

Figure 198: Language Translations screen — Miscellaneous-Features (Page 5)

```
change display-messages miscellaneous-features                  Page 5 of x
                             LANGUAGE TRANSLATIONS

   42.     English: Add Skill: Enter number, then # sign
       Translation: ****************************************

   43.     English: Remove Skill: Enter number, then # sign
       Translation: ****************************************

   44.     English: Enter Skill Level, then # sign
       Translation: ****************************************

   45.     English: Enter Agent LoginID
       Translation: ****************************************

   46.     English: Call Type
       Translation: ***************

   47.     English: Call Charge
       Translation: ********************
```

Figure 199: Language Translations screen — Miscellaneous-Features (Page 6)

```
change display-messages miscellaneous-features                 Page 6 of x
                               LANGUAGE TRANSLATIONS

   48.     English: Station Security Code Violation
       Translation: ***************************************
   49.     English: ENTER REASON CODE
       Translation: ***************************************
   50.     English: Whisper From
       Translation: ********************
   51.     English: Whisper To
       Translation: ********************
   52.     English: Press button to remove.
       Translation: ***************************************
   53.     English: Press # to remove.
       Translation: ********************
```

Figure 200: Language Translations screen — Miscellaneous-Features (Page 7)

```
change display-messages miscellaneous-features                    Page 7 of x
                               LANGUAGE TRANSLATIONS

   54.     English: Button removed.
       Translation: ***************************************
   55.     English: Ringer On
       Translation: ***************************************
   56.     English: Ringer Off
       Translation: ***************************************
   57.     English: Ringer Abbreviated
       Translation: ***************************************
   58.     English: Ringer Delayed
       Translation: ***************************************
   59.     English: Select a held party's line to talk.
       Translation: ***************************************
```

Figure 201: Language Translations screen — Property-Management (Page 1)

```
change display-messages property-management                      Page  1 of  x
                              LANGUAGE TRANSLATIONS

   1.     English:                     CHECK IN - Ext:
      Translation: *********************************:

   2.     English: CHECK-IN: ROOM ALREADY OCCUPIED
      Translation: ****************************************

   3.     English: CHECK IN COMPLETE
      Translation: ****************************************

   4.     English: CHECK IN FAILED
      Translation: ****************************************

   5.     English:                    CHECK OUT - Ext:
      Translation: *********************************:

   6.     English: CHECK OUT: ROOM ALREADY VACANT
      Translation: ****************************************
```

Figure 202: Language Translations screen — Property-Management (Page 2)

```
change display-messages property-management                     Page  2 of  x
                              LANGUAGE TRANSLATIONS

   7.     English: CHECK OUT FAILED
      Translation: ****************************************

   8.     English: MESSAGE NOTIFICATION FAILED
      Translation: ****************************************

   9.     English:    MESSAGE NOTIFICATION ON - Ext:
      Translation: *********************************:

  10.     English:   MESSAGE NOTIFICATION OFF - Ext:
      Translation: *********************************:
  11.     English: CHECK OUT COMPLETE: MESSAGE LAMP OFF
      Translation: ****************************************

  12.     English: CHECK OUT COMPLETE: MESSAGE LAMP ON
      Translation: ****************************************
```

Figure 203: Language Translations screen — Property-Management (Page 3)

```
change display-messages property-management                      Page  3 of  x
                              LANGUAGE TRANSLATIONS


 13.     English: MESSAGE LAMP ON
     Translation: ****************************************

 14.     English: MESSAGE LAMP OFF
     Translation: ****************************************
 15.     English: Occupied Rooms
     Translation: ****************************************
 16.     English: Enter Room Status (1-6)
     Translation: ****************************************
 17.     English: State, Enter number from 1 - 6
     Translation: ****************************************
 18.     English: DID
     Translation: *********************
```

Figure 204: Language Translations screen — Property-Management (Page 4)

```
change display-messages property-management                     Page 4 of x
                              LANGUAGE TRANSLATIONS
   19.     English: DID VIEW: EXT?
       Translation: ****************************************
   20.     English: DID=                   CHANGE?
       Translation: ****                   **********************
   21.     English: DID VIEW DONE
       Translation: ****************************************
   22.     English: NO DID AVAILABLE
       Translation: ****************************************
   23.     English: CHECK IN COMPLETE, DID=
       Translation: *********************************
```

Figure 205: Language Translations screen — Property-Management (Page 5)

```
change display-messages property-management                    Page 5 of x

                               LANGUAGE TRANSLATIONS
   24.     English: REMOVE(1), REPLACE(2)?
       Translation: ****************************************
   25.     English: DID REMOVED
       Translation: ****************************************
   26.     English: VIP CHECK IN - Ext:
       Translation: ****************************************
   27.     English: SPECIFY VID DID:
       Translation: ****************************************
   28.     English: CHECK IN COMPLETE, INVALID DID
       Translation: ****************************************
   29.     English: CHECK IN COMPLETE, DID UNAVAILABLE
       Translation: ****************************************
```

Figure 206: Language Translations screen — Property-Management (Page 6)

```
change display-messages property-management                   Page 6 of x

                               LANGUAGE TRANSLATIONS
   30.     English: DID REMOVE - Ext:
       Translation: ****************************************
   31.     English: DID CHANGED
       Translation: ****************************************
   32.     English: AUTOMATIC ASSIGN(1), SELECT(2)?
       Translation: ****************************************
   33.     English: DID UNAVAILABLE
       Translation: ****************************************
```

Figure 207: Language Translations screen — Self-Administration (page 1)

```
hange display-messages self-administration                     Page 1 of x

                              LANGUAGE TRANSLATIONS

 English                             Translation

  1. SECURITY CODE:              1. **************
  2. INCORRECT SECURITY CODE     2. ************************
  3. SELECT FEATURE              3. ************************
  4. EXTENSION:                  4. ******************
  5. OPTIONAL EXTENSION:         5. ******************
  6. TEL NUM:                    6. ********
  7. PRESS BUTTON TO PROGRAM     7. ************************
  8. BUTTON PROGRAMMED!          8. ************************
  9. INCORRECT BUTTON            9. ************************
 10. BUTTON LIMIT REACHED       10. ************************
 11. BUSY, TRY AGAIN LATER      11. ************************
 12. YOUR PHONE IS BUSY         12. ************************
 13. SECURITY CODE NEEDED       13. ************************
 14. BUTTON NOT CHANGED         14. ************************
```

Figure 208: Language Translations screen — Self-Administration (page 2)

```
change display-messages self-administration                     Page 2 of x

                              LANGUAGE TRANSLATIONS

 English     Translation    English                Translation

  1. Acct    *****          CDR Account Code       ************************
  2. AutoD   *****          Automatic Dialing      ************************
  3. CFrwd   *****          Call Forwarding        ************************
  4. CPark   *****          Call Park              ************************
  5. CPkUp   *****          Call Pickup            ************************
  6. DPkUp   *****          Directed Call Pickup   ************************
  7. GrpPg   *****          Group Paging           ************************
  8. SAC     *****          Send All Calls         ************************
  9. Swap    *****          Conf/Trans Toggle-Swap ************************
 10. WspPg   *****          Activate whisper Page  ************************
 11. WspAn   *****          Answerback for Whisper ************************
 12. WsOff   *****          Whisper Page Off       ************************
 13. Blank   *****          Blank Button           ************************
```

Figure 209: Language Translations screen — Self-Administration (page 3)

```
change display-messages self-administration                     Page 3 of x

                              LANGUAGE TRANSLATIONS

 English                          Translation

  1. Whisper Page Off 1. *****
  2. Blank Button     2. *****
  3. Done             3. *****
  4. Cont             4. **********
  5. Expl?            5. *****
  6. ShortMode?       6. *****
  7. Next             7. *****
  8. Selct            8. *****
  9. Clear            9. *****
 10. Cancel          10. **********
 11. Delete          11. *****
 12. Replace         12. *****
 13. Bksp            13. *****
```

Figure 210: Language Translations screen — View-buttons (page 1)

```
change display-messages view-buttons                            Page 1 of x
                              LANGUAGE TRANSLATIONS

 English                          Translation

  1. Analog Bridge Appearance       1. ************************
  2. Abbreviated Dial Program       2. ************************
  3. Abbrev Dial Character          3. ************************
  4. Abbreviated Dial               4. ************************
  5. Abrv/Delayed Ring Change       5. ************************
  6. Auto Circuit Assurance         6. ************************
  7. Admin Connection Alarm         7. ************************
  8. CDR Account Code               8. ************************
  9. TSA Administration Mode        9. ************************
 10. ACD After Call Work           10. ************************
 11. ACD Change Alert              11. ************************
 12. Alternate FRL                 12. ************************
 13. Supervisor Assist             13. ************************
 14. SVN Auth Code Halt            14. ************************
 15. Attendant Queue Calls         15. ************************
 16. Attendant Queue Time          16. ************************
```

Figure 211: Language Translations screen — View-buttons (page 2)

```
change display-messages view-buttons                             Page 2 of x
                              LANGUAGE TRANSLATIONS

 English                          Translation

  1. Auto Message Waiting           1. ************************
  2. Automatic Call Back            2. ************************
  3. Automatic Dialing              3. ************************
  4. Automatic Intercom             4. ************************
  5. Auto-In Work Mode              5. ************************
  6. Automatic Wakeup               6. ************************
  7. Auxiliary Work Mode            7. ************************
  8. Bridged Appearance             8. ************************
  9. Busy Indicator                 9. ************************
 10. Call Appearance               10. ************************
 11. Call Displayed Number         11. ************************
 12. Call Forwarding               12. ************************
 13. Call Park                     13. ************************
 14. Call Pickup                   14. ************************
 15. Caller Information            15. ************************
 16. CAS (Branch) Backup           16. ************************
```

Figure 212: Language Translations screen — View-buttons (page 3)

```
change display-messages view-buttons                            Page 3 of x
                              LANGUAGE TRANSLATIONS

 English                          Translation

  1. CDR 1st Printer Alarm         1. ************************
  2. CDR 2nd Printer Alarm         2. ************************
  3. Check In Hotel/Motel          3. ************************
  4. Check Out Hotel/Motel         4. ************************
  5. Call Forwarding Busy/DA       5. ************************
  6. Clocked Override              6. ************************
  7. Consult/Return                7. ************************
  8. Coverage Callback             8. ************************
  9. Cover Message Retrieve        9. ************************
 10. Data Extension               10. ************************
 11. Time of Day/Date Display     11. ************************
 12. Delete LWC Message           12. ************************
 13. Dial Intercom                13. ************************
 14. Integrated Directory         14. ************************
 15. Directed Call Pickup         15. ************************
 16. Normal/Local Mode Toggle     16. ************************
```

Figure 213: Language Translations screen — View-buttons (page 4)

```
change display-messages view-buttons                            Page 4 of x
                              LANGUAGE TRANSLATIONS

 English                          Translation

  1. Do Not Disturb                1. ************************
  2. Drop                          2. ************************
  3. Off-Board DS1 Alarm           3. ************************
  4. Manual Exclusion              4. ************************
  5. Do Not Disturb Extension      5. ************************
  6. Flash                         6. ************************
  7. Go To Cover                   7. ************************
  8. Do Not Disturb Group          8. ************************
  9. Group Paging                  9. ************************
 10. Headset On/Off               10. ************************
 11. Hunt Night Service           11. ************************
 12. Coverage Call Identify       12. ************************
 13. Inspect Call Appearance      13. ************************
 14. Internal Auto Answer         14. ************************
 15. Last Number Dialed           15. ************************
 16. Link Failure Alarm           16. ************************
```

Figure 214: Language Translations screen — View-buttons (page 5)

```
change display-messages view-buttons                              Page 5 of x
                              LANGUAGE TRANSLATIONS

 English                          Translation

  1. Login Security Violation       1. ************************
  2. Cancel Leave Word Call         2. ************************
  3. Lockout Leave Word Call        3. ************************
  4. Leave Word Call Message        4. ************************
  5. Major Hardware Alarm           5. ************************
  6. Manual Message Waiting         6. ************************
  7. Manual Override                7. ************************
  8. ACD Manual-In                  8. ************************
  9. MCT Call Trace Activate        9. ************************
 10. MCT Call Trace Control        10. ************************
 11. Major/Minor Alarm             11. ************************
 12. Message Retrieve              12. ************************
 13. Message Waiting On            13. ************************
 14. Message Waiting Off           14. ************************
 15. Next                          15. ************************
 16. Night Service Activate        16. ************************
```

Figure 215: Language Translations screen — View-buttons (page 6)

```
change display-messages view-buttons                              Page 6 of x
                              LANGUAGE TRANSLATIONS

 English                          Translation

  1. Redirect On No Answer          1. ************************
  2. Normal Display Mode            2. ************************
  3. Personal CO Line               3. ************************
  4. Property Management Fail       4. ************************
  5. Wakeup Printer Alarm           5. ************************
  6. Print Messages                 6. ************************
  7. Priority Calling               7. ************************
  8. PMS Printer Alarm              8. ************************
  9. System Printer Alarm           9. ************************
 10. Number of Queued Calls        10. ************************
 11. Oldest Queued Time            11. ************************
 12. ACD Release                   12. ************************
 13. Ringer Cut-Off                13. ************************
 14. System Reset Alert            14. ************************
 15. Remote Access Violation       15. ************************
 16. Scroll Mode                   16. ************************
```

Figure 216: Language Translations screen — View-buttons (page 7)

```
change display-messages view-buttons                                 Page 7 of x
                              LANGUAGE TRANSLATIONS

 English                          Translation

  1. Send All Calls                1. ************************
  2. TEG Send All Calls            2. ************************
  3. Service Observing             3. ************************
  4. Manual Signalling             4. ************************
  5. Station Security Call         5. ************************
  6. Stored Number Display         6. ************************
  7. Stroke Counts                 7. ************************
  8. Term Extension Group          8. ************************
  9. Timer                         9. ************************
 10. Facility Test Call Alarm     10. ************************
 11. Trunk ID                     11. ************************
 12. Trunk Name                   12. ************************
 13. Trunk Night Service          13. ************************
 14. UUI Info                     14. ************************
 15. Busy Verification            15. ************************
 16. VDN of Origin Announce       16. ************************
```

Figure 217: Language Translations screen — View-buttons (page 8)

```
change display-messages view-buttons                                 Page 8 of x
                              LANGUAGE TRANSLATIONS

 English                          Translation

  1. Vu-Stats Displays             1. ************************
  2. Activate Whisper Page         2. ************************
  3. Whisper Page Answerback       3. ************************
  4. Whisper Page Off              4. ************************
  5. Call Work Code                5. ************************
  6. Unassigned Button             6. ************************
  7. View Button                   7. ************************
  8. Call Timer                    8. ************************
  9. Add Busy Indicator            9. ************************
 10. Remove Busy Indicator        10. ************************
 11. VIP Wakeup                   11. ************************
 12. VIP Retry                    12. ************************
 13. Crisis Alert                 13. ************************
 14. DID View                     14. ************************
 15. DID Remove                   15. ************************
 16. VIP Check In                 16. ************************
```

Figure 218: Language Translations screen — View-buttons (page 9)

```
change display-messages view-buttons                            Page   9 of x
                              LANGUAGE TRANSLATIONS

 English                            Translation

  1.   Station Lock                  1. ************************
  2.   License Error                 2. ************************
  3.   Conference Display            3. ************************
  4.   Conf/Trans Toggle-Swap        4. ************************
  5.   Far End Mute                  5. ************************
  6.   Paging 1st Link Alarm         6. ************************
  7.   Paging 2nd Link Alarm         7. ************************
  8.   EC500                         8. ************************
  9.   No Hold Conference            9. ************************
 10.   Posted Messages              10. ************************
 11.   Audix Recording              11. ************************
 12.   Extend Call                  12. ************************
```

Figure 219: Language Translations screen — Vustats

```
change display-messages vustats                                Page   1 of x

                              LANGUAGE TRANSLATIONS

 English                            Translation

  1.   FORMAT                        1. *******

  2.   NOT DEFINED                   2. *****************************
  3.   DOES NOT ALLOW OR REQUIRE ID  3. *****************************

  4.   AGENT                         4. ************
  5.   SPLIT/SKILL                   5. ************
  6.   TRUNK GROUP                   6. ************
  7.   VDN                           7. ************

  8. NOT ADMINISTERED                8. *****************
  9. NOT MEASURED                    9. *****************

 10. AGENT NOT LOGGED IN            10. ****************************************
```

Figure 220: Language Translations screen — Softkey-Labels

```
change display-messages softkey-labels                  Page   1 of   x
                         LANGUAGE TRANSLATIONS
English      Translation   English      Translation   English      Translation
 1. Acct      1. *****     17. Drop     17. *****     33. RmBsy    33. *****
 2.  AD       2. *****     18. Excl     18. *****     34. RngOf    34. *****
 3. AdBsy     3. *****     19. FMute    19. *****     35.  SAC     35. *****
 4. Admin     4. *****     20. GrpPg    20. *****     36. SFunc    36. *****
 5. AutCB     5. *****     21. HFAns    21. *****     37. Spres    37. *****
 6. BtnVu     6. *****     22. IAuto    22. *****     38. Stats    38. *****
 7. CFrwd     7. *****     23. IDial    23. *****     39. Stop     39. *****
 8. CnfDs     8. *****     24. Inspt    24. *****     40. Swap     40. *****
 9. CnLWC     9. *****     25. Last     25. *****     41. Timer    41. *****
10. Cnslt    10. *****     26.  LWC     26. *****     42. TmDay    42. *****
11. Count    11. *****     27. Mark     27. *****     43. View     43. *****
12. CPark    12. *****     28. NHCnf    28. *****     44. Wait     44. *****
13. CPkUp    13. *****     29. Pause    29. *****     45. WspAn    45. *****
14. CTime    14. *****     30. PCall    30. *****     46. WspPg    46. *****
15.  Dir     15. *****     31. PoMSG    31. *****
16. DPkUp    16. *****     32. Prog     32. *****
```

In order to provide unique labeling for abbreviated dialing button types for softkey-labels, Communication Manager replaces the last two characters with digits for the 12-key 8400 and 15-key 8434D telephones.

On the softkey label Language Translation screen, the digits following the "**AD**" are derived from the button position. If the first button is an AD button, then it is **AD1** and the fifteenth button is **AD15**. All the AD buttons between 1 and 15 have the position number appended to "**AD**."

Figure 221: Language Translations screen — Time-Of-Day-Routing

```
change display-messages time-of-day-routing                    Page  1 of  x
                              LANGUAGE TRANSLATIONS

   1.     English: ENTER ACTIVATION ROUTE PLAN, DAY & TIME
      Translation: ****************************************

   2.     English: ENTER DEACTIVATION DAY AND TIME
      Translation: ****************************************

   3.     English:    OLD ROUTE PLAN:      ENTER NEW PLAN:
      Translation: *****************:   *****************:

   4.     English:    OLD ROUTE PLAN:         NEW PLAN:
      Translation: *****************:   **************:

   5.     English: ROUTE PLAN:    FOR       ACT-TIME:
      Translation: **********:   ****     **********:

   6.     English: ROUTE PLAN:    FOR     DEACT-TIME:
      Translation: **********:   ****     **********:
```

Figure 222: Language Translations Transfer-conference screen (page 1)

```
change display-messages transfer-conference                  Page 1 of x
                         LANGUAGE TRANSLATIONS

 1.     English: Transfer completed.
    Translation: ********************************

 2.     English: Call next party.
    Translation: ****************************************

 3.     English: Press conference to add party.
    Translation: ****************************************

 4.     English: ^-party conference in progress.
    Translation: ****************************************

 5.     English: Conference canceled.
    Translation: ****************************************

 6.     English: Select line ^ to cancel or another line.
    Translation: ****************************************
```

Note:
For Messages 4, 6, 12, you manually must change “~” to “^” in your user-defined language. The software is not update automatically.

Message 4

The character "^" is a place holder.

English Text	Replacement Info
^-party conference in progress	"^" is replaced with the number of parties currently on the conference call.

Message 6

The character "^" is a place holder.

English Text	Replacement Info
Select line ^ to cancel or another line.	"^" is replaced with the letter of the line that is on soft hold.

Figure 223: Language Translations Transfer-conference screen (page 2)

```
change display-messages transfer-conference          Page 2 of x
                         LANGUAGE TRANSLATIONS

 7.      English: Dial number.
     Translation: ****************************************

 8.      English: Press transfer to complete.
     Translation: ****************************************

 9.      English: Hang-up to complete transfer.
     Translation: ****************************************

10.      English: Dial number or select held party.
     Translation: ****************************************

11.      English: Select held party to conference.
     Translation: ****************************************

12.      English: Select line ^ to add party.
     Translation: ****************************************
```

Figure 224: Language Translations Transfer-conference screen (page 3)

```
change display-messages transfer-conference          Page 3 of x
                         LANGUAGE TRANSLATIONS

13.     English: Select alerting line to answer call.
    Translation: ****************************************

14.     English: Transfer canceled.
    Translation: ****************************************

15.     English: Connecting to ^.
    Translation: ****************************************

16.     English: Called party ^is busy.
    Translation: ****************************************

17.     English: Invalid number dialed -
    Translation: ***************************

18.     English: Party ^ is not available.
    Translation: ***************************
```

Message 15, 16, 18

The character "^" is a place holder.

English Text	Replacement Info
Select line ^ to add party.	"^" is replaced with the letter of the line that is on soft hold.

Figure 225: Language Translations Transfer- Conference screen (page 4)

```
change display-messages transfer-conference                   Page 4 of x
LANGUAGE TRANSLATIONS
19. English: Mute
Translation: ****
20. English: ^-party conference:
Translation: *********************************
```

Message 12

The character "^" is a place holder.

English Text	Replacement Info
Select line ^ to add party.	"^" is replaced with the letter of the line that is on soft hold.

Figure 226: Language Translations screen — VuStats

```
change display-messages vustats                             Page  1 of  x
                              LANGUAGE TRANSLATIONS

        English                          Translations

    1.   FORMAT                          1.   *******
    2.   NOT DEFINED                     2.   *******
    3.   DOES NOT ALLOW OR REQUIRE ID    3.   *******

    4.   AGENT                           4.   *****************************
    5.   SPLIT/SKILL                     5.   *****************************
    6.   TRUNK GROUP                     6.   *****************************
    7.   VND                             7.   *****************************

    8.   NOT ADMINISTERED                8.   *****************
    9.   NOT MEASURED                    9.   *****************

   10.   AGENT NOT LOGGED IN            10.    *************************
```

Listed Directory Numbers

Allows Direct Inward Dialing (DID) numbers to be treated as public Listed Directory Numbers (LDNs). When one of these numbers is direct inward dialed, the calling party is routed to the attendant. The attendant display indicates a Listed Directory Number call and the name associated with the dialed extension.

The number of Listed Directory Numbers that can be assigned varies depending on system configuration. See the *Avaya Aura™ Communication Manager Hardware Description and Reference*, 555-245-207, for maximum values.

Field descriptions for page 1

Figure 227: Listed Directory Numbers screen

```
change listed-directory-number                                  Page   1 of  x
                         LISTED DIRECTORY NUMBERS

                          Night Destination:

         Ext     Name                        TN
      1:
      2:                                     1
      3:                                     1
      4:                                     1
      5:                                     1
      6:                                     1
      7:                                     1
      8:                                     1
```

Ext

Valid entries	Usage
0 to **9**	Enter any valid extension number.

Name

Valid entries	Usage
Up to 27 alphanumeric characters	Enter a name used to identify the Listed Directory Number

Night Destination

Enter the valid assigned extension number that receives calls to these numbers when Night Service is active.

Valid entries	Usage
0 to **9**	For DEFINITY CSI, S87XX Series IP-PNC. Enter a night service extension, a recorded announcement extension, a Vector Directory Number, an individual attendant extension, or a hunt group extension.

TN

Valid entries	Usage
1 to **100**	Enter the Tenant Partition number.

Local Survivable Processor

See **Survivable Processor**.

Locations

Use the Locations screen to provide daylight savings time displays to users, to set the area code for each location, and to administer different location information for each location. If the Multiple Locations feature is enabled, you can administer up to 250 location specifications, depending on the configuration of the server that is running Communication Manager. Otherwise, information for Location No.1 applies to all your locations.

Field descriptions for page 1

Figure 228: Locations screen

```
display locations                                                Page 1 of x
                                 LOCATIONS

               ARS Prefix 1 Required For 10-Digit NANP Calls? y

Loc  Name           Timezone Rule   NPA  ARS  Atd Loc  Disp Prefix     Proxy Sel
 No                   Offset             FAC  FAC Parm Parm              Rte Pat
 1:  _______________ _ __:__   __   ___  ____  __  __   __  ___________   ___
 2:  _______________ _ __:__   __   ___  ____  __  __   __  ___________   ___
 3:  _______________ _ __:__   __   ___  ____  __  __   __  ___________   ___
```

ARS FAC

This field is controlled by the **Multiple Locations** field on the System Parameters Customer-Options (Optional Features) screen (use the **`system-parameters customer-options`** command). Administration of this field must follow the same rules that exist for administering an ARS code on the Feature Access Code (FAC) screen.

Valid entries	Usage
0 to **9**	Any valid FAC format is acceptable, up to four digits. Characters * or # are permitted, but only in the first position. Many locations are expected to share the same access code.

ARS Prefix 1 Required for 10-Digit NANP Calls?

Valid entries	Usage
y/n	Enter **y** when a 1 must be dialed before all 10-digit NANP calls.

Attd FAC

The **Attd FAC** field is controlled by the **Multiple Locations** field on the System Parameters Customer-Options (Optional Features) screen (use the **`system-parameters customer-options`** command).

A user cannot administer an Attendant FAC unless an Attendant Access Code has first been administered on either the Dial Plan Analysis Table screen or the Feature Access Code (FAC) screen.

Note:
Within a dial plan, **FAC/DAC** codes and extensions cannot both start with the same first digits.Either the **FAC/DAC** entries or the block of extensions must be changed to have a different first digit.

Valid entries	Usage
0 to **9**	.Values up to two digits are permitted. Characters ***** or **#** are not permitted. Many locations are expected to share the same access code.

Disp Parm

This field is an index to the corresponding location on the **Display Parameters** screen. It shows the display parameters for the location.

Loc Parm

This field is an index to the corresponding Location Parameters n screens for a specific location. If Multinational Locations is activated, and you enter information into any other field on a location row, you must make an entry in the **Loc. Parms** field. If you don't, an error message displays, and your IP telephones might not be usable.

Valid entries	Usage
1 to **25** or blank	Enter the number of the corresponding Location Parameter set for this location. Default is blank.

Name

Identifies the server running Communication Manager associated with each location number.

Valid entries	Usage
up to 15 alphanumeric characters	A name you use for the location. Names are easier to remember than location numbers. **NOTE:** Avaya BRI stations support ASCII characters only. They do not support non-ASCII characters, such as Eurofont or Kanafont. Therefore, if you use non-ASCII characters in any Communication Manager **Name** field, such characters do not display correctly on a BRI station.

NPA

Valid entries	Usage
0 to **9**	Enter the 3-digit numbering plan area code for each location.

Prefix

This field is used for prepending the leading digits for Uniform Dial Plan Table screen entries for calls that have, in the Insert Digits field, an `Lx` value, where `x` is the number of leading digits of the Prefix digits to prepend for the location of an originating call. This field is controlled by the **Multiple Locations** field on the System Parameters Customer-Options (Optional Features) screen (use the `system-parameters customer-options` command).

Valid entries	Usage
0 to **9**	Values from one to five digits (0 to 99999) are permitted.

Proxy Sel Rte Pat

The **Proxy Selection Route Pattern** field identifies the routing pattern that is used to get to the proxy server. This is the route pattern assigned on the **Route Pattern** screen.

Valid entries	Usage
1 to **999** or blank	Enter the number of the routing pattern to be used to get to the proxy server.

Rule

This field must be filled in for each administered location.

Valid entries	Usage
0	No Daylight Savings
1 to **15** or blank	Specifies the number for each Daylight-Savings Rule (set up on the Daylight Savings Rule screen) that is applied to this location.

Timezone Offset

Timezone Offset is actually 3 fields (+/-, hour, and minute) that specify how much time each location differs from the system time. This field must be completed for each administered location. Use +00:00 for the time zone offset for a single location Avaya S8XXX Server.

Valid entries	Usage
+	Shows that the time set on this location is a certain amount of time ahead of the system time.
-	Shows that the time set on this location is a certain amount of time behind the system time.

Valid entries	Usage
0 to **23**	Shows the number of hours difference between this location and system time.

Valid entries	Usage
0 to **59**	Shows the number of minutes difference between this location and system time.

Location Parameters

The Location Parameters screen allows you to set or change certain administrable characteristics that determine part of a location's behavior. These include recall timing intervals and loss plans for 2-party and conference calls.

Multiple instances of the Location Parameters screen are accessible if the **Multiple Locations** field on the System Parameters Customer-Options (Optional Features) screen is set to **y**. If the **Multinational Locations** field on the System Parameters Customer-Options (Optional Features) screen is set to **y**, Location Parameters 2-25 contain the same fields as for Location Parameters 1 (see Figure 229). If the **Multinational Locations** field on the **System Parameters Customer-Options (Optional Features)** screen is set to **n**, the system does not display the following fields for Location Parameters 1:

- **Tone Generation Plan**
- **DCP Terminal-parameters Plan**
- **Country Code for CDR**

Field descriptions for page 1

Figure 229: Location Parameters screen

```
change location-parameters 1                                Page   1 of  x

                    LOCATION PARAMETERS 1

               Tone Generation Plan: 1
             Analog Ringing Cadence: 1        International Access Code:
           Analog Line Transmission: 1         Local E.164 Country Code:
       DCP Terminal-parameters Plan: 1
               Country code for CDR: 1
                    Companding Mode: Mu-law

RECALL TIMING
                 Flashhook Interval? _               Upper Bound (msec): 1000
           Disconnect Timing (msec): 150             Lower Bound (msec): 200


            Forward Disconnect Timer (msec): 600
                  MF Interdigit Timer (sec): 10
Outgoing Shuttle Exchange Cycle Timer (sec): 4
```

Analog Ringing Cadence

The country code identifies the ringing cadence to be used by analog telephones in the system

Valid entries	Usage
1 to **25**	See the Country code table at the beginning of the System-Parameters Country-Options screen description. For more information, see Audible Ringing Patterns in *Avaya Aura™ Communication Manager Hardware Description and Reference,* 555-245-207. **Note:** This field must be set to **1** (US) in order for the **Message Waiting Indicator** field on the Station screen to be set to **neon**.

Analog Line Transmission

The country code identifies the transmission and signaling parameters.

Valid entries	Usage
1 to **25**	See the Country code table at the beginning of the System-Parameters Country-Options screen description.

Companding Mode

Identifies the companding algorithm to be used by system hardware.

Valid entries	Usage
A-Law	Generally used outside the U.S.
Mu-law	Generally used in the U.S.

Country code for CDR

Appears only when the Multinational Locations feature is enabled in the license file.

Valid entries	Usage
1 to **999**	The value in this field corresponds to the country code to be used for Call Detail Recording information for a location. Default is **1**. For a list of country codes, see the Country code table on page 886.

DCP Terminal-parameters Plan

Appears only when the Multinational Locations feature is enabled in the license file. The value in this field corresponds to the DCP terminal transmission parameters administered for location *n* on the Terminal Parameters n screens.

Valid entries	Usage
1 to **25**	Enter terminal-parameters plan number 1 to 25. Default is **1**.

International Access Code

Valid entries	Usage
up to 5 digits (**0** to **9**), or blank	Enter up to 5 digits for the International Access Code. Default is blank.

Local E.164 Country Code

Valid entries	Usage
up to 3 digits (**0** to **9**), or blank	Enter up to 3 digits for the E.164 Country Code. Default is blank. For a list of country codes, see the International Telecommunications Union "List of ITU-T Recommendation E.164 Assigned Country Codes .

Tone Generation Plan

Appears only when the Multinational Locations feature is enabled in the license file. The value in this field corresponds to the tone generation characteristics administered for location *n* on the Tone Generation *n* screens.

Valid entries	Usage
1 to **25**	Enter tone-generation plan number 1 to 25. Default is **1**.

RECALL TIMING

Disconnect Timing (msec)

Appears when the **Flashhook Interval** field is **n**.

Valid entries	Usage
80 to **1250** (in increments of 10).	An on-hook that lasts for a period of time less than this value is ignored; greater than or equal to this value is regarded as a disconnect.

Flashhook Interval

Valid entries	Usage
y	Enter **y** to indicate that a flashhook interval (recall window) is required. If a **y** is entered, **Upper Bound** and **Lower Bound** appear.
n	If **n** is entered, **Disconnect Timing** appears.

Forward Disconnect Timer (msec)

Valid entries	Usage
25 to **1500** (in increments of 25).	Specify the duration of a momentary disconnect sent by the server/switch to an analog station user when that user is the last party still off-hook on a call.

Lower Bound (msec)

The lower bound of the station-to-switch recall signal timing interval in milliseconds. Specifies the lower bound of the flashhook interval. Appears when the **Flashhook Interval** field is **y**.

Valid entries	Usage
80 to **1250** (in increments of 10).	Specify the lower bound of the flashhook interval.

MF Interdigit Timer (sec)

Applies only to multifrequency signaling trunks. Specify the maximum number of seconds Communication Manager waits for the first forward signal (digit) to arrive, and for subsequent digits to arrive. Intercept returns to the calling party if this timer expires.

Valid entries	Usage
1 to **255**	This number must be less than the number of seconds entered in the short interdigit timer.

Outgoing Shuttle Exchange Cycle Timer (sec)

Appears when the **Incoming Call Type** field is **group-ii-mfc** or **non-group-ii-mfc** and the **Outgoing Call Type** field is **group-ii-mfc** or **none** on the Multifrequency-Signaling-Related System Parameters screen. This field applies only to multifrequency signaling calls made from Communication Manager.

Valid entries	Usage
1 to **25**	Enter the number of seconds to time an exchange cycle (starts when the far end requests a digit until Communication Manager sends the requested digit).

Upper Bound (msec)

Specifies the upper bound of the flashhook interval. Specifies the upper bound of the station-to-switch recall signal timing interval in milliseconds. Appears when the **Flashhook Interval** field is **y**.

Valid entries	Usage
80 to **1250** (in increments of 10).	A flash of 50 msec to 130 msec is always acceptable from a 2500-type set regardless of the setting of the Upper and Lower bounds and is treated as the digit one.

Field descriptions for page 2

Figure 230: Location Parameters screen - page 2

```
display location-parameters 1                                   Page   2 of x


 LOSS PLANS

             Inter-location Loss Group: 18

                     2 Party Loss Plan: 1              Customize? n

                        Tone Loss Plan: 1              Customize? n

 End-to-End total loss (dB) in a n-party conference:
   3: 15   4: 15   5: 15   6: 15                       Customize? n
```

LOSS PLANS

2-Party Loss Plan/Tone Loss Plan

Provides the default values for digital loss plan and for n-party conference loss.

Valid entries	Usage
1 to **25**	See the Country code table at the beginning of the **System Parameters Country-Options** screen description. Note that different codes might have similar plans.

Customize

This field appears when the **Digital Loss Plan Modification** field is **y** on the System Parameters Customer-Options (Optional Features) screen. This setting is controlled by your license file. It enables customization on the corresponding loss plan table. For the **End-to-End total loss (dB) in a n-party conference** field, when **Customize** is set to **y** (yes), the fields can be changed by the administrator. When set to **n**, the **End-to-End total loss (dB) in a n-party conference** fields are reset to the values that they would have had under the **2 Party Loss Plan** administered on page 3 of this screen. They also become display only.

Valid entries	Usage
y/n	Enables customization on the corresponding loss plan table.

End-to-End total loss (dB) in a n-party conference

Provides total loss for a conference call with the designated number of parties.

Note:
The End-to-End total loss for multi-party conference calls that is administered on this screen is not always applied to a specific call. For more information on how loss is applied to a multi-party conference call, see the Loss Plans feature description in the *Avaya Aura™ Communication Manager Feature Description and Implementation*, 555-245-205.

Valid entries	Usage
0 to **99**	The higher the number listed for a call with a fixed display number of parties, the more loss Communication Manager adds into a conference call with that number of parties; therefore, the conference call is quieter.

Inter-location Loss Group

Appears only when the Multinational Locations feature is enabled in the license file. When inserting loss for a call, the server treats parties on the call who are in separate locations as if the location with the most parties were connected by an equal number of IP tie trunks as there are parties at other locations. The **Inter-location Loss Group** field specifies the digital loss group number that is used by these "virtual" IP tie trunks.

Valid entries	Usage
1 to **19**	Enter the digital loss group number to use on inter-location calls involving this location. Default is **18**.

Field descriptions for page 3

Figure 231: 2 Party Loss Plan screen

```
change location-parameters                                          Page 3 of x


                                2 PARTY LOSS PLAN

        TO:   1   2   3   4   5   6   7   8   9  10  11  12  13  14  15  16  17  18  19
         1:   0   0   0   0   3   0   0   0   0   3   0   6   6   6   0   3   3   0   0
         2:   0   0   0   0   0   3   3   2   2   3   0   6   6   6   2   3   3   0   0
         3:   0   0   0   0   0   3   3   3   2   3   0   6   6   6   0   3   3   6   6
         4:  15   0   0   0   6   0   0   0   0   3   0   6   6   6   0   3   3   0   0
         5:   0  -3  -3   0   0  -3  -3  -3  -3   0  -3   3   0   0  -3   3   3   0   0
         6:   0   3   3   0   0   6   8   6   5   5   5   9   9   9   5   3   3   0   0
    F    7:   0   3   3   0   0   8   8   6   5   5   5   9   9   9   5   3   3   0   0
    R    8:   0   3   3   0   0   6   6   6   3   5   3   9   6   6   3   3   3   0   0
    O    9:   0   2   2   0   0   5   5   3   0   0   2   3   3   3   9   3   3   0   0
    M   10:   3   3   3   3   3   5   5   5   0   0   3   3   3   3   3   3   3   3   3
        11:   0   0   0   0   0   5   5   3   2   3   0   6   6   3   0   3   3   0   0
        12:   0   0   0   0   0   3   3   3  -3  -3   0   0   0   0   0   3   3   6   6
        13:   0   0   0   0   0   3   3   3  -3  -3   0   0   0   0   0   3   3   6   6
        14:   0   0   0   0   0   3   3   3  -3  -3  -3   0   0   0   0   3   3   6   6
        15:   0   2   0   0   0   5   5   3   0   3   0   6   6   6   0   3   3   0   0
        16:   3   3   3   3   3   3   3   3   3   3   3   3   3   3   3   3   3   3   3
        17:   3   3   3   3   3   3   3   3   3   3   3   3   3   3   3   3   3   3   3
        18:   0   0   0   0   3   0   0   0   0   3   0   6   6   6   0   3   3   0   0
        19:   0   0   0   0   0   3   3   2   2   3   0   6   6   6   2   3   3   0   0
```

FROM / TO

Display-only fields that identify the variable digital loss values.

Valid entries	Usage
-3 through **15**	An unsigned number is a decibel loss, while a number preceded with a minus sign is a decibel gain.

Field descriptions for page 4

Figure 232: Tone Loss Plan screen

```
change location-parameters                                    Page 4 of x


                              TONE LOSS PLAN

                                TO
              1   2   3   4   5   6   7   8   9  10  11  12  13  14  15  16  17  18  19
       Dial:  0   3   3   0   0   6   6   6   5   0   6   5   5   5   5   0   0   0   0
    Confirm:  0   0   0   0   0   0   0   0   0   0   0   0   0   0   0   0   0   0   0
    Reorder:  0   0   0   0   0   0   0   0   0   0   0   0   0   0   0   0   0   0   0
       Busy:  0   0   0   0   0   0   0   0   0   0   0   0   0   0   0   0   0   0   0
    Ringing:  0   0   0   0   0   0   0   0   0   0   0   0   0   0   0   0   0   0   0
  Spec Ring:  0   0   0   0   0   0   0   0   0   0   0   0   0   0   0   0   0   0   0
  Intercept:  0   0   0   0   0   0   0   0   0   0   0   0   0   0   0   0   0   0   0
    Waiting:  0   0   0   0   0   0   0   0   0   0   0   0   0   0   0   0   0   0   0
     Verify:  0   0   0   0   0   0   0   0   0   0   0   0   0   0   0   0   0   0   0
    Intrude:  0   0   0   0   0   0   0   0   0   0   0   0   0   0   0   0   0   0   0
        Zip:  3  -3  -3   3   3  -3  -3  -3  -3  -3   3  -3  -3  -3  -3   0   0  -3  -3
      Music:  0   3   3   0   0   6   6   6   3   0   6   3   3   3   3   0   0   0   0
```

FROM / TO

Display-only fields that identify the variable digital tone values.

Valid entries	Usage
-3 through **15**	An unsigned number is a decibel loss, while a number preceded with a minus sign is a decibel gain.

Login Administration

Beginning with Communication Manager 4.0, there is no longer a Login Administration screen. For details on screens used for login administration, see *Maintenance Commands for Avaya Aura™ Communication Manager, Media Gateways and Servers*, 03-300431, and AAA Services in *Avaya Aura™ Communication Manager Feature Description and Implementation*, 555-245-205.

Logging Levels

Use the Logging Levels screen to administer logging of SAT activities. You can specify that commands associated with specific actions shown on this screen is logged by the system. The amount of detail to be logged is the same for all enabled actions and is specified by the **Log Data Values** field on page 1 of this screen.

Note:
The defaults on this screen provide the same amount and type of logging as in Communication Manager releases prior to 4.0.

Field descriptions for page 1

Figure 233: Logging Levels screen - page 1

```
change logging-levels                                         Page   1 of   x

                               LOGGING LEVELS

Enable Command Logging? y
       Log Data Values: none

When enabled, log commands associated with the following values:

           add? y                       export? y                   refresh? y
       busyout? y                          get? n                   release? y
campon-busyout? y                           go? y                    remove? y
        cancel? n                       import? y                     reset? y
        change? y                         list? n                      save? y
         clear? y                         mark? n                       set? y
       disable? y                      monitor? y                    status? n
       display? n                      netstat? n                      test? y
     duplicate? y                       notify? n                traceroute? n
        enable? y                         ping? n                    upload? n
         erase? y                      recycle? y
```

Enable Command Logging

Valid entries	Usage
y(es)	SAT activity is logged based on the actions selected on the remainder of the Logging Level screen.
n(o)	SAT activity is not logged.

Log Data Values

Valid entries	Usage
none	Only the object, the qualifier, and the command action are logged.
new	The new value of any field is logged. The old value is not logged.
both	Both the prior field value and the field value after the change are logged.

Field descriptions for page 2

Figure 234: Logging Levels screen - page 2

```
change logging-levels                                         Page   2 of   x

                               LOGGING LEVELS

    Log All Submission Failures? y
        Log PMS/AD Transactions? n
Log IP Registrations and events? n
   Log CTA/PSA/TTI Transactions? y
```

Log All Submission Failures

When set to **y**, an event is logged when Communication Manager rejects a form submission for any reason, such as an invalid entry in a field or a missing value. When the field is set to **n**, a submission failure is not logged. Form submission failures due to a security violation are always logged and are not affected by this field.

Valid entries	Usage
y/n	Enter **y** to record submission failures on the history log.

Log CTA/PSA/TTI Transactions in History Log

Appears when the **Terminal Translation Initialization (TTI) Enabled** field is **y**. Use this field to record when extensions and physical telephones move between ports without additional administration from the administrator of Communication Manager.

Valid entries	Usage
y	Enter **y** if you want the system to record Customer Telephone Activation (CTA), Personal Station Activation (PSA), and TTI transactions in the system history log.
n	Enter **n** if you do not want the system to record Customer Telephone Activation (CTA), Personal Station Activation (PSA), and TTI transactions in the system history log.

Log IP Registrations and events

Allows the logging of IP registrations in the history log.

Valid entries	Usage
y/n	Enter **y** to record IP registrations on the history log.

Log PMS/AD Transactions

Valid entries	Usage
y/n	Enter **y** to record Property Management System (PMS) and Abbreviated Dialing (AD) events to the log.

Loudspeaker Paging

The Loudspeaker Paging screen administers voice paging, deluxe voice paging, and chime paging.

Note:
To set up paging on a H.248 gateway, connect the paging system to a port on an MM711 and administer the port as an analog station on the Station screen. No entries on the Loudspeaker Paging screen are required.

Field descriptions for page 1

Figure 235: Loudspeaker Paging screen

```
change paging loudspeaker                                         Page   1 of   x
                             LOUDSPEAKER PAGING

                                   CDR? _
            Voice Paging Timeout (sec): ___
           Code Calling Playing Cycles: _

PAGING PORT ASSIGNMENTS
                 Voice Paging      Code Calling
 Zone  Port      TAC  COR  TN      TAC  COR  TN      Location:
   1: _______    ____ __   _       ____ __   _       ___________________________
   2: _______    ____ __   _       ____ __   _       ___________________________
   3: _______    ____ __   _       ____ __   _       ___________________________
   4: _______    ____ __   _       ____ __   _       ___________________________
   5: _______    ____ __   _       ____ __   _       ___________________________
   6: _______    ____ __   _       ____ __   _       ___________________________
   7: _______    ____ __   _       ____ __   _       ___________________________
   8: _______    ____ __   _       ____ __   _       ___________________________
   9: _______    ____ __   _       ____ __   _       ___________________________
 ALL: _______    ____ __   _       ____ __   _
```

CDR

This field determines whether CDR data is collected for the paging ports.

Valid entries	Usage
y/n	Enter **y** if you want the server running Communication Manager to collect CDR data on the paging ports.

Code Calling — COR

This field assigns a Class of Restriction to a paging zone.

Valid entries	Usage
0 to **995**	You can assign different classes of restriction to different zones.
blank	Leave this field blank for unused paging zones.

Code Calling Playing Cycles

This field sets the number of times a chime code will play when a user makes a chime page. To determine the best setting, consider who your code calling users are and whether they are likely to hear the code chime the first time.

Valid entries	Usage
1 to **3**	Enter the number of times you want the chime code to play when a user makes a page.
blank	The field cannot be blank when you administer chime paging (code calling).

Code Calling — TAC

This field assigns a Trunk Access Code (TAC) to a paging zone. Users dial this code to make a page to a specific zone. One TAC must be assigned to each zone you want to use. Two zones cannot have the same TAC. If you enter a TAC in the **ALL** field, users can activate speakers in all the zones by dialing that code.

Valid entries	Usage
1 to 4 digits	Enter a Trunk Access Code (TAC) allowed by your dial plan.
*	Can be used as first digit.
#	Can be used as first digit.
blank	Leave this field blank for unused paging zones.

Code Calling — TN

<table>
<tr><th>Valid entries</th><th>Usage</th></tr>
<tr><td>1 to 20 (DEFINITY CSI)</td><td rowspan="2">If your system uses Tenant Partitioning, you can use this field to assign a paging zone to a specific tenant partition.</td></tr>
<tr><td>1 to 100 (S87XX/S8300 Servers)</td></tr>
</table>

Location

Valid entries	Usage
1 to 27 characters	Assign a descriptive name for the physical location corresponding to each zone. Typical entries might be "conference room A," "warehouse," or "storeroom."

Port

This field assigns a port on an auxiliary trunk circuit pack to a paging zone. Enter the necessary characters.

Valid entries	Usage
01 to **03** (DEFINITY CSI) or **1** to **64** (S87XX/S8300 Servers)	First and second characters are the cabinet number
A to **E**	Third character is the carrier
0 to **20**	Fourth and fifth character are the slot number
01 to **04** (Analog TIE trunks) **01** to **31**	Six and seventh characters are the circuit number
1 to **80** (DEFINITY CSI) or **1** to **250** (S87XX/S8300 Servers)	Gateway
V1 to **V9**	Module
01 to **31**	Circuit
blank	Leave this field blank for unused paging zones.

Voice Paging — COR

This field assigns a Class of Restriction to a paging zone.

Valid entries	Usage
0 to **995**	You can assign different classes of restriction to different zones.
blank	Leave this field blank for unused paging zones.

Voice Paging — TAC

This field assigns a Trunk Access Code (TAC) to a paging zone. Users dial this code to make a page to a specific zone. One TAC must be assigned to each zone you want to use. Two zones cannot have the same TAC. If you enter a TAC in the **ALL** field, users can activate speakers in all the zones by dialing that code.

Valid entries	Usage
1 to 4 digits	Enter a Trunk Access Code (TAC) allowed by your dial plan.
*****	Can be used as first digit.
#	Can be used as first digit.
blank	Leave this field blank for unused paging zones.

Voice Paging Timeout (sec)

This field limits the duration of voice pages. When this interval ends, calls are disconnected. To determine the best setting, time the typical pages you expect to broadcast and then add another 4 to 5 seconds.

Valid entries	Usage
10 to **600** seconds	Enter the maximum number of seconds you want any page to last.
blank	The field cannot be blank when you administer voice paging.

Note:
To use a port that has no hardware associated with it, place an **x** in this field.

Voice Paging — TN

Valid entries	Usage
1 to **20** (DEFINITY CSI)	If your system uses Tenant Partitioning, you can use this field to assign a paging zone to a specific tenant partition.
1 to **100** (S8300/S87XX Servers)	

Related topics

See Loudspeaker Paging in *Avaya Aura™ Communication Manager Feature Description and Implementation,* 555-245-205, for more information.

Maintenance-Related System Parameters

This screen is described in *Maintenance Commands for Avaya Communication Manager, Media Gateways and Servers*, 03-300431.

MCT Group Extensions

See **Extensions Administered to have an MCT-Control Button**.

Media-Gateway

This screen is described in *Maintenance Commands for Avaya Communication Manager, Media Gateways and Servers*, 03-300431.

Mode Code Related System Parameters

This screen establishes parameters associated with the Mode Code Voice Mail System Interface.

Note:
You can only administer this screen if the **Mode Code Interface** field on the Feature-Related System Parameters screen is set to **y**.

Field descriptions for page 1

Figure 236: Mode Code Related System Parameters screen

```
change system-parameters mode-code                              Page   1 of x

                   MODE CODE RELATED SYSTEM PARAMETERS

MODE CODES  (FROM SWITCH TO VMS)
      Direct Inside Access: __
Direct Dial Access - Trunk: __
         Internal Coverage: __
         External Coverage: __

           Refresh MW Lamp: __

     System In Day Service: __
   System In Night Service: __

                     OTHER RELATED PARAMETERS
DTMF Duration On (msec): __ Off (msec): __ Sending Delay (msec):__
 VMS Hunt Group Extension : ____
Remote VMS Extensions - First:     Second:
```

MODE CODES (FROM SWITCH TO VMS)

Direct Dial Access - Trunk

This value defines a mode code that the Avaya S8XXX Server sends when an external caller dials the VMS access number.

Valid entries	Usage
0 to **9**, **#**, *****, **#00**	Up to six digits that can include these characters.

Direct Inside Access

This value defines a mode code that the Avaya S8XXX Server sends when a caller at an internal extension dials the Voice Mail System (VMS) access number.

Valid entries	Usage
0 to **9**, **#**, *****, **#00**	Up to six digits that can include these characters.

External Coverage

This value defines a mode code that the Avaya S8XXX Server sends when an external caller tries to reach a user at another extension and the call goes to the user's voice-mail coverage.

Valid entries	Usage
0 to **9**, **#**, *****, **#00**	Up to six digits that can include these characters.

Internal Coverage

This value defines a mode code that Communication Manager sends when an internal caller tries to reach a user at another extension and the call goes to the user's voice mail coverage.

Valid entries	Usage
0 to **9**, **#**, *****, **#00**	Up to six digits that can include these characters.

Refresh MW Lamp

This value defines a mode code that Communication Manager sends during a system level 3 or higher reset that requests the VMS to refresh the Message Waiting (MW) lamps.

Valid entries	Usage
0 to **9**, **#**, *****, **#00**	Up to six digits that can include these characters.

System In Day Service

This value indicates to the VMS that the Communication Manager has changed from Night to Day Service.

Valid entries	Usage
0 to **9**, **#**, *****, **#11**	Up to six digits that can include these characters.

System In Night Service

This value indicates to the VMS that the Communication Manager has changed from Day to Night Service.

Valid entries	Usage
0 to **9**, **#**, *****, **#12**	Up to six digits that can include these characters.

OTHER RELATED PARAMETERS

DTMF Duration On

Valid entries	Usage
Between **75** and **500** in multiples of 25	Define in milliseconds the length of mode code digits sent to the VMS. This field cannot be blank.

Off

Valid entries	Usage
Between **75** and **200** in multiples of 25	Define in milliseconds the pause between mode code digits as they are sent to the VMS. This field cannot be blank.

Remote VMS Extensions - First

You can administer this field if the **Mode Code for Centralized Voice Mail** field on the System Parameters Customer-Options (Optional Features) screen is set to **y**. Specifies the first remote UDP VMS hunt group extension.

Valid entries	Usage
Remote assigned hunt group extension	Enter the first UDP VMS hunt group extension.

Remote VMS Extensions - Second

You can administer this field if the **Mode Code for Centralized Voice Mail** field on the System Parameters Customer-Options (Optional Features) screen is set to **y**. Specifies the second remote UDP VMS hunt group extension.

Valid entries	Usage
Remote assigned hunt group extension	Enter the second UDP VMS hunt group extension. This extension cannot be the same as the first Remote VMS Extension.

Sending Delay

Valid entries	Usage
75 to **1000** in multiples of 25	Define in milliseconds the delay between the time answer supervision is received from the VMS and the time the first mode code digit is sent. This field cannot be blank.

VMS Hunt Group Extension

A check is made to verify that a valid hunt group extension is entered, but a check is not made to verify that the hunt group members are VMI extensions.

Valid entries	Usage
Valid extension.	Enter the extension of a hunt group containing VMI extensions.

Modem Pool Group

There are two types of conversion resources for Modem Pooling. The first type, an *integrated conversion resource*, is a circuit pack that emulates a Trunk Data Module connected to a 212A-type modem. Two conversion resources are on each circuit pack.

The second type, a *combined conversion resource*, is a separate Trunk Data Module and modem administered as a unit. The Trunk Data Module component of the conversion resource can be either a Modular Trunk Data Module (MTDM) or 7400A Data Module and connects to a digital port using Digital Communications Protocol (DCP); the modem connects to an analog port.

Field descriptions for page 1

Figure 237: Modem Pool Group screen — if Group Type is integrated

```
change modem-pool num                                        Page 1 of x
                                   MODEM POOL GROUP
                    Group Number: 1                  Group Type: integrated
Receiver Responds to Remote Loop? n             Hold Time (min): 5
           Send Space Disconnect? y     Receive Space Disconnect? y
                    CF-CB Common? y   Loss of Carrier Disconnect? y

Speed: LOW/300/1200      Duplex: full     Synchronization: a/sync

CIRCUIT PACK ASSIGNMENTS
               Circuit Pack   Circuit Pack
               Location       Location
               1: ___          9: ___
               2: ___         10: ___
               3: ___         11: ___
               4: ___         12: ___
               5: ___         13: ___
               6: ___         14: ___
               7: ___         15: ___
               8: ___         16: ___
```

Figure 238: Modem Pool Group screen — if Group Type is combined

```
change modem-pool num                                          Page 1 of x
                                 MODEM POOL GROUP
                   Group Number: _            Group Type: combined
                     Modem Name: ______ Hold Time (min): 5_
                     Time Delay: 0_            Direction: two-way
Answer Supervision Timeout(sec): _

     Speed: LOW/300/1200___     Duplex: full     Synchronization: async

PORT PAIR ASSIGNMENTS
       Analog Digital    Analog Digital     Analog Digital      Analog Digital
     1: _____ _____    9: _____ _____     17: _____ ______    25: _____ _____
     2: _____ _____   10: _____ _____     18: _____ ______    26: _____ _____
     3: _____ _____   11: _____ _____     19: _____ ______    27: _____ _____
     4: _____ _____   12: _____ _____     20: _____ ______    28: _____ _____
     5: _____ _____   13: _____ _____     21: _____ ______    29: _____ _____
     6: _____ _____   14: _____ _____     22: _____ ______    30: _____ _____
     7: _____ _____   15: _____ _____     23: _____ ______    31: _____ _____
     8: _____ _____   16: _____ _____     24: _____ ______    32: _____ _____
```

Note:
The **Speed**, **Duplex**, and **Synchronization** fields cannot be filled out for the "integrated" pooled modem screens but can be assigned on the "combined" pooled modem screen. The integrated conversion resource automatically will adjust its speed and synchronization to the endpoint it is connected to. In synchronous mode, the integrated modem pool can operate at 1200 baud. In asynchronous mode, it can operate at 300 or 1200 baud. Full-duplex operation is always used.

Answer Supervision Timeout (sec)

This field appears only when the **Group Type** field is **combined**.

Valid entries	Usage
1 to **255**	Enter the number of seconds to wait before the far-end answers.
0	No answer supervision

CF-CB Common

This field appears only when the **Group Type** field is **integrated**.

Valid entries	Usage
y/n	Enter **y** to indicate that the CF and CB leads on the conversion resource are logically connected.

Direction

Enter the direction of the call for which modem pool will operate. This field appears only when the **Group Type** field is **combined**.

Valid entries	Usage
incoming	Converts an analog signal to digital for the data endpoint.
outgoing	Converts analog to digital (or digital to analog) for data calls.
two-way	Allows incoming and outgoing data communication.

Duplex

Display-only when the **Group Type** field is **integrated**. When the **Group Type** field is **combined**, enter the duplex mode of the conversion resources in the group.

Valid entries	Usage
full	Can talk and listen at the same time.
half	Cannot talk and listen at the same time.

Group Number

A display-only field when the screen is accessed using an administration command such as **add** or **change**.

Group Type

This field designates what physical model pool you are going to.

Valid entries	Usage
integrated	Maps to the Pooled Modem circuit pack.
combined	Maps to an external modem pool (when you have a data module and a modem).

Hold Time (min)

Valid entries	Usage
1 to **99**	Enter the maximum number of minutes that a conversion resource in the group can be held while a call waits in a queue or reserved after Data Call Preindication.

Loss of Carrier Disconnect

This field appears only when the **Group Type** field is **integrated**.

Valid entries	Usage
y/n	Enter **y** to permit conversion resource to disconnect if it detects a dropped carrier.

Modem Name

Indicates the name of the modem pool. This field appears only when the **Group Type** field is **combined**.

Valid entries	Usage
1 to 6 alphanumeric character string	Enter the name of the modem pool.

Receive Space Disconnect

This field appears only when the **Group Type** field is **integrated**.

Valid entries	Usage
y/n	Enter **y** to allow the conversion resource to disconnect after receiving 1.6 seconds of space.

Receiver Responds to Remote Loop

This field appears only when the **Group Type** field is **integrated**.

Valid entries	Usage
y/n	Enter **y** to allow the far-end modem to put conversion resource into loop back mode.

Send Space Disconnect

This field appears only when the **Group Type** field is **integrated**.

Valid entries	Usage
y/n	Enter **y** to allow the conversion resource to send 4 seconds of space before disconnecting.

Speed

Display-only when the **Group Type** field is **integrated**. When the **Group Type** field is **combined**, enter the communication speed in bits per second of the conversion resources in the group. Enter one to three speeds separated by slashes (for example, 300/1200/2400) to indicate a maximum of three running speeds.

Valid entries	Usage
LOW	0 to 300 blind sampled
300	
1200	
	1 of 2

Valid entries	Usage
2400	
4800	
9600	
19200	
	2 of 2

Synchronization

Display-only when the **Group Type** field is **integrated**. When the **Group Type** field is **combined**, enter the synchronization mode of the conversion resources in the group.

Valid entries	Usage
sync	Synchronous
async	Asynchronous

CIRCUIT PACK ASSIGNMENTS are optional on "integrated" conversion resource screens only.

Time Delay

This field appears only when the **Group Type** field is **combined**.

Valid entries	Usage
0 to **255**	Enter the time delay in seconds to insert between sending the ringing to the modem and the off-hook alert to the data module.

CIRCUIT PACK ASSIGNMENTS

Circuit Pack Location

Displays when the **Group Type** field is **integrated**. Enter the port associated with the conversion resource on the integrated modem pool circuit pack. Enter the necessary characters.

Valid entries	Usage
01 to **03** (DEFINITY CSI) or **1** to **64** (S87XX/S8300 Servers)	First and second characters are the cabinet number
A to **E**	Third character is the carrier
0 to **20**	Fourth and fifth character are the slot number
01 to **04** (Analog TIE trunks) **01** to **31**	Six and seventh characters are the circuit number
1 to **80** (DEFINITY CSI) or **1** to **250** (S87XX/S8300 Servers)	Gateway
V1 to **V9**	Module
01 to **31**	Circuit

For example, 01A0612 is in cabinet 01, carrier A, slot 06, and circuit number (port) 12.

PORT PAIR ASSIGNMENTS are optional on "combined" pooled modem screens only.

PORT PAIR ASSIGNMENTS

Analog Digital

Displays when the **Group Type** field is **combined**. Enter the port numbers of the modem/TDM pair in a conversion resource.

Two port entries are required. Enter the necessary characters.

Valid entries	Usage
01 to **03** (DEFINITY CSI) or **1** to **64** (S87XX/S8300 Servers)	First and second characters are the cabinet number

Valid entries	Usage
A to **E**	Third character is the carrier
0 to **20**	Fourth and fifth character are the slot number
01 to **04** (Analog TIE trunks) **01** to **31**	Six and seventh characters are the circuit number
1 to **80** (DEFINITY CSI) or **1** to **250** (S87XX/S8300 Servers)	Gateway
V1 to **V9**	Module
01 to **31**	Circuit

Note:
For example, 01A0612 is in cabinet 01, carrier A, slot 06, and circuit number (port) 12.

MOH Group

Use the MOH Group screen to define a collection of analog station and/or aux trunk port circuit pack ports that are connected to external audio sources for use with the Music on Hold feature.

Field descriptions for page 1

Figure 239: MOH Group screen

```
change moh-analog-group n                                         Page 1 of x
                              MOH Group 2

                            Group Name:

MOH SOURCE LOCATION
     1:          16:          31:          46:          61:
     2:          17:          32:          47:          62:
     3:          18:          33:          48:          63:
     4:          19:          34:          49:          64:
     5:          29:          35:          50:          65:
     6:          21:          36:          51:          66:
     7:          22:          37:          52:          67:
     8:          23:          38:          53:          68:
     9:          24:          39:          54:          69:
    10:          25:          40:          55:          70:
    11:          26:          41:          56:          71:
    12:          27:          42:          57:          72:
    13:          28:          43:          58:          73:
    14:          29:          44:          59:          74:
    15:          30:          45:          60:          75:
```

MOH Source Location

Type in the Music-on-hold analog or aux-trunk port location: enter the port; cabinet(1-64):carrier(A-E):slot(1-20):circuit(1-31) OR gateway(1-250):module(V1-V9):circuit(1-31).

Group Name

Enter an alpha-numeric name of the MOH group for identification.

Multifrequency-Signaling-Related Parameters

This screen sets the system or location parameters associated with multifrequency signaling. With the Multinational Locations feature enabled, multifrequency signaling can be administered per location, rather than system-wide. This screen appears when **Incoming Call Type** is **group-ii-mfc** and **Outgoing Call Type** is **none**. Page 2 of this screen appears when both **Incoming Call Type** and **Outgoing Call Type** are **group-ii-mfc**.

If the field **Use COR for All Group II Responses** is set to **y**, the **Group II Called Party Category** and **Use COR for Calling Party Category** fields do not appear.

Note:
With the Multinational Locations feature enabled, you can assign MFC signal sets per trunk group, rather than system-wide.

Field descriptions for page 1

Figure 240: Multifrequency-Signaling-Related Parameters screen

```
change multifrequency-signaling                                Page  1 of  X

             MULTIFREQUENCY-SIGNALING-RELATED SYSTEM PARAMETERS

          Incoming Call Type:                        ANI Prefix:
          Outgoing Call Type:                       Default ANI:
       Maintenance Call Type:            NEXT ANI DIGIT
                                                       Incoming:
     Maximum Resend Requests:                          Outgoing:
   Received Signal Gain (dB):
Transmitted Signal Gain (dB):

                           Request Incoming ANI (non-AAR/ARS)?
                    Outgoing Forward Signal Present Timer (sec):
                     Outgoing Forward Signal Absent Timer (sec):
MF Signaling Intercept Treatment - Incoming? _  Outgoing: ____________ _____
                              Collect All Digits Before Seizure?
                     Overlap Sending on Link-to-Link Tandem Calls?
             Private Group II Permissions and Public Interworking?
                              Convert First Digit End-of-ANI To: _
                             Use COR for All Group II Responses? _
                                 Group II Called Party Category:
                             Use COR for Calling Party Category?
```

The **ANI Prefix**, **Default ANI**, and **Collect All Digits Before Seizure** fields appear only when the value of the **Outgoing Call Type** field is **group-ii-mfc** or **mfe**.

If **Collect All Digits Before Seizure** is **y**, **Overlap Sending on Link-to-Link Tandem Calls and Convert First Digit End-of-ANI** are not displayed.

Default ANI

This field appears only when **Outgoing Call Type** is **group-ii-mfc** or **mfe**.

Valid entries	Usage
2 to **15**	Enter the PBX identification number that is sent to the CO when ANI is requested (by the CO) on a particular call but is not available, such as on tandem tie trunk calls.
blank	Use for tandeming. If this field is blank, you must enter a value in the **ANI-Not-Available** field.

ANI Prefix

This field appears only when **Outgoing Call Type** is **group-ii-mfc** or **mfe**.

Valid entries	Usage
1 to 6 digits or blank	Enter the prefix to apply to an extension when ANI is sent to the CO.

Backward Cycle Timer (sec)

Appears when the **Incoming Call Type** field is **mfe**.

Valid entries	Usage
1 to **255**	Enter the number of seconds to wait to send the check frequency after receiving an MFE signal.

Collect All Digits Before Seizure

Appears when the **Outgoing Call Type** is **group-ii-mfc** or **mfe**.

Valid entries	Usage
y	The system collects all the digits before seizing the trunk and the **ANI Req** field on the AAR and ARS Digit Conversion Table does not apply.
n	Enter **n** to control ANI collection via the ARS screens.

Convert First Digit End-of-Dial To

Appears when the **Private Group II Permissions and Public Interworking** field is **y**.

Valid entries	Usage
0 to **9**, **#**, or blank	Enter the digit used when the incoming initial end-of-ani or end-of-dial MF signal is converted on a per-switch basis.

Forward Cycle Timer (sec)

Appears when the **Incoming Call Type** field is **mfe**.

Valid entries	Usage
1 to **255**	Enter the number of seconds to wait to receive the check frequency after sending an MFE signal. Communication Manager drops the call if the time runs out before it receives check frequency.

Group II Called Party Category

Appears when the **Outgoing Call Type** field is **group-ii-mfc** and the **Use COR for All Group II Responses** field is **n**. Enter the type of group II signals that should be used on the outgoing R2-MFC call.

Valid entries	Usage
user-type	The type of telephone making the call determines the type of group II signal that the server sends (normal = ordinary telephone set, attendant = attendant console, data-call = data modules and similar data endpoints).
call-type	The dialed digits determine the type of group II signal that the server sends.

Incoming Call Type

This field defines the signal type that a CO uses to place an incoming call to the server.

Valid entries	Usage
group-ii-mfc	If the value of this field is **group-ii-mfc**, the second page of the screen displays entries for all group-I, group-II, group-A, and group-B signal types with a set of default values (see page 2 of screen).
non-group-ii-mfc	If the value is **non-group-ii-mfc**, the second page displays only group-I and group-A signal types.
mfe	Use only in Spain (multi-frequency Espanol)

Incomplete Dial Timer (sec)

Appears when the **Incoming Call Type** field is **mfe**.

Valid entries	Usage
45 to **255**	Enter the number of seconds to wait from the start of a call until the end of the check frequency of the last signal. Communication Manager drops the call if the time runs out before it receives the check frequency.

Maintenance Call Type

Appears when the **Incoming Call Type** field is **group-ii-mfc** or **non-group-ii-mfc**.

Valid entries	Usage
1	The Belgium maintenance sequence is indicated when the CO sends an MFC maintenance tone.
2	The Saudi Arabian sequence is indicated when the CO sends an MFC maintenance tone.
none	

Maximum Resend Requests

Valid entries	Usage
1 to **99**	Enter the threshold number of resend type MFC signals your server running Communication Manager accepts during an outgoing call.
1	The call is dropped if one resend signal is received.
blank	An unlimited number of resend requests is allowed.

MF Signaling Intercept Treatment - Incoming

Valid entries	Usage
y	Send the group B signal for the intercept to the CO and play intercept tone on the trunk.
n	Use normal DID/TIE/ISDN intercept treatment.

MF Signaling Intercept Treatment - Outgoing

Displays when the **Outgoing Call Type** field is **group-ii-mfc**.

Valid entries	Usage
announcement	Plays a recorded announcement for outgoing calls that cannot be completed as dialed. You select and record the message. Enter the extension number for the announcement in the associated field.
tone	Plays intercept tone for outgoing calls that cannot be completed as dialed.

MFE Type

This field only appears when **Incoming Call Type** is **mfe**.

Valid entries	Usage
2/5	Determines which public signaling Communication Manager will use.
2/6	

Outgoing Call Type

This field defines the signal type that the PBX uses to place an outgoing call into a CO.

Valid entries	Usage
group-ii-mfc	If the content of this field is **group-ii-mfc**, the system displays the third page of the screen. The third page displays entries for all group-I, group-II group-A, and group-B signal types with a set of default values.
mfe	Use only in Spain (multi-frequency Espanol)
none	If the content of this field is **none**, the system does not display the third page. In addition, **Outgoing Forward Signal Present Timer**, **Outgoing Forward Signal Absent Timer**, **ANI Prefix**, **Default ANI**, **Next ANI Digits**, and **Collect All Digits Before Seizure** will not display on field descriptions for page 1.

Outgoing Forward Signal Absent Timer (sec)

This field appears only when the content of **Outgoing Call Type** is **group-ii-mfc**.

Valid entries	Usage
11 to **255**	Enter the maximum number of seconds to elapse between forward signals on outgoing calls. The timer starts (and restarts) when a forward tone is taken off the link and it stops when the next forward tone is applied to the link.

Outgoing Forward Signal Present Timer (sec)

This field appears only when the value of **Outgoing Call Type** is **group-ii-mfc**.

Valid entries	Usage
1 to **255**	Enter the maximum number of seconds to elapse between signals on a call. This timer runs when MFC tones are being sent or received on an outgoing call. The timer starts (and restarts) when Communication Manager begins sending a forward signal and stops when Communication Manager receives the backward signal.

Outgoing Start Timer (sec)

Appears when the **Incoming Call Type** field is **mfe**.

Valid entries	Usage
1 to **255**	Enter the number of seconds to time from seizure until the beginning of the first Group A signal from the receiving end, and from the end of the check frequency until the beginning receipt of the first digit following the Group II signal.

Overlap Sending on Link-to-Link Tandem Calls

Does not appear if the **Collect All Digits Before Seizure** field is **y**. When Communication Manager has this field set to **y**, and calls are tandeming between servers, then Default ANI will be sent to the terminating switch if that switch requests ANI before Communication Manager receives it from the originating server/switch. The terminating server/switch can request ANI before the receipt of the last address digit if it is not running Communication Manager, or if it is Communication Manager with the **Request Call Category at Start of Call** field set to **y**.

Valid entries	Usage
y/n	If **y**, Communication Manager sends and receives digits one digit at a time instead of enbloc. (With enbloc, digits are not sent until the entire group of digits is received).

Private Group II Permissions and Public Interworking

Displays when the **Incoming Call Type** field is **group-ii-mfc** or **non-group-ii-mfc** and the **Outgoing Call Type** field is **group-ii-mfc** or **none**.

Valid entries	Usage
y/n	If **y**, then Communication Manager: ● Sends the category for MFC ANI for the COR of the originating party for non-private-MFC-trunk to MFC-private-trunk calls. ● Sends the Group II category received over the incoming private trunk as the outgoing Group II category on tandem private MFC calls. ● Applies MFC group II-CPC termination restrictions on incoming MFC private trunk calls. ● Checks station permissions if you call forward off-net.

Received Signal Gain (dB)

Valid entries	Usage
-15 to **3**	Enter the number for the loss/gain when the MFC port listens to the trunk port. Communication Manager listens with a range of -5 to -35 and this value moves the range (for example, a value of -5 provides a range of -10 to -40). This value also applies to Russian MF Shuttle trunks.

Request Incoming ANI (non-AAR/ARS)

Appears when the **Incoming Call Type** field is **group-ii-mfc** or **mfe** and the **Outgoing Call Type** field is **group-ii-mfc** or **mfe**. This field only applies if the incoming call via the R2-MFC trunk is terminating to a local station on this PBX.

Valid entries	Usage
y/n	If **y**, ANI should be requested on incoming R2-MFC calls.

Transmitted Signal Gain (dB)

Valid entries	Usage
-15 to **3**	Enter the number for the loss/gain when the trunk port listens to the MFC port. The MFC port generates at -5 for MFC and -8 for MFE, and this field adds gain or loss to the starting value of -5. This value also applies to Russian Shuttle trunks and Russian multi-frequency ANI.

Use COR for All Group II Responses

Appears if the **Outgoing Call Type** field is **group-ii-mfc**.

Valid entries	Usage
y/n	Enter **y** to allow the COR administered category to be used for both the calling party and called party categories.

Use COR for Calling Party Category

Appears when the **Outgoing Call Type** field is **group-ii-mfc** and the **Use COR for All Group II Responses** field is **n**. Indicates the category to send with ANI if requested on an outgoing R2-MFC call.

Valid entries	Usage
y	Use the calling facility's COR to determine category.
n	Use the calling party's user-type COR to determine category.

NEXT ANI DIGIT

Incoming

Appears when the **Incoming Call Type** field is **group-ii-mfc** and the **Outgoing Call Type** field is **group-ii-mfc** or **mfe**.

Valid entries	Usage
next-digit **next_ani_digit** **send-ani**	Enter a value to determine whether the **next_ani_digit** signal is the same as the **send-ani** signal or the **next-digit** signal or another signal defined as **next_ani_digit**.

Outgoing

Appears when the **Outgoing Call Type** field is **group-ii-mfc**.

Valid entries	Usage
next-digit **next_ani_digit** **send-ani**	Enter a value to determine whether the **next_ani_digit** signal is the same as the **send-ani** signal or the **next-digit** signal or another signal defined as **next_ani_digit**.

Field descriptions for page 2

The fields on Page 2 define call category and ANI information. For India, the ANI can be requested without the call category information.

Figure 241: Multifrequency-Signaling-Related Parameters screen

```
change multifrequency-signaling n                          Page   2 of   x
             MULTIFREQUENCY-SIGNALING-RELATED PARAMETERS  1

     Request Call Category at Start of Call? n            Outgoing II by COR
           Restart ANI from Caller Category? y             1:  1
              Number of Incoming ANI Digits: 0             2:  2
              Number of Outgoing ANI Digits: 0             3:  3
             Truncate Station Number in ANI: no            4:  4
Address Digits Include End-of-digits Signal? n             5:  5
         Call Category for Vector ii-digits? n             6:  6
               Request CPN at Start of Call? n             7:  7
          Do Not Send Group B Signals to CO? n             8:  8
   ANI Source for Forwarded & Covered Calls: caller        9:  9
                                                          10: 10
                        INCOMING    OUTGOING
          ANI Available:
      ANI Not Available:
```

Address Digits Include End-of-Digits Signal

Indicates that an outgoing forward Group I end-of-digit signal is always sent after completion of address digits upon request from the Central Office for outgoing calls.

Valid entries	Usage
y/n	Enter **y** to send an outgoing forward Group I end-of-digit signal after completion of address digits upon request from the Central Office for outgoing calls.

ANI Source for Forwarded & Covered Calls

Valid entries	Usage
caller	Send the calling party’s ANI when calls are redirected.
forwarder	Send the forwarding party’s ANI when calls are redirected.

Call Category for Vector ii-digits

Allows you to use the call category digit as the ii-digits on call vector steps.

Valid entries	Usage
y/n	If **y**, the call category digit, which is a part of ANI, is used as the ii-digits on call vector steps.

Do Not Send Group B Signals to CO

This field appears only if the **Incoming Call Type** field is **group-ii-mfc**. This field allows completion of a call without Group-B signals.

Valid entries	Usage
y	If **y**, does not send Group-B signals to complete an incoming call.
n	If **n**, sends Group-B signals to complete an incoming call.

Number of Incoming ANI Digits

Valid entries	Usage
0 to **15**	Enter the number of ANI digits for incoming MFC calls.

Number of Outgoing ANI Digits

This field applies to Russian shuttle trunks, and MFC and MFE trunks.

Valid entries	Usage
0 to **15**	Enter the number of ANI digits for outgoing MFC calls. In India or any country where end-of-ani and end-of-digits are not defined for Tones to CO on Outgoing Forward Calls - Group I, Communication Manager appends ANI-Not-Available digits to ANI digits if the actual ANI length is less than the number entered in this field. If end-of-ani or end-of-digits are defined, this field is used in conjunction with Truncate Station Number in ANI as a maximum ANI length. For India, even if the length of ANI is defined, if the timeout occurs during the ANI collection, the call is routed with the ANI digits already collected.

Outgoing II by COR

Appears only if either **Use COR for Calling Party Category** or **Use COR for All Group II Responses** on page 1 are set to **y**. The Group II signal sent to the CO on outgoing calls can be administered per COR (Class of Restriction) and per trunk group. The Group II signal is administered per COR. That per-COR value in turn can be mapped into a possibly different outgoing signaling parameter set. The values in the **Outgoing II by COR** fields administer that outgoing mapping.

Valid entries	Usage
1 to **10**	Enter a number between 1 and 10 that maps to the Group II signal Communication Manager sends to the CO on outgoing calls

Request Call Category at Start of Call

Indicates that the Send-ANI backward signal requesting for the caller-category information will be sequenced differently in the MFC signaling flow. The Caller-category Request backward signal is disjointed from the ANI request.

Valid entries	Usage
y/n	If **y**, the Send-ANI backward signal corresponds exclusively to the caller-category request. In response to this signal, Communication Manager sends a forward signal containing the caller-category information on outgoing calls. On incoming calls, Communication Manager sends the Send-ANI backward signal upon receipt of the first address signal.

Request CPN at Start of Call

This field appears only if the **Incoming Call Type** field is **group-ii-mfc**. Provides for Communication Manager to collect ANI and call category immediately after receipt of the first address digit.

Valid entries	Usage
y/n	If **y**, provides ANI (Calling Party Number (CPN) and call category) immediately after receiving the first address digit.

Restart ANI from Caller Category

Display-only field.

Valid entries	Usage
y/n	If **y**, Communication Manager sends the caller-category signal later again when the signals for Caller-Category and ANI requests are the same and this signal is received after the Next-Digit forward signals have been received.

Truncate Station Number in ANI

This field applies to Russian shuttle trunks, and MFC and MFE trunks.

Valid entries	Usage
beginning **ending** **no**	This field defines the side of the extension number from which to truncate when station ANI is sent to the CO and the combined length of the ANI prefix and extension number is greater than Number of Outgoing ANI Digits. The ANI prefix (either MFC or COR) is not truncated. There is no effect if Default ANI is sent.

INCOMING / OUTGOING

ANI Available

Valid entries	Usage
1 to **15** or blank	Enter the number for the signal to be used for incoming ANI-Available.

ANI Not Available

You must enter a value if the **Default ANI** field is blank.

Valid entries	Usage
1 to **15** or blank	Enter the number for the signal to be used for outgoing ANI-Available. Communication Manager outpulses the End-of-Dial backward signal when the ANI-Not-Available forward signal is received on incoming calls. Communication Manager outpulses the ANI-Not-Available forward signal to the CO on outgoing calls where ANI is not possible.

Field descriptions for page 3

The fields shown on Page 3 of the Multifrequency-Signaling-Related System Parameters screen define the meaning of MFC tones for calls originated at the CO. See Definitions of Group I, II, A, and B signals on page 674 for more information. This screen appears only if the **Incoming Call Type** field is **group-ii-mfc** or **non-group-ii-mfc**.

When the screen initially appears, either of two sets of default values is possible. One set is for the group II call type; the other set is for non-group II call type. In each set, the default value for each field is set to the most common value.

The Multifrequency-Signaling-Related Parameters screen shows the defaults when the **Incoming Call Type** field is **group-ii-mfc**. A variation appears if **Incoming Call Type** is **non-group-ii-mfc**. When **Incoming Call Type** is **non-group-ii-mfc**, group II and group B columns do not appear.

Figure 242: Multifrequency-Signaling-Related Parameters screen

```
change multifrequency-signaling                                 Page 3 of X
              MULTIFREQUENCY-SIGNALING-RELATED PARAMETERS

  INCOMING FORWARD SIGNAL TYPES         INCOMING BACKWARD SIGNAL TYPES
  (Tones from CO)                       (Tones to CO)

        Group-I             Group-II            Group-A             Group-B
   11: ignored          1: normal           1: next-digit      1: free
   12: ignored          2: normal           3: end-of-dial     2: busy
   13: ignored          3. normal          __:                 4: congestion
   14: ignored          4: normal          __: ________________7: intercept
   15: ignored          5: normal          __: ________________ __: __________
                        6: normal          __: ________________ __: __________
                        7: normal          __: ________________ __: __________
                        8: normal          __: ________________ __: __________
                        9: normal          __: ________________ __: __________
                       10: normal          __: ________________ __: __________
                       11: normal          __: ________________ __: __________
                       12: normal          __: ________________ __: __________
                       13: normal          __: ________________ __: __________
                       14: normal          __: ________________ __: __________
                       15: normal          __: ________________ __: __________
```

INCOMING FORWARD SIGNAL TYPES (Tones from CO)

Group I

Message codes 11 to 15 display. (Numbers 1 through 10 are assigned to the digits of the destination telephone number.) Assign a meaning to each code. See Definitions of Group I, II, A, and B signals on page 674 for signal type.

Valid entries	Usage
drop	If **Incoming Call Type** is **group-ii-mfc**
ani-avail	
end-of-ani	
end-of-dial	
ignored	
maint-call	
ani-not-avail	
send-congest	
drop	If the **Incoming Call Type** is **non-group-ii-mfc**
ignored	

Group II

Message codes 1 to 15 display. Assign a meaning to each code.

Valid entries	Usage
attendant	See Definitions of Group I, II, A, and B signals on page 674 for signal type.
busy-rt-attd	
data-call	
data-verify	
drop	
	1 of 2

Valid entries	Usage
maint-call	
send-intercept	
toll-auto	
toll-operator	
normal	
	2 of 2

INCOMING BACKWARD SIGNAL TYPES (Tones to CO)

Group A

Message codes 11 to 15 display. (Numbers 1 through 10 are assigned to the digits of the destination telephone number.) Assign a meaning to each code.

Valid entries	Usage
congestion	See Definitions of Group I, II, A, and B signals on page 674 for signal type.
end-of-dial	
intercept	
next-ani-digit	
next-digit	
send-ani	
setup-sppath	

Group B

This field does not appear if the **Do Not Send Group B Signals to CO** field is **y**. Message codes between 1 and 15 display. Assign a meaning to each code.

Valid entries	Usage
busy	See Definitions of Group I, II, A, and B signals on page 674 for signal type.
congestion	
free	
mct	
tariff-free	
tie-free	
toll-busy	
intercept	

Field descriptions for page 4

The fields shown on this page define the meaning of MFC tones for calls originated at the PBX. See Definitions of Group I, II, A, and B signals on page 674 for more information.

Page 4 of the Multifrequency-Signaling-Related System Parameters screen only appears if **Outgoing Call Type** is **group-ii-mfc** or **mfe**.

Figure 243: Multifrequency-Signaling-Related System Parameters screen

```
change system-parameters multifrequency-signaling                 Page 4 of x
                MULTIFREQUENCY-SIGNALING-RELATED SYSTEM PARAMETERS

  OUTGOING FORWARD SIGNAL TYPES               OUTGOING BACKWARD SIGNAL TYPES
  (Tones to CO)                               (Tones from CO)

        Group-I             Group-II             Group-A             Group-B
   11: ______________   2: normal           1: next-digit       1: free
   12: ______________   5: attendant        2: congestion       2: busy
   13: ______________   6: data-call        3: end-of-dial      3: congestion
   14: ______________  __: ______________   4: congestion       4: congestion
   15: ______________  __: ______________   5: call-info-ani    5: congestion
                       __: ______________   6: congestion       6: free
                       __: ______________   7: last-2-digits    7: intercept
                       __: ______________   8: last-3-digits    8: congestion
                       __: ______________   9: congestion       9: congestion
                       __: ______________  10: congestion      10: congestion
                       __: ______________  11: congestion      11: congestion
                       __: ______________  12: congestion      12: congestion
                       __: ______________  13: congestion      13: congestion
                       __: ______________  14: congestion      14: congestion
                       __: ______________  15: congestion      15: congestion
```

OUTGOING FORWARD SIGNAL TYPES (Tones to CO)

Group I

Message codes 11 to 15 appear. (Numbers 1 through 10 are assigned to the digits of the destination telephone number.)

Valid entries	Usage
end-of-digits	Assign a meaning to each code. See Definitions of Group I, II, A, and B signals on page 674 for signal type.
ani-avail	
end-of-ani	
ani-not-avail	

Group II

Message codes between 1 and 15 display. Assign a meaning to each code. Each entry can only appear once in the group II column.

Valid entries	Usage
attendant	See Definitions of Group I, II, A, and B signals on page 674 for signal type.
data-call	
toll-auto	
normal	

OUTGOING BACKWARD SIGNAL TYPES (Tones from CO)

Group A

Message codes between 1 and 15 display. Assign a meaning to each code.

Valid entries	Usage
send-ani	See Definitions of Group I, II, A, and B signals on page 674 for signal type.
congestion	
drop	
end-of-dial	
last-2-digits	
last-3-digits	
last-digit	
next-ani-digit	
next-digit	
restart	
intercept	
	1 of 2

Valid entries	Usage
resend-digit	
setup-sppath	
	2 of 2

Group B

Valid entries	Usage
busy	Message codes between 1 and 15 display. Assign a meaning to each code. See Definitions of Group I, II, A, and B signals on page 674 for signal type.
congestion	
free	
mct	
tariff-free	
toll-busy	
intercept	

Definitions of Group I, II, A, and B signals

Group I signals

Group I signals are a set of forward signals generated by the originating Avaya S8XXX Server.

ani-avail

Used in Hungary. If this signal is defined and ANI is requested on outgoing R2-MFC calls, ANI is sent to the CO before ANI caller digits are sent. This signal is sent after the ANI caller category signal.

ani-not-avail

Used on DOD calls in Brazil and Columbia. Communication Manager sends this signal to the CO when it receives an ANI request and the caller's number is not available.

digits 1 to 10

The signals from group I.1 to I.10 are reserved for address digits 0 to 9.

drop

When this signal is received from the CO, Communication Manager starts the disconnect sequence and drops the call.

end-of-ani

This signal is used on DOD and DID calls. Communication Manager sends this signal to indicate the end-of-ANI digits when ANI digits are sent to the CO.

end-of-dial

This signal is used when open numbering is used on DID calls. The CO sends this signal to indicate the end-of-dial digits and Communication Manager responds with a request for a group II signal.

end-of-digits

This signal is sent by the originating Avaya S8XXX Server that makes outgoing calls, sends digits, and receives a next-digit group A signal from the destination server or switch when there are no more digits to be sent.

This signal is also sent when Communication Manager does not have end-of-ani assigned, makes an outgoing call, sends ANI, and receives a call-info-ani group A signal from the destination end when there are no more ANI digits to be sent.

If both end-of-digits and end-of-ani are assigned, Communication Manager uses end-of-ani after it sends the last ANI digit and end-of-digits after sending the last called-number digit.

ignored

If this signal is received from the CO, Communication Manager sends a corresponding signal (A.1, and so on) but no action is taken in the response and it is not counted as a digit. In Belgium, this signal is not acknowledged.

maint-call

The CO sends a signal to indicate that a call is a maintenance call and Communication Manager prepares the special maintenance call sequences for the CO. This signal can be used on DID calls in Saudi Arabia.

send-congestion

When Communication Manager receives this signal from the CO on a DID call, it returns a congestion signal (group A), in compel (not pulse) mode, to the CO.

Group II signals

Group II signals are a more elaborate set of forward signals generated by the originating server.

attendant

If Communication Manager receives this signal on DID calls, the call terminates at an attendant regardless of the extension dialed. On DOD calls, this signal is sent to the CO if the CO requests calling-category information and the originating extension is an attendant. This signal is used on both DID and DOD calls.

busy-rt-attd

If Communication Manager receives this signal on DID calls, the call terminates at an attendant if the called extension is busy or at the called extension if it is not busy. This signal is used on DID calls.

data-call

This signal is treated the same as the data-verify signal except that it does not require a terminating extension to be a data extension.

data-verify

If Communication Manager receives this signal on DID calls and the terminating extension is not a data extension, it sends intercept treatment. On DOD calls, this signal is sent to the CO if the CO requests calling-category information and the originating extension is a data extension. This signal is used on both DID and DOD calls.

drop

When this signal is received from the CO, Communication Manager starts the disconnect sequence and drops the call.

maint-call

The CO sends a signal to indicate that a call is a maintenance call and Communication Manager prepares the special maintenance call sequences for the CO.

normal

This signal indicates that the caller is a normal subscriber. If it is received on a DID call, the call is terminated at the called extension. For an outgoing MF signaling call that uses group II signaling, this signal is sent to the CO when the CO requests calling-category information and the originating extension is a station. This signal is used in both DID and DOD calls.

send-intercept

If Communication Manager receives this signal from the CO on a DID call, it returns group B intercept signal to the CO.

toll-auto

This signal is used in China. This signal indicates that a call is an automatic toll call. When the call terminates at a busy station and a special busy signal is defined, the busy signal is sent to the CO. You can define the special busy signal by choosing the option toll-busy on the incoming group B signals.

toll-operator

This signal, used in China, is treated as a normal subscriber signal. See the normal definition.

Group A signals

Group A signals are backward signals generated by the destination server/switch.

send-ani

The CO sends this signal to request calling-party category and sends additional signals to request ANI digits. This signal is sent to the CO when Communication Manager requests ANI digits on DID calls. This signal is used on both DOD and DID calls.

congestion

The CO sends this signal to indicate that it is experiencing network congestion. When Communication Manager receives this signal on DOD calls, it drops the trunk and plays reorder tone to the calling party. This signal is used on DOD calls.

drop

When this signal is sent, the receiving end starts the disconnect sequence.

end-of-dial

This signal is sent to indicate the end of the address digit string. For MF group II calls, this signal requests a group II signal and switches the sender over to the group B signaling mode. This signal is used on both DID and DOD calls.

intercept

The CO sends this signal to indicate the call has been terminated to an invalid destination. When Communication Manager receives this signal on DOD calls, it drops the trunk and plays intercept tone to the calling party. This signal is used on DOD calls.

resend-digit

Communication Manager sends this signal to adjust the outpulsing pointer so that the last digit can be resent again. This signal is used on DOD calls.

last-digit

Communication Manager sends this signal to adjust the outpulsing pointer so that the last 2 digits can be resent. This signal is used on DOD calls.

last-2-digits

Communication Manager sends this signal to adjust the outpulsing pointer so that the last 3 digits can be resent. This signal is used on DOD calls.

last-3-digits

Communication Manager sends this signal to adjust the outpulsing pointer so that the last 4 digits can be resent. This signal is used on DOD calls.

next-digit

Communication Manager sends this signal to request the next digit. This signal is used on both DID and DOD calls.

next-ani-digit

Communication Manager sends this signal to request the next ANI digit. This signal is used on DID and DOD calls.

restart

Communication Manager sends this signal to request the whole digit string again. This signal is used on DOD calls.

setup-sppath

The CO sends this signal to Communication Manager to set up a speech path. This signal is used on DOD calls and on DID calls in Belgium.

Group B signals

Group B signals enhance group A signals for backward signaling from the destination end by providing the status of the called party. In addition, if the originating server uses group II signals, the destination end answers with group B signals. Group B signals are as follows:

busy

This signal is sent to indicate that the called party is busy. On DID calls, the signal is sent to the CO if there is no coverage point to terminate the call. If Communication Manager receives this signal on DOD calls, it plays busy tone to the calling party and drops the trunk.

congestion

This signal is sent to indicate that the system is congested and the call cannot be terminated successfully. On DID calls, the signal is sent to the CO to indicate that a resource is not available. On DOD calls, if Communication Manager receives this signal, reorder tone is played to the calling party and the trunk is dropped.

free

This signal indicates that the called party is idle. On DID calls, the signal is sent to the CO to indicate that the called party is idle and the call is terminated successfully. If Communication Manager receives this signal on DOD calls, it connects the trunk to the calling party.

intercept

This signal indicates that the called party number is not in service or is not correct. On DID calls, if intercept treatment is set to provide a tone, tone is sent to the CO to indicate that the called number is not valid. If Communication Manager receives the signal on DOD calls, it plays intercept tone to the calling party and drops the trunk.

mct

This signal identifies the call as one that needs to be traced by the CO. Communication Manager then generates an MFC Call Trace Backward Signal (administered on the Multifrequency-Signaling-Related System-Parameters screen) during call setup instead of the “free” signal. If the terminating station’s COR has this feature set to **y**, the CO collects trace information before releasing the calling party.

Note:
If the station’s COR has **MF Incoming Call Trace** set to **y** and the "mct" signal is not defined, then the "free" signal is sent.

tariff-free

This signal is sent when the trunk group provides an “800” service. Communication Manager generates an MFC tariff-free backward signal (administered on the System-Parameters Multifrequency-Signaling screen) during call setup instead of the “free” signal, facilitating CO billing.

Note:
If the trunk is administered as a tariff-free trunk and the "tariff-free" signal is not defined, then the "free" signal is sent.

tie-free

This signal is used only when an incoming call is received and defined and the incoming facility is a tie trunk. Otherwise, the free signal is used.

toll-busy

This signal, used in China, is sent to indicate that the called party is busy if the call is an automatic toll call.

Multiple Level Precedence & Preemption (MLPP) Parameters

Use this screen to set up system parameters for the Multiple Level Precedence & Preemption feature.

Field descriptions for page 1

Figure 244: Multiple Level Precedence and Preemption Parameters screen

```
change system-parameters mlpp                                   Page  1 of  x

                 MULTIPLE LEVEL PRECEDENCE & PREEMPTION PARAMETERS
  ANNOUNCEMENTS
          Blocked Precedence Level: 6801          Service Interruption: 6803
     Unauthorized Precedence Level: 6802            Busy, Not Equipped: 6804
                       Vacant Code: 6805

  PRECEDENCE CALLING-DIALED DIGIT ASSIGNMENT
     Flash Override: 0  Flash: 1  Immediate: 2  Priority: 3  Routine: 4

         Attendant Diversion Timing (sec): 60
            Remote Attendant Route String:
     Worldwide Numbering Dial Plan Active? y      Default Route Digit: 0
            Precedence Call Timeout (sec): 30
      Line Load Control Restriction Level: 0
          WNDP Emergency 911 Route String:
                   Preempt Emergency Call?
                   Default Service Domain: 1
       ISDN Precedence Call Timeout (sec): 30
```

ANNOUNCEMENTS

Blocked Precedence Level

Valid entries	Usage
Valid extension or blank	Enter the extension of the Blocked Precedence Level announcement you want to use.

Busy, Not Equipped

Valid entries	Usage
Valid extension or blank	Enter the extension of the Busy, Not Equipped for Preemption announcement you want to use.

Service Interruption

Valid entries	Usage
Valid extension or blank	Enter the extension of the Service Interruption announcement you want to use.

Unauthorized Precedence Level

Valid entries	Usage
Valid extension or blank	Enter the extension of the Unauthorized Precedence Level announcement you want to use.

Vacant Code

Valid entries	Usage
Valid extension or blank	Enter the extension of the Vacant Code announcement you want to use.

PRECEDENCE CALLING-DIALED DIGIT ASSIGNMENT

CAUTION:
Avaya recommends that you do not change the default Precedence Calling dialed digits unless you are coordinating this change with other companion networks in your system. If the Precedence Calling digits do not match across networks, the system does not properly process the calls. Each of the Precedence Calling digits must be different. You cannot use the same digit for two different precedence levels.

Attendant Diversion Timing (sec)

Valid entries	Usage
10-99 or blank	

Default Route Digit

Appears only when Worldwide Numbering Dial Plan Active is **y**. You must enter a valid digit in this field.

Valid entries	Usage
0	Voice call (the default value)
1	Circuit switched data call
2	Satellite avoidance call
3	(reserved)
4	(reserved)
5	Hotline voice grade call
6	Hotline data grade call
7	(reserved)
8	(reserved)
9	(reserved)

Default Service Domain

Valid entries	Usage
0 to **16777215**	This number defines the system service domain, and must be unique within a switching network. The system uses the system service domain to determine eligibility for precedence calling when interswitch precedence calls over non-ISDN trunks occur.

Flash

Valid entries	Usage
0 to **9** or blank	Enter the digit assignment for Flash precedence level calls.Default is **1**.

Flash Override

Valid entries	Usage
0 to **9** or blank	Enter the digit assignment for Flash Override precedence level calls. Default is **0**.

Immediate

Valid entries	Usage
0 to **9** or blank	Enter the digit assignment for Immediate precedence level calls.Default is **2**.

ISDN Precedence Call Timeout (sec)

Valid entries	Usage
4 to **30**	This timeout is used instead of the Precedence Call Timeout when the call is from an MLPP ISDN-PRI trunk. Default is **30** seconds.

Line Load Control Restriction Level

These system levels determine what stations, based on their COR, will be restricted from originating calls.

Valid entries	Usage
0	Feature not active (no restrictions) (default).
2	Restrict stations with a COR assigned to LLC levels 2, 3, and 4.
3	Restrict stations with a COR assigned to LLC levels 3 and 4.
4	Restrict stations with a COR assigned to LLC level 4.

Precedence Call Timeout (sec)

A busy user receives a precedence call waiting tone only if the incoming call cannot be connected and cannot preempt the user. The called party hears the tone every 10 seconds until answered or the administered time-out occurs. If ignored, the caller is diverted to an attendant or a call-forwarded station.

Valid entries	Usage
4 to **30**	Default is **30**. Enter the number of seconds before a precedence call remains in call waiting status before it is diverted.

Preempt Emergency Call

When this field is set to **y**, an Emergency 911 call made from a preemptable station can be preempted by a higher precedence call

Valid entries	Usage
y/n	Enter **y** to allow preemption of an Emergency 911 call by a higher precedence call.

Priority

Valid entries	Usage
0 to **9** or blank	Enter the digit assignment for Priority precedence level calls.Default is **3**.

Remote Attendant Route String

Valid entries	Usage
1 to 24 digits or blank	Enter a user-defined telephone to which a precedence call can be routed when no console or night telephone is administered.

Routine

Valid entries	Usage
0 to **9** or blank	Enter the digit assignment for Routine precedence level calls. Default is **4**.

WNDP Emergency 911 Route String

Valid entries	Usage
1 to 24 digits or blank	Valid entries for this field can be a trunk access code (TAC), the AAR or the ARS access code, a WNDP access code, or an extension. If you use a WNDP access code, use the access code for the lowest precedence calling level in the system. **Note:** An Emergency/911 call is a call that routes using the ARS table with the call type defined as either "alrt" or "emer."

Worldwide Numbering Dial Plan Active

Valid entries	Usage
y/n	Enter **y** to enable the Worldwide Numbering Dial Plan. Default is **n**.

Music Sources

This screen appears only when, on the **System Parameters Customer-Options (Optional Features)** screen, **Tenant Partition?** is **y**. Use this screen to define music sources for Tenant Partitions. Each music source defined on this screen can be used by one or more Tenant Partitions. However, a partition can have only one music source.

Note:
If you use equipment that rebroadcasts music or other copyrighted materials, you might be required to obtain a copyright license from, or pay fees to, a third party. You can purchase a Magic Hold system, which does not require such a license, from Avaya Inc. or Avaya's business partners.

Field descriptions for page 1

Figure 245: Music Sources screen

```
change music-source           Music Sources                      Page 1 of X

         Source No     Type       Source             Description

                1      music      Type: ext 30002   music-on-extension
                2      music      Type: group 10    music-on-group
                3      music      Type: a0904       music-on-part
                4      tone                         tone-on-hold
                5      none
                6      none
                7      none
                8      none
                9      none
               10      none
               11      none
               12      none
               13      none
               14      none
               15      none
```

Description

This field appears only if you entered **music** or **tone** in **Type**.

Note:
When Tenant Partitioning is enabled, **Music/Tone on Hold** on the Feature-Related System Parameters screen disappears. However, the value in that field (**tone**, **music**, or **none**) appears as the first entry on the Music Sources screen. If the value was **music**, the port number also appears on the Music Sources screen. When Tenant partitioning is disabled, **Music/Tone on Hold** reappears on the Feature-Related System Parameters screen, along with the values from the Music Sources screen.

Valid entries	Usage
20 alpha-numeric character (max)	Enter a description of the administered music source.

Source

This field appears only if you entered **music** in **Type**. Enter the necessary characters.

Valid entries	Usage
ext	audio source extension for a single or group audio source
group	a Music-on-Hold analog group number
port	an analog or auxiliary trunk source location

Source No

Display only field - the number assigned to this source. The maximum number of music sources is 20 for DEFINITY CSI. This screen appears with the appropriate pages to accommodate the number of music sources your system can support.

Type (column)

If you entered a value in **Music/Tone on Hold** on the Feature-Related System Parameters screen, that value appears in this field.

Valid entries	Usage
music	Enter the type of treatment to be provided by the music source.
tone	Only one music source can use this value.
none	

Type (field)

This field appears only when the entry in the **Type** column is **music**.

Valid entries	Usage
ext **group** **port**	Indicate whether the source is an announcement extension, an audio group, or a port on a VAL board. **Note:** After a valid value is entered, a blank field appears for entry of the appropriate source identifier (extension number, audio group number, or port number).

Network Facilities

The ISDN Network-Facilities screen is used to administer new network-provided service or feature names and corresponding ISDN PRI (network specific facilities information element) encodings, for call-by-call trunk groups. Values for pre-defined facilities are displayed at the top of the screen and are display-only. User-defined facilities and services can be entered in the fields below.

When **Usage Allocation Enhancements** on the System Parameters Customer Options screen is set to **y**, page 2 of the Network Facilities screen appears, allowing for administration of additional user-defined entries.

For more information on usage allocation, see Call-by-call Service Selection in *Avaya Aura™ Communication Manager Feature Description and Implementation*, 555-245-205.

Figure 246: Network Facilities screen - page 1

```
change isdn network-facilities                                  Page 1 of x

                              NETWORK-FACILITIES
               Facility                                Facility
   Name      Type  Coding                      Name      Type  Coding
sub-operator  0    00110                   mega800        1    00010
operator      0    00101                   megacom        1    00011
outwats-bnd   1    00001                   inwats         1    00100
sdn           1    00001                   wats-max-bnd   1    00101
accunet       1    00110                   lds            1    00111
i800          1    01000                   multiquest     1    10000
____________ ___  ______                  ____________ ___  ______
____________ ___  ______                  ____________ ___  ______
____________ ___  ______                  ____________ ___  ______
____________ ___  ______                  ____________ ___  ______
____________ ___  ______                  ____________ ___  ______
____________ ___  ______                  ____________ ___  ______
____________ ___  ______                  ____________ ___  ______
____________ ___  ______                  ____________ ___  ______
____________ ___  ______                  ____________ ___  ______
```

Figure 247: Network Facilities screen - page 2

```
change isdn network-facilities                                  Page 2 of x

                              NETWORK-FACILITIES
               Facility                                Facility
   Name      Type  Coding                      Name      Type  Coding
___________ ___  ______                  ____________ ___  ______
____________ ___  ______                  ____________ ___  ______
____________ ___  ______                  ____________ ___  ______
____________ ___  ______                  ____________ ___  ______
____________ ___  ______                  ____________ ___  ______
____________ ___  ______                  ____________ ___  ______
____________ ___  ______                  ____________ ___  ______
____________ ___  ______                  ____________ ___  ______
____________ ___  ______                  ____________ ___  ______
```

Field descriptions for page 1

Name

Valid entries	Usage
printable alphanumeric characters	Enter the name for the feature or service.

Facility Type

Enter the facility type. For types **2** and **3**, **Usage Allocation Enhancements** on the System Parameters Customer Options screen must be **y**.

Valid entries	Usage
0 - feature	Enter **0** for predefined features.
1 - service	Enter **1** for predefined services.
2 - incoming	Enter **2** for an incoming-type user-defined entry.
3 - outgoing	Enter **3** for an outgoing-type user-defined entry.

Facility Coding

Valid entries	Usage
characters	Enter the ISDN-specified value for this service or feature.

Node Names

See **IP Node Names**.

Node Number Routing

This screen specifies the routing pattern associated with each node in a public or private network. Node Number Routing is a required capability for Extension Number Portability (ENP) and is associated with the Uniform Dial Plan (UDP).

Field descriptions for page 1

Figure 248: Node Number Routing screen

```
change node-routing n                                              Page 1 of x
                                NODE NUMBER ROUTING
                             Partitioned Group Number: 1

    Route      Route      Route      Route      Route      Route      Route
     Pat        Pat        Pat        Pat        Pat        Pat        Pat

              15 ___     30 ___     45 ___     60 ___     75 ___     90 ___
   1 ___      16 ___     31 ___     46 ___     61 ___     76 ___     91 ___
   2 ___      17 ___     32 ___     47 ___     62 ___     77 ___     92 ___
   3 ___      18 ___     33 ___     48 ___     63 ___     78 ___     93 ___
   4 ___      19 ___     34 ___     49 ___     64 ___     79 ___     94 ___
   5 ___      20 ___     35 ___     50 ___     65 ___     80 ___     95 ___
   6 ___      21 ___     36 ___     51 ___     66 ___     81 ___     96 ___
   7 ___      22 ___     37 ___     52 ___     67 ___     82 ___     97 ___
   8 ___      23 ___     38 ___     53 ___     68 ___     83 ___     98 ___
   9 ___      24 ___     39 ___     54 ___     69 ___     84 ___     99 ___
  10 ___      25 ___     40 ___     55 ___     70 ___     85 ___
  11 ___      26 ___     41 ___     56 ___     71 ___     86 ___
  12 ___      27 ___     42 ___     57 ___     72 ___     87 ___
  13 ___      28 ___     43 ___     58 ___     73 ___     88 ___
  14 ___      29 ___     44 ___     69 ___     74 ___     89 ___
```

Partitioned Group Number

This display-only field displays the partitioned group number associated with the node numbers being administered.

Valid entries	Usage
Display only	The partitioned group number is either specified on the command line or defaults to partitioned group number 1.

Node Number

This display-only field lists the node number to be changed.

Valid entries	Usage
Display only	Two pages display simultaneously for a total of 200 nodes (100 per page). For example, entering change node-routing 87 displays nodes 1 through 199, and entering change node-routing 151 displays nodes 100 through 299. However, entering change node-routing 999 displays nodes 900 through 999 on one page.

Route Pat

Enter the routing pattern associated with the corresponding node number. This field repeats the same number of times as there are node numbers on the page.

Valid entries	Usage
1 to **254**	Enter a number between 1 and 254, or blank.

Numbering — Private Format

This screen supports Private Numbering Plans (PNP). It allows you to specify the digits to be put in the Calling Number information element (IE), the Connected Number IE, and the QSIG Party Number for extensions in the Private Numbering Plan.

Communication Manager supports private-network numbers up to 15 digits long. If the total number — including the level 1 and 2 prefixes, the PBX identifier, and the extension — is more than 15 digits long, neither QSIG Party Numbers nor the information elements are created or sent.

Field descriptions for page 1

Figure 249: Numbering — Private Format screen

```
change private-numbering 0                                Page 1 of 2

                          NUMBERING - PRIVATE FORMAT

Ext  Ext          Trk         Private Total
Len  Code         Grp(s)      Prefix  Len                 Total Administered: 8
5    attd                     3920    4                      Maximum Entries: 540
5    70                       30353   10
4    200                      303538  10
5    510                      30353   10
4    2100                     303538  10
5    5000                     30353   10
5    5200                     30353   10
6    6000                     3035    10
```

Ext Code

Allows for groups of extensions to be administered.

Note:
When **0** alone is entered, the **Ext Len** field must be 1 and the DDD number must be 10-digits.

Valid entries	Usage
0 to **13** or blank	The **Ext Code** can be up to 13-digits long depending on the **Ext Len** field entry. The entry cannot be greater than the **Ext Len** field entry. For example, in the case of a 4-digit **Ext Len** field entry, an Ext Code of **12** is the equivalent of all extensions of the screen 12xx, excluding any explicitly listed longer codes. If a code of 123 is also listed, the 12 code is equivalent of all extensions of the screen 12xx except extensions of the screen 123x. The coding precludes having to list all the applicable 12xx extensions.
attd	To generate a private calling number for a call from the attendant group.

Ext Len

Specifies the number of digits the extension can have. On page 1, this field displays the extension length entered as a qualifier on the command line (change private-numbering n).

Valid entries	Usage
0 to **13** or blank	Corresponds to the extension lengths allowed by the dial plan.

Maximum Entries

Valid entries	Usage
System maximum	Display only. Indicates the maximum number of private numbering entries that can be administered on the system.

Private Prefix

Valid entries	Usage
0 to **9**, or blank	The number that is added to the beginning of the extension to form a Private Identification Number. The length of the prefix and the extension must at least equal the total length.

Total Administered

Valid entries	Usage
0 to system maximum	Display only. Indicates the number of private numbering entries that are currently administered on the system.

Total Len

Valid entries	Usage
0 to **13**	The total number of digits to send.

Trk Grp(s)

Communication Manager generates the station's identification number if there is an entry in the **Ext Code** field, and this field is administered with the trunk group number carrying the call.

Valid entries	Usage
1 to 7 digits	Enter the valid administered ISDN trunk-group number or a range of group numbers. For example, if trunk groups 10 through 24 use the same CPN Prefix, enter **10** to **24**.
blank	The identification numbers are not dependent on which trunk group the call is carried.

Field descriptions for page 2

Page 2 of the Numbering — Private screen provides blank fields for new entries. See page 1 for field descriptions.

Numbering — Public/Unknown Format

The screen allows you to specify the desired digits for the Calling Number IE and the Connected Number IE (in addition to the QSIG Party Number) for any extension in the Public and/or Unknown Number Plans.

This screen is used for ARS public trunks as well as SIP Enablement Services (SES) trunks. It supports the ISDN Call Identification Display feature. The feature provides a name/number display for display-equipped stations within an ISDN network. The system uses the caller's name and number and displays it on the called party's display. Likewise, the called party's name and number can be displayed on the caller's display.

In Communication Manager 3.1 and later, the Public-Unknown Numbering screens support 9,999 entries. The ANI table, which this screen uses, is increased from 240 to 9,999 entries. This increase is for S8500 and S87XX Servers only. The other servers keep the maximum of 240 entries.

Access the Numbering — Public/Unknown screen with the command **`change public-unknown-numbering`** ***`n`***, where ***`n`*** is the length of a value between **0** and **7** appearing in the **Ext Code** column.

Note:

Use the command **`change public-unknown-numbering`** ***`n [ext-digits x] [trunk-group trunk-group-number]`*** to administer the desired digits for name and number display on display-equipped stations in an ISDN network. This trunk-group command option displays valid results only when used in conjunction with the ***`ext-digits`*** option. Otherwise, an error message is returned.

The screen consists of two pages: page 1 displays up to 30 **Ext Code** entries matching the requested **Ext Code** length entered on the command line, and page 2 provides 30 blank entries for new user input. If there is sufficient room on the screen, **Ext Code** entries that are longer than the specified length are also displayed. Enter a length of **0** to designate the attendant. If there are more entries of length ***`n`*** than can be displayed, modify your command to use the **`ext-digits`** ***`x`*** command line modifier.

Administer these screens if either the **Send Calling Number**, **Send Connected Number** field is specified, or the **Supplementary Service Protocol** field is **b** on the Trunk Group screen.

Note:

If the table is not properly administered and the **Send Calling Number** or **Send Connected Number** field is **y** or **r** and the **Numbering Format** field on the ISDN Trunk Group screen is **public** or **unknown**, the Calling Number and Connected Number IE are not sent. If the table is not administered, but the **Send Calling Number** or **Send Connected Number** field is **public** or **unknown**, the Identification Number (PartyNumber data type) is not sent for QSIG PartyNumbers. In this case, the ASN.1 data type containing the PartyNumber (PresentedAddressScreened, PresentedAddressUnscreened, PresentedNumberScreened, or PresentedNumberUnscreened) will be sent marked as **PresentationRestricted** with **NULL** for the associated digits.

Following are examples and explanations of the output of common public-unknown-numbering commands.

The command **`list public-unknown-numbering`** operates as follows:

- **`list public-unknown-numbering start 4`**—displays the first entry starting with Ext Len of 4 followed by subsequent entries.
- **`list public-unknown-numbering start 4 count 50`**—displays the first 50 entries starting with Ext Len 4.
- **`list public-unknown-numbering`**—displays all entries.

The command **`change/display public-unknown-numbering`** operates as follows:

- **`change/display public-unknown-numbering 0`**—the screen displays the attendant entry first, followed by the subsequent entries.
- **`change/display public-unknown-numbering 4`**—the screen displays the first Ext Code of length 4 followed by the subsequent entries.
- **`change/display public-unknown-numbering 5 ext-digits 10010`**—the screen displays the first entry of Ext Code 10010 followed by the subsequent entries

- **`change/display public-unknown-numbering 5 ext-digits 10020`**—If 10020 has not been assigned, the screen displays the next entry following 10020 and subsequent entries.
- When used with the `Ext-Len` argument, for example, **`change public 5`**, the display starts with the first record found that matches the entered Extension Length, that is, 5. Then the system displays subsequent records.

Field descriptions for page 1

Figure 250: Numbering Public/Unknown screen

```
change public-unknown-numbering 5                                  Page 1 of X
                       NUMBERING - PUBLIC/UNKNOWN FORMAT

                                            Total
Ext Extension       Trk       CPN             CPN
Len Code            Grp(s)    Prefix          Len
 12 1234567890123   123456789 123456789012345 12

 5 4                777777                     10
 5 4                250       30379            10
 5 4                253       30379            10
 5 41               40        303222           11
 5 41               45                         5
 5 41               87        30323            10
 5 43               538                        7
 5 45               222                        7
 5 47               2222                       9
 5 61               45                         5
 5 406              250       30379            10
 5 406              253       30379            10
 5 418                        303538           11
 5 419                        222222222222222 15
 5 770                        970              8
```

CPN Prefix

Use this field to specify the number that is added to the beginning of the extension to form a Calling or Connected Number.

Valid entries	Usage
1 to 15 digits	Only digits are allowed in the CPN Prefix column. Leading spaces, or spaces in between the digits, are not allowed. ● If the length of the CPN Prefix matches the Total CPN Length, the extension number is not used to formulate the CPN number. ● If the number of digits in the CPN Prefix plus the extension length exceeds the administered Total CPN Length, excess leading digits of the extension are deleted when formulating the CPN number. ● If the number of CPN Prefix digits plus the extension length is less than the Total CPN Length, the entry is not allowed. ● If the Total CPN Length is 0, no calling party number information is provided to the called party and no connected party number information is provided to the calling party.
blank	If this field is blank, the extension is sent unchanged. This is useful in countries where the public network is able to insert the appropriate CPN Prefix to form an external DID number.

Ext Code

Allows for groups of extensions to be administered.

Note:
When **0** alone is entered, the **Ext Len** field must be 1 and the DDD number must be 10-digits.

Valid entries	Usage
leading extension digits (**0** to **9**)	The **Ext Code** can be up to 13 digits long depending on the **Ext Len** field entry. The entry cannot be greater than the **Ext Len** field entry. For example, in the case of a 4-digit **Ext Len** field entry, an Ext Code of **12** is the equivalent of all extensions of the screen 12xx, excluding any explicitly listed longer codes. If a code of 123 is also listed, the 12 code is equivalent of all extensions of the screen 12xx except extensions of the screen 123x. The coding precludes having to list all the applicable 12xx extensions.
attd	For attendant
blank	No extension code is entered.

Ext Len

Specifies the number of digits the extension can have. On page 1, this field displays the extension length entered as a qualifier on the command line (**`change public-unknown-numbering`** ***`n`***).

Valid entries	Usage
0 to **13** or blank	Corresponds to the extension lengths allowed by the dial plan.

Total CPN Len

Valid entries	Usage
0 to **15**	Enter the total number of digits to send.
Blank	This is the default. Leave blank when deleting an entry.

Trk Grp(s)

Communication Manager generates the station's identification number if there is an entry in the **Ext Code** field, and this field is administered with the trunk group number carrying the call.

Valid entries	Usage
1 to 7 digits	Enter the valid administered ISDN trunk-group number or a range of group numbers. For example, if trunk groups 10 through 24 use the same CPN Prefix, enter **10** to **24**.
blank	The identification numbers are not dependent on which trunk group the call is carried.

Field descriptions for page 2

Page 2 of the Numbering — Public/Unknown screen provides blank fields for new entries. See page 1 for field descriptions.

Off-PBX Telephone Configuration Set

See Configuration Set.

Off-PBX Telephone Feature-Name-Extensions

See Extensions to Call Which Activate Features by Name.

Off-PBX Telephone Mobile Feature Extension

Field descriptions for page 1

Figure 251: Off-PBX Telephone Mobile Feature Extension screen

```
change off-pbx-telephone mobile-feature-ext              Page 1 of x
                  OFF-PBX TELEPHONE MOBILE FEATURE EXTENSIONS

              Mobile Call (CTI) Extension:
```

Mobile Call (CTI) Extension

Valid entries	Usage
numeric value of an unassigned extension	A CTI call to this Mobile Feature Extension (MCE) creates an OPTIM call under CTI influence. A call to the MCE triggers an Off-PBX extend-call from a desk phone to its mapped cell phone number and to the destination. All calls made using the MCE appear to the destination as if they were dialed from the desk phone.

Off-PBX Telephone Station-Mapping

See **Stations With Off-PBX Telephone Integration**.

Optional Features

See **System Parameters Customer-Options (Optional Features)**.

Partition Routing Table

Use this table to identify routing for partition groups associated with an ARS analysis entry.

Field descriptions for page 1

Figure 252: Partition Routing Table screen

```
change partition route-table                                   Page 1 of X
                             Partition Routing Table

                                Routing Patterns
 Route
 Index        PGN 1   PGN 2   PGN 3   PGN 4   PGN 5   PGN 6   PGN 7   PGN 8
 -----        -----   -----   -----   -----   -----   -----   -----   -----
 196          ____    ____    ____    ____    ____    ____    ____    ____
 197          ____    ____    ____    ____    ____    ____    ____    ____
 198          ____    ____    ____    ____    ____    ____    ____    ____
 199          ____    ____    ____    ____    ____    ____    ____    ____
 200          ____    ____    ____    ____    ____    ____    ____    ____
 201          ____    ____    ____    ____    ____    ____    ____    ____
 202          ____    ____    ____    ____    ____    ____    ____    ____
 203          ____    ____    ____    ____    ____    ____    ____    ____
 204          ____    ____    ____    ____    ____    ____    ____    ____
 205          ____    ____    ____    ____    ____    ____    ____    ____
 206          ____    ____    ____    ____    ____    ____    ____    ____
 207          ____    ____    ____    ____    ____    ____    ____    ____
 208          ____    ____    ____    ____    ____    ____    ____    ____
 209          ____    ____    ____    ____    ____    ____    ____    ____
 210          ____    ____    ____    ____    ____    ____    ____    ____
```

PGN 1 (through PGN 8)

Enter the routing for each partition group associated with each route index number.

Valid entries	Usage
1 to **640**	Specifies the route pattern used to route the call
r1 to **r32**	Specifies the remote home numbering plan area table used to route the call
node	Designates node number routing
deny	Blocks the call

Personal CO Line Group

Use this screen to set up a personal central office line trunk group.

Field descriptions for page 1

Figure 253: Personal CO Line Group screen

```
add personal-CO-line                                                Page 1 of x
                            PERSONAL CO LINE GROUP

 Group Number: __                    Group Type:_____________  CDR Reports: _
   Group Name: _________________                                       TAC: ____
Security Code: ____               Coverage Path: ____     Data Restriction? _
                               Outgoing Display? _

TRUNK PARAMETERS
                 Trunk Type: _____________             Trunk Direction: ________
                 Trunk Port: _______            Disconnect Timing(msec): ____
                 Trunk Name: __________              Trunk Termination: ______
         Outgoing Dial Type: _________               Analog Loss Group: ___
                   Prefix-1? _                      Digital Loss Group: ___
Disconnect Supervision - In? _                         Call Still Held? _
 Answer Supervision Timeout: ___             Receive Answer Supervision? _
                 Trunk Gain: ____                              Country: __
          Charge Conversion: _____               DS1 Echo Cancellation: _
              Decimal Point: ______
            Currency Symbol: ___
                Charge Type: _______
```

The **Coverage Path** and **Security Code** fields are unique to this screen and are described below. For descriptions of other fields on this screen, see Trunk Group on page 971.

Coverage Path

Valid entries	Usage
1 to **9999**	Enter the number of the call coverage path you want to use for incoming calls.
t1 to **t999**	Enter the number of a time-of-day table.
blank	Assigning a coverage path is optional: leave this field blank if you do not want to assign one.

Security Code

Valid entries	Usage
3 to 8 digits	Enter a code that users must dial to retrieve voice messages and to use the Demand Print Message feature.
blank	Leave this field blank if you do not want to use a security code to control access.

Field descriptions for page 2

Figure 254: Personal CO Line Group screen

```
change personal-CO-line 1                                          Page 2 of 3
                              PERSONAL CO LINE GROUP

ASSIGNED MEMBERS (Stations with a button for this PCOL Group)

    Ext              Name
    1234567890123    123456789012345678901234567

 1: 1010             tst 4bri 1b0701
 2:
 3:
 4:
 5:
 6:
 7:
 8:
 9:
10:
11:
12:
13:
14:
15:
16:
```

Ext

This display-only field shows the extension of telephones that have a **CO Line** button.

Name

This display-only field shows the name assigned to telephones that have a **CO Line** button.

Field descriptions for page 3

Administrable timers for Personal CO Line groups appear on field descriptions for page 3. See Administrable Timers for Trunk Group screen for standard field definitions of the available timers.

Related topics

See Trunk Group on page 971 for definitions of all trunk group fields that are *not* unique to the PCOL screen.

Pickup Group

This screen implements call pickup groups with up to 50 extensions per group. A pickup group is a group of users authorized to answer calls to a telephone extension within that group of users. A telephone extension can belong to only one pickup group.

Field descriptions for pages 1 and 2

Figure 255: Pickup Group screen

```
add pickup-group 1                                                  Page 1 of x
                                   PICKUP GROUP

           Group Number: 1
             Group Name: Test Group
GROUP MEMBER ASSIGNMENTS

    Ext              Name
 1:1234567890123   Mohandas Karamchand Gandhi
 2:
 3:
 4:
 5:
 6:
 7:
 8:
 9:
10:
11:
12:
13:
```

Ext

Enter the extension assigned to a station.

Valid entries	Usage
Valid extension number.	A VDN cannot be assigned to a Call Pickup group.

Extended Group Number

This field appears only when the **Group Call Pickup** field is set to **flexible** on the Feature-Related System Parameters screen. The extended group is a collection of pickup groups that can answer calls from other pickup groups in the same extended group.

Valid entries	Usage
1 to **100** (DEFINITY CSI)	Enter the extended group number or blank.

Group Number

Valid entries	Usage
Pickup Group number	This display-only field appears when the screen is accessed using an administration command such as **add** or **change**.

Name

This display-only field shows the name assigned to the above extension number when the users and their associated extensions were administered.

Policy Routing Table

This feature allows you to distribute calls among a set of call centers based on specified percent allocation. Various types of incoming calls that arrive at a particular VDN can be directed to a Policy Routing Table (PRT) instead of to a vector. The PRT then distributes the calls to the administered Route-to VDNs based on the specified percent allocation targets.

Use this screen to implement and monitor percentage allocation routing by assigning destination routes and target percentages.

Note:
For a detailed description of Policy Routing Table screen and its fields, see *Avaya Aura™ Call Center 5.2 Automatic Call Distribution (ACD) Reference*, 07-602568.

Field descriptions

Figure 256: Policy Routing Table screen

```
POLICY ROUTING TABLE
 Number: 1957  Name: % distribution      Type: percentage   Period: max count


                                                  Target Actual Call
 Index Route-to VDN  VDN NAME                     %      %      Counts
   1   2220071       Gizmo Support                25     27.2      3
   2   2221501       Ultra Support                 5      9.0      1
   3   2220601       Customer Service South       35     27.2*     3
   4   2220511       Outsourcer Charlie           10      9.0      1
   5   2220501       Survey after service         10      9.0      1
   6   2220072       Outsourcer International     15     18.1      2
   7
   8
   9
  10
  11
  12
  13
  14
  15
                                           Totals 100              11
```

Number

Displays the table number that you entered on the command line.

Name

Enter a string of up to 15 characters as the name of the PRT table. Any alpha-numeric character is valid. Default is blank.

Type

Specify the type of algorithm the PRT table supports. The only valid entry in this field is percentage, as only percentage allocation is supported as of now.

Period

Specify the period for resetting the call counts and actual percentages. Following are the valid entries for this field:

Valid entries	Usage
100_count (default)	Resets the call counts (and displayed %) when total calls for the PRT reach 100, which is when the total calls match the target routing pattern percentages. This ensures that the routing points have equal distribution of calls all the time.
max_count	Call counts are maintained until calls delivered to at least one of the VDNs exceed 65,400. At that point calls are continued to be distributed over the VDNs but the call counts are reset when the actual percentages equal the targets for all of the VDNs at the same time.
Half-hour	Resets the call counts at the top of the hour and at the 30 minute point.
hour	Resets the call counts at the top of the hour.
daily	Resets the call counts at midnight, every night.
weekly	Resets the call counts at midnight on Saturday.

Index

Displays the sequential number of the row. You can enter upto 15 Route-to VDN entries in a PRT table.

Route-to VDN

Enter up to 13-digit long valid and assigned VDN extension to which calls are to be routed. Default is blank.

VDN Name

Displays the assigned name of the VDN specified in the Route-to VDN field or "name not assigned" if the VDN name is not assigned yet. The name must be assigned or changed on the VDN form.

Target %

Specifies the target percent of total calls to be routed to a VDN. Valid entries are 0 to 100. Use whole numbers only, no decimal fractions.

Actual %

Specifies the actual percentage of total calls routed to a VDN. Actual % is calculated to 6 decimal places, but only the first decimal place is displayed.

Call Counts

This field displays the current number of calls routed to a VDN.

Totals

This field displays of values in the Target % and Call Counts for all the assigned VDNs in the policy routing table. The total for Target % is always 100 for form submittal.

Precedence Routing Digit Analysis Table

Communication Manager compares dialed numbers with the dialed strings in this table and determines the route pattern of an outgoing Multiple Level Precedence and Preemption (MLPP) call.

Field descriptions for page 1

Figure 257: Precedence Routing Digit Analysis Table screen

```
change precedence-routing analysis nn                         Page 1 of x

                      PRECEDENCE ROUTING DIGIT ANALYSIS TABLE
                                                            Percent Full: 22

          Dialed            Total     Route     Preempt
          String           Min  Max  Pattern   Method
    002383                 9    9     36       group
    002385                 9    9     35       group
    002388                 9    9     86       group
    003032383              12   12    36       group
    003032388              12   12    86       group
    003033383              12   12    34       group
    003033388              12   12    84       group
    003034383              12   12    32       group
    003034388              12   12    82       group
    003035383              12   12    30       group
    003035388              12   12    80       group
    003383                 9    9     34       group
    003385                 9    9     33       group
    003388                 9    9     84       group
    004383                 9    9     32       group
```

Dialed String

User-dialed numbers are matched to the dialed string entry that most closely matches the dialed number.

Valid entries	Usage
0 to **9**	Enter up to 18 digits that the call-processing server analyzes.
***, x, X**	wildcard characters

Max

Valid entries	Usage
Between Min and **28**	Enter the maximum number of user-dialed digits the system collects to match to the dialed string.

Min

Valid entries	Usage
Between **1** and Max	Enter the minimum number of user-dialed digits the system collects to match to the dialed string.

Percent Full

Valid entries	Usage
0 to **100**	Display only. Shows the percent of the Precedence Routing Digit Analysis Table that is currently in use.

Preempt Method

Enter the preemption method you want the server running Communication Manager to use for this dialed string.

Valid entries	Usage
group	The system checks the first trunk group in the route pattern to determine if any trunks are idle. If the system finds an idle trunk, the system connects the call. This is the default.
route	The system checks each trunk group in the route pattern to determine if any trunks are idle. If the system finds an idle trunk, the call is connected.

Route Pattern

Enter the route number you want the server running Communication Manager to use for this dialed string.

Valid entries	Usage
1 to **999**	Specifies the route pattern used to route the call.
deny	Blocks the call

Precedence Routing Digit Conversion Table

Use the Precedence Routing Digit Conversion screen to assign the Precedence Routing digit conversion. Digit conversion takes digits dialed on incoming calls and converts the digits to local telephone numbers, usually extension numbers.

Field descriptions for page 1

Figure 258: Precedence Routing Digit Conversion Table screen

```
change precedence-routing digit-conversion n           Page   1 of   x

                   PRECEDENCE ROUTING DIGIT CONVERSION TABLE
                                                              Percent Full: 11

  Matching Pattern     Min    Max    Del   Replacement String   Net   Conv

  x2386                8      8      4                          ext    n
  x3032386             11     11     7                          ext    n
  x3033386             11     11     7                          ext    n
  x3034386             11     11     7                          ext    n
  x3035386             11     11     7                          ext    n
  x3386                8      8      4                          ext    n
  x4386                8      8      4                          ext    n
  x5386                8      8      4                          ext    n
  x6                   5      5      1                          ext    n
  xx2386               9      9      5                          ext    n
  xx3032386            12     12     8                          ext    n
  xx3033386            12     12     8                          ext    n
  xx3034386            12     12     8                          ext    n
  xx3035386            12     12     8                          ext    n
```

Conv

Valid entries	Usage
y	Enter **y** to allow more conversions.
n	Enter **n** to prevent further conversions.

Del

Valid entries	Usage
0 to **Min**	Enter the number of leading digits to delete.

Matching Pattern

Valid entries	Usage
0 to **9**, *****, **x**, or **X**	Enter the precedence digit and the address digits. For WNDP dialing, you must also enter the route code.

CAUTION:
The **Matching Pattern** field requires the following format for routing DSN numbers: For precedence dialing (non-WNDP dialing), enter the precedence digit (typically 0-4) and the address digits. For WNDP dialing, enter the precedence digit (typically 0-4), the route code, and the address digits. An **x** in the digit string is a wildcard that matches on any single digit.

Max

Valid entries	Usage
Between **Min** and **28**	Enter the maximum number of user-dialed digits the system collects to match to the dialed string.

Min

Valid entries	Usage
Between **1** and **Max**	Enter the minimum number of user-dialed digits the system collects to match to the dialed string.

Net

Valid entries	Usage
ext	Extension. Uses ARS tables or AAR tables to route the call.
pre	Precedence routing. Uses the Precedence Analysis Tables to route the call.

Replacement String

Valid entries	Usage
0 to **9**, *, #, or blank, up to 18 characters	Enter the digits that replace the deleted portion of the dialed number. Leave this field blank to simply delete the digits. The **#** sign, if present in the string, should be the last character in the string. This signifies the end of the modified digit string.

Route Pattern

Valid entries	Usage
1 to **999**	Specifies the route pattern used to route the call.
deny	Blocks the call.

PRI Endpoint

This screen administers PRI Endpoints for the Wideband Switching feature.

Note:
A PRI Endpoint with a width greater than 1 can be administered only if the **Wideband Switching** feature has been enabled on the System Parameters Customer-Options (Optional Features) screen.

A PRI Endpoint is an endpoint application connected to line-side ISDN-PRI facilities and has standard ISDN-PRI signaling interfaces to the system. For information on endpoint applications connected to line-side non-ISDN T1 or E1 facilities, see Access Endpoint on page 24 in this module.

A PRI Endpoint is defined as 1 to 31 adjacent DS0s/B-channels, addressable via a single extension, and signaled via a D-channel (Signaling Group) over a standard T1 or E1 ISDN-PRI interface.

Field descriptions for page 1

Figure 259: PRI Endpoint screen

```
add pri-endpoint next                                            Page   1 of  x
                                  PRI ENDPOINT

                     Extension: 300
                          Name: 27 character PRI Endpoint 1
               (Starting) Port:                            Width: 1
  Originating Auto Restoration? n                Signaling Group:
                           COR: 1                            COS: 1
                            TN: 1             Simultaneous Calls? n
           Maintenance Tests? y

                             WIDEBAND SUPPORT OPTIONS

                                H0? n
                               H11? n
                               H12? n
                             NXDS0? y   Contiguous? n
```

COR

Valid entries	Usage
0 to **995**	Enter class of restriction (COR) to determine calling and called party privileges

COS

Valid entries	Usage
0 to **15**	Enter the Class of Service (COS) to determine the features that can be activated by, or on behalf of, the endpoint.

Extension

A display-only field when the screen is accessed using an administration command such as **change** or **display**.

Valid entries	Usage
Extension	This is the extension number used to access the PRI endpoint. Enter a valid unassigned extension number when completing a paper screen.

Maintenance Tests

Valid entries	Usage
y/n	Enter **y** to run hourly maintenance tests on this PRI Endpoint.

Name

Identifies the endpoint.

Valid entries	Usage
Up to 27 alphanumeric characters.	Enter a name for the endpoint.

Originating Auto Restoration

Valid entries	Usage
y/n	Enter **y** to automatically restore calls originating from this PRI Endpoint (while maintaining endpoint call status) in the case of network failure if the call is over SDDN network facilities.

Signaling Group

Valid entries	Usage
1 to **416**, blank (S8300/S87XX Servers) **1** to **110** or blank (DEFINITY CSI)	Enter the D-channel or D-channel pair that provides the signaling information for the set of B-channels that make up the PRI Endpoint.

Simultaneous Calls

Valid entries	Usage
y/n	Enter **y** to specify that multiple simultaneous calls can be placed to/from the PRI Endpoint.

(Starting) Port

Enter the seven-character starting port of the PRI Endpoint. Enter the necessary characters.

Valid entries	Usage
01 to **03** (DEFINITY CSI) or **1** to **64** (S87XX/S8300 Servers)	First and second characters are the cabinet number
A to **E**	Third character is the carrier
0 to **20**	Fourth and fifth character are the slot number
01 to **04** (Analog TIE trunks) **01** to **31**	Six and seventh characters are the circuit number

TN

Valid entries	Usage
1 to **20** (DEFINITY CSI) **1** to **100** (S8300/S87XX Servers)	Enter the Tenant Partition number.

Width

Enter the number of adjacent DS0 ports beginning with the specified Starting Port, that make up the PRI Endpoint.This field cannot be blank.

Valid entries	Usage
1 to **31**	A width of **6** defines a PRI Endpoint that can support data rates up to 384 Kbps.

WIDEBAND SUPPORT OPTIONS

Contiguous

Specifies whether to hunt contiguous NXDS0 channels. This field only appears if **y** is entered in **NXDS0**. The hunt algorithm to satisfy an NXDS0 call is as follows:

Valid entries	Usage
y/n	Enter **y** to specify the "floating" scheme. NXDS0 calls are placed on a contiguous group of B-channels large enough to satisfy the requested bandwidth without constraint on the starting channel (no fixed starting point trunk).
n	Enter **n** to specify the "flexible" scheme. NXDS0 calls are placed on any set of B-channels on the same facility as long as the requested bandwidth is satisfied. There are no constraints, such as contiguity of B-channels or fixed starting points.

H0

Valid entries	Usage
y/n	Enter **y** to specify the ISDN information transfer rate for 384 Kbps of data, which is comprised of six B-channels. When a PRI Endpoint is administered to support H0, the hunt algorithm to satisfy a call requiring 384 Kbps of bandwidth uses a fixed allocation scheme.

H11

Valid entries	Usage
y/n	Enter **y** to specify the ISDN information transfer rate for 1536 Kbps of data, which is comprised of 24 B-channels. When a PRI Endpoint is administered to support H11, the hunt algorithm to satisfy a call requiring 1536 Kbps of bandwidth uses a fixed allocation scheme.

H12

Valid entries	Usage
y/n	Enter **y** to specify the ISDN information transfer rate for 1920 Kbps data, which includes 30 B-channels. When a PE is administered to support H12, the hunt algorithm to satisfy a call requiring 1920 Kbps of bandwidth uses a fixed allocation scheme.

NXDS0

Valid entries	Usage
y/n	Enter **y** to specify the NXDS0 multi-rate service.

Processor Channel Assignment

Use this screen to assign each local processor channel to an interface link channel, and to define the information associated with each processor channel on an Ethernet link.

Note:
You cannot remove a service from this screen if that service has overrides defined on the **Survivable Processor** screen.

Field descriptions for page 1

Figure 260: Processor Channel Assignment screen

```
change communication-interface processor-channel              page 1 of x

                                                                    page 2 of x

                     SURVIVABLE PROCESSOR - PROCESSOR CHANNELS

Proc          Gtwy                Interface          Destination       Session    Mach ID
Chan  Enable   To   Appl.   Mode  Link/Chan         Node       Port    Local/Remote
 1      y           mis      s     9   5001       CMS_hogan      0        1      1      1
 2      y           ccr      s     10  5002       ccrhost1       0        2      2      1
 3
 4
 5
 6
 7
 8
 9
 10
 11
 12
 13
 14
 15
 16
```

Appl

Use this field to specify the server application type or adjunct connection used on this channel.

Valid entries	Usage
audix	Voice Messaging
ccr	Contact Center Reporting
dcs	Distributed Communication System
echo	
fp-mwi	ISDN Feature Plus Message Waiting Indication. This channel passes message waiting light information for subscribers on the messaging system, from a messaging adjunct on a main switch for a phone on a satellite switch. The terminating location (far end) of this channel must be a Communication Manager system compatible with ISDN Feature Plus proprietary protocol.

Valid entries	Usage
gateway	Supports an X.25 connected AUDIX connected to an ISDN DCS network.
gteway-tcp	Supports a TCP-connected AUDIX connected to an ISDN DCS network.
mis	Management Information System, otherwise known as CMS (Communication Management System)
msaaawl **msaclk** **msahlwc** **msallwc** **msamcs**	All **msa** entries refer to an obsolete product. The system does not accept these entries.
qsig-mwi	QSIG Message Waiting Indication. Used with a QSIG-based interface to a messaging system, this channel passes message waiting light information for subscribers on the messaging system.

Destination Node

Use this field to identify the server or adjunct at the far end of this link.

Valid entries	Usage
valid administered node name	Enter an adjunct name, server name, far end IP address, node name, or leave blank for services local to this Avaya S8XXX Server. For ppp connections, match the **Destination Node Name** on the ppp Data Module screen.

Destination Port

Use this field to identify the port number of the destination.

Valid entries	Usage
0, **5000** to **64500**	Enter the number of the destination port. An entry of **0** means any port can be used.

Enable

Use this field to enable or disable this processor channel.

Valid entries	Usage
y/n	Enter **y** to enable this processor channel on the main server. Enter **n** to disable this processor channel.

Gtwy to

This field identifies which processor channel the specified processor channel is serving as a gateway to.

Valid entries	Usage
1 to system max or blank	Enter the number of the processor channel.

Interface Channel

This field identifies the channel number or the TCP/IP listen port channel to carry this processor (virtual) channel. For TCP/IP, interface channel numbers are in the range **5000** to **64500**. The value **5001** is recommended for CMS, and **5003** is recommended for DCS.

Valid entries	Usage
0, **5000** to **64500**	For **ethernet** or **ppp**. The channel number **0** means any port can be used.

Interface Link

This field identifies the physical link carrying this processor (virtual) channel.

Valid entries	Usage
1 to **254**	Enter the physical link carrying this processor (virtual) channel.
p (processor)	Enter **p** to use the Communication Manager's Processor Ethernet interface for adjunct connectivity.
blank	

Mach ID

Valid entries	Usage
1 to **63** for MWI, **1** to **63** for DCS, **1** to **99** for AUDIX, or blank	Enter the destination server ID defined on the dial plan of the destination server.

Mode

Valid entries	Usage
c(lient) **s(erver)** blank	Indicate whether the IP session is passive (client) or active (server). This field must be blank if the interface link is **procr-intf**. This field cannot be blank if the type of interface link is **ethernet** or **ppp**.

Proc Chan

This display-only field shows the number assigned to each processor channel you administer. Range is from **1** to **384**.

Session - Local/Remote

Local and Remote Session numbers must be consistent between endpoints.

Valid entries	Usage
1 to **256** (si) **1** to **384** (r) or blank	For each connection, the Local Session number on this Avaya S8XXX Server must equal the Remote Session number on the remote server and vice versa. It is allowed, and sometimes convenient, to use the same number for the Local and Remote Session numbers for two or more connections.

QSIG to DCS TSC Gateway

The QSIG to DCS TSC Gateway screen determines when and how to convert messages from a QSIG NCA-TSC to an administered AUDIX NCA-TSC. This screen maps the QSIG subscriber number to the appropriate AUDIX signaling group and TSC index.

This screen only appears if the **Interworking with DCS** field is **y** on the System Parameters Customer-Options (Optional Features) screen.

Field descriptions for page 1

Figure 261: QSIG to DCS TSC Gateway screen

```
change isdn qsig-dcs-tsc-gateway                                  Page 1 of x

                              QSIG TO DCS TSC GATEWAY

    Subscriber          Sig   TSC           Subscriber          Sig       TSC
     Number             GRP  Index           Number             GRP      Index
  __________________    ___   ___        __________________     ___        ___
  __________________    ___   ___        __________________     ___        ___
  __________________    ___   ___        __________________     ___        ___
  __________________    ___   ___        __________________     ___        ___
  __________________    ___   ___        __________________     ___        ___
  __________________    ___   ___        __________________     ___        ___
  __________________    ___   ___        __________________     ___        ___
  __________________    ___   ___        __________________     ___        ___
```

Sig Grp

Valid entries	Usage
1 to **110**	Enter the assigned signaling group number for DEFINITY CSI.
1 to **650**	Enter the assigned signaling group number for S8300/S87XX Servers.

Subscriber Number

You can enter up to 28 subscriber numbers.

Valid entries	Usage
0 to **9**, *****, ‘**x**’, ‘**X**’	Enter a subscriber number up to 20 characters in length. You can use wildcards ‘**x**’ and ‘**X**’ to enter subscriber numbers.

TSC Index

You must complete the **TSC Index** field for each machine ID.

Valid entries	Usage
1 to **64**	Enter the assigned signaling group number for **qsig-mwi** application type on the Signaling Group screen.

Valid entries	Usage
0 to **9**, *****, **#**	Enter up to 4-digit access code.

Reason Code Names

Use the Reason Code Names screen to assign names to reason codes. You can assign a different name to each reason code for Aux Work and for Logout. Pages 2 and 3 appear when, on the **Feature-Related System Parameters** screen, the **Two-Digit Aux Work Reason Codes** field is **y**. These additional pages accommodate names for Aux Work Reason Codes 10 to 99. Note that **Logout** reason codes can only be in the range of 0 to 9, even if the Two-Digit Aux Work Reason Codes option is active.

Figure 262: Reason Code Names screen - page 1

```
change reason-code-names                                        Page   1 of   x

                                REASON CODE NAMES

                             Aux Work/            Logout
                          Interruptible?

        Reason Code 1:                 /n
        Reason Code 2:                 /n
        Reason Code 3:                 /n
        Reason Code 4:                 /n
        Reason Code 5:                 /n
        Reason Code 6:                 /n
        Reason Code 7:                 /n
        Reason Code 8:                 /n
        Reason Code 9:                 /n

  Default Reason Code:
```

Figure 263: Reason Code Names screen - page 2

```
change reason-code-names                                  Page   2 of   x

                        REASON CODE NAMES - AUX WORK

   10:                       28:                       46:
   11:                       29:                       47:
   12:                       30:                       48:
   13:                       31:                       49:
   14:                       32:                       50:
   15:                       33:                       51:
   16:                       34:                       52:
   17:                       35:                       53:
   18:                       36:                       54:
   19:                       37:                       55:
   20:                       38:                       56:
   21:                       39:                       57:
   22:                       40:                       58:
   23:                       41:                       59:
   24:                       42:                       60:
   25:                       43:                       61:
   26:                       44:                       62:
   27:                       45:                       63:
```

Figure 264: Reason Code Names screen - page 3

```
change reason-code-names                                    Page   3 of   x

                        REASON CODE NAMES - AUX WORK

   64:                      82:
   65:                      83:
   66:                      84:
   67:                      85:
   68:                      86:
   69:                      87:
   70:                      88:
   71:                      89:
   72:                      90:
   73:                      91:
   74:                      92:
   75:                      93:
   76:                      94:
   77:                      95:
   78:                      96:
   79:                      97:
   80:                      98:
   81:                      99:
```

Field descriptions for page 1

Aux Work

Valid entries	Usage
up to 16 alphanumeric characters	Enter the name to be associated with a reason code when the agent uses the reason code to enter Aux Work mode. Default is blank.

Default Reason Code

Use this field to enter a name for the default reason codes. You can enter a separate name for the Aux Work Reason Code of 0 and for the Logout Reason Code of 0. If an agent changes to Aux Work mode and the Aux Work Reason Code Type is set to none, the agent is put into Aux Work mode with the default Aux Work reason code, even if you have administered a different reason code for the Aux button. If an agent logs out when the Logout Reason Code Type is set to none, the agent is logged out with the default Logout reason code.

Valid entries	Usage
up to 16 alphanumeric characters	Enter a name for the default reason code. Default is blank.

Logout

Valid entries	Usage
up to 16 alphanumeric characters	Enter the name to be associated with a reason code when the agent uses the reason code to log out. Default is blank. Note that Logout reason codes can only be in the range of 0 to 9, even if the Two-Digit Aux Work Reason Codes option is active.

Interruptible?

For each reason code, enter /n or /y to specify whether or not the reason code is interruptible or not. /n signifies that the reason code is not interruptible and /y signifies that the reason code is interruptible.

Note:
The Default Reason Code cannot be made interruptible, so there is no Interruptible qualifier for that field. There are two more fields that cannot be made interruptible, **Auto-answer IP Failure Aux Work Reason Code** and **Maximum Agent Occupancy Aux Work Reason Code**.

Remote Access

The Remote Access screen is used to implement the Remote Access feature. Remote Access permits a caller located outside the system to access the system through the public or private network and then use the features and services of the system.

Remote Access users can dial into the system using central office (CO), Foreign Exchange (FX), Wide Area Telecommunications trunks (WATS), and Integrated Services Digital Network Primary Rate Interface (ISDN-PRI) trunks. In addition, a dedicated Remote Access Direct Inward Dialing number can be provided.

SECURITY ALERT:
Avaya designed the Remote Access feature incorporated in this product that, when properly administered by the customer, enables the customer to minimize the ability of unauthorized persons to gain access to the network. It is the customer's responsibility to take the appropriate steps to properly implement the features, evaluate and administer the various restriction levels, protect access codes and distribute them only to individuals who have been advised of the sensitive nature of the access information. Each authorized user should be instructed concerning the proper use and handling of access codes.

In rare instances, unauthorized individuals make connections to the telecommunications network through use of remote access features. In such an event, applicable tariffs require the customer pay all network charges for traffic. Avaya cannot be responsible for such charges, and will not make any allowance or give any credit for charges that result from unauthorized access.

To ensure the security of your system, consider the following:

- Make all remote access facilities unlisted directory telephone numbers.
- Require users to enter a Barrier Code of at least seven random digits AND an Authorization Code of at least 13 random digits to make network calls.
- Make Authorization Codes nonconsecutive (random) and change them, at least, quarterly.
- Deactivate Authorization Codes immediately if the user leaves the company or changes assignments.
- Assign the minimum level of calling permissions required to each Authorization Code.
- Block off-hours and weekend remote access calling, when possible. Use Alternative Facility Restriction Levels, if available.
- Use a voice recording, warble tone, or no tone and avoid use of a dial tone as a prompt when the remote access unit answers.
- Assign the lowest possible FRL to only allow calls internal to Communication Manager.

As an additional step to ensure system security, you can permanently disable the Remote Access feature if you do not intend to use it now or in the future. If you do decide to permanently disable the feature, it requires Avaya Services intervention to activate the feature again.

CAUTION:
Your attempt to disable the Remote Access feature will be lost if the server running Communication Manager is rebooted without saving translations. Therefore, execute a **`save translation`** command after permanently disabling the Remote Access feature.

Field descriptions for page 1

Figure 265: Remote Access screen

```
change remote-access

                                REMOTE ACCESS
   Remote Access Extension: ________            Barrier Code Length:____
Authorization Code Required? y            Remote Access Dial Tone: n

     Barrier      COR  TN  COS   Expiration    No. of     Calls
      Code                          Date        Calls      Used
   1:________     1__  1_  1__   __/__/__      ______     ______
   2:________     1__  1_  1__   __/__/__      ______     ______
   3:________     1__  1_  1__   __/__/__      ______     ______
   4:________     1__  1_  1__   __/__/__      ______     ______
   5:________     1__  1_  1__   __/__/__      ______     ______
   6:________     1__  1_  1__   __/__/__      ______     ______
   7:________     1__  1_  1__   __/__/__      ______     ______
   8:________     1__  1_  1__   __/__/__      ______     ______
   9:________     1__  1_  1__   __/__/__      ______     ______
  10:_______      1__  1_  1__   __/__/__      ______     ______

Permanently Disable? __ Disable Following A Security Violation? y
(NOTE:  You must logoff to effect permanent disabling of Remote Access)
```

Authorization Code Required

When you use an authorization code in conjunction with a barrier codes it increases the security of the Remote Access feature.

Valid entries	Usage
y/n	Enter **y** to require an authorization code be dialed by Remote Access users to access the system's Remote Access facilities.

Barrier Code

You must assign a barrier code that conforms to the number entered in the **Barrier Code Length** field. You can enter up to 10 barrier codes per system. Duplicate entries are not allowed. You must keep your own records regarding the distribution of these barrier codes to your personnel.

Note:
After you make an entry in the **Barrier Code** field, additional fields in the same row (**COR**, **TN**, **COS**, **Expiration Date**, and **No. of Calls**) become editable.

Valid entries	Usage
0 to **9** or blank	Enter a 4- to 7-digit number in any combination of digits.
none	Must be specified in the first **Barrier Code** field, if the **Barrier Code Length** field is blank.

Barrier Code Length

Assign a barrier code length of **7** to provide maximum security.

Valid entries	Usage
4 to **7** or blank	Enter a number to indicate the length of the barrier code.

Calls Used

This display-only field shows the number of calls placed using the corresponding barrier code. This field is incremented each time a barrier code is successfully used to access the Remote Access feature. A usage that exceeds the expected rate indicates improper use.

COR

This field changes from display-only to editable after you make an entry in the **Barrier Code** field. Assign the most restrictive class of restriction (COR), that provides only the level of service required, to provide the maximum security.

Valid entries	Usage
0 to **995**	Enter the COR number associated with the barrier code that defines the call restriction features.

COS

This field changes from display-only to editable after you make an entry in the **Barrier Code** field. Assign the most restrictive class of service (COS), that provides only the level of service required to provide the maximum security.

Valid entries	Usage
0 to **15**	Enter the COS number, associated with the barrier code, that defines access permissions for Call Processing features.

Disable Following a Security Violation

This field appears on the screen when the **SVN Authorization Code Violation Notification Enabled** field on the Security-Related System Parameters screen is set to **y**.

Valid entries	Usage
y/n	Enter **y** to disable the remote access feature following detection of a remote access security violation. The system administrator can re-enable Remote Access using the `enable remote-access` command.

Expiration Date

This field changes from display-only to editable after you make an entry in the **Barrier Code** field. Assign an expiration date based on the expected length of time the barrier code will be needed. If it is expected the barrier code is to be used for a 2-week period, assign a date two weeks from the current date. If the Expiration Date is assigned, a warning message is displayed on the System Copyright screen seven days prior to the expiration date. The system administer can modify the expiration date to extend the time interval if needed.

Valid entries	Usage
A date greater than the current date or blank	Enter the date you want the barrier code to expire.

No. of Calls

This field changes from display-only to editable after you make an entry in the **Barrier Code** field. The **Expiration Date** and **No. of Calls** fields can be used independently or in conjunction to provide the maximum security. If both the **Expiration Date** and **No. of Calls** fields are assigned, the corresponding barrier code expires when the first of these criteria is satisfied.

Valid entries	Usage
1 to **9999** or blank	Enter the number of Remote Access calls that can be placed using the associated barrier code.

Permanently Disable

Reactivation of remote access to the interface requires Avaya Services intervention.

Valid entries	Usage
y/n	Enter **y** to permanently block remote access to the administration interface.

Remote Access Dial Tone

Set this field to **n** for maximum security. This field appears when the **Authorization Code Required** field is set to **y**.

Valid entries	Usage
y/n	Enter **n** so that there is no Remote Access Dial Tone prompt.

Remote Access Extension

The remote access extension is used as if it was a DID extension. Only one DID extension can be assigned as the remote access extension. Calls to that number are treated the same as calls on the remote access trunk.

When a trunk group is dedicated to Remote Access, the remote access extension number is administered on the trunk group's incoming destination field.

Valid entries	Usage
Extension number	Enter the extension number for Remote Access associated with each trunk that supports the Remote Access feature. You cannot assign a Vector Directory Number (VDN) extension as the remote access extension. Can be blank if no barrier codes.

TN

This field changes from display-only to editable after you make an entry in the **Barrier Code** field.

Valid entries	Usage
1 to **20** (DEFINITY CSI) **1** to **100** (S8300/S87XX Servers))	Enter the Tenant Partition number.

Related Topics

See Remote Access in *Avaya Aura™ Communication Manager Feature Description and Implementation,* 555-245-205, for more information.

Remote Call Coverage Table

The Remote Call Coverage Table allows you to provide automatic redirection of certain calls to alternate non-local answering positions in a coverage path.

Non-local numbers can be any ARS or AAR number, any number on the public network, any international number, or a UDP/DCS extension up to 16 digits or blank, which includes any ARS/AAR facility access code, any trunk dial access code (TAC), long distance dialing code, or international dial code. Up to 999 remote call coverage points can be entered on the multiple pages of this screen.

Field descriptions for page 1

Figure 266: Remote Call Coverage Table screen

```
change coverage remote

                          REMOTE CALL COVERAGE TABLE
                          ENTRIES FROM 1  TO 1000

01: ________________      16: ________________      31: ________________
02: ________________      17: ________________      32: ________________
03: ________________      18: ________________      33: ________________
04: ________________      19: ________________      34: ________________
05: ________________      20: ________________      35: ________________
06: ________________      21: ________________      36: ________________
07: ________________      22: ________________      37: ________________
08: ________________      23: ________________      38: ________________
09: ________________      24: ________________      39: ________________
10: ________________      25: ________________      40: ________________
11: ________________      26: ________________      41: ________________
12: ________________      27: ________________      42: ________________
13: ________________      28: ________________      43: ________________
14: ________________      29: ________________      44: ________________
15: ________________      30: ________________      45: ________________
```

01 to 1000

Valid entries	Usage
0 to **9** ***** (DTMF digit asterisk) **#** (DTMF digit pound) **L** (use coverage point only when in LSP or ESS mode) **D** (represents the called extension digits) **’** (pause for 1.5 seconds) **’** (pause for 1.5 seconds) **%** (rest of digits are for end-to end signaling) blank (**L**, **D**, **’**,**’**, and **%** use 2 places)	Enter the destination coverage point up to 16 digits.

Remote Office

This screen supports an arrangement whereby a user can set up a remote office without having an on-premises physical desk-set. An R300 is issued to connect remote DCP and analog telephones, IP telephones, and H.323 trunks to the Communication Manager server via IP.

Field descriptions for page 1

Figure 267: Remote Office screen

```
change remote office x                                          Page 1 of x

                                    REMOTE OFFICE

     Node Name: _______________
Network Region: ___
      Location: __
     Site Data: ______________________________
                ______________________________
                ______________________________
```

Location

Valid entries	Usage
1 to **64**	Specify the location (comprised of the associated time zone and the appropriate numbering plan).

Network Region

Valid entries	Usage
1 to **250** or blank	Specify the network region to be assigned to all stations supported on this remote office.

Node Name

Valid entries	Usage
character string	Specify the node name of the remote office.

Site Data

Valid entries	Usage
30 characters or blank	Any desired information.

RHNPA Table

The RHNPA Table defines route patterns for specific 3-digit codes, usually direct distance dialing (DDD) prefix numbers. The appearance of the screen is different slightly depending on the type of Avaya S8XXX Server.

Figure 268: RHNPA Table screen

```
change rhnpa                                                    Page 1 of X
                                 RHNPA TABLE: __
                                 CODES: 000-999

                              Pattern Choices
           1: ___    3: ___    5: ___    7: ___    9: ___    11: ___
           2: ___    4: ___    6: ___    8: ___   10: ___    12: ___

       Code - Pattern Choice Assignments (from 1-12 above)
00: 1__ 10: 1__ 20: 1__ 30: 1__ 40: 1__ 50: 1__ 60: 1__ 70: 1__ 80: 1__ 90: 1__
01: 1__ 11: 1__ 21: 1__ 31: 1__ 41: 1__ 51: 1__ 61: 1__ 71: 1__ 81: 1__ 91: 1__
02: 1__ 12: 1__ 22: 1__ 32: 1__ 42: 1__ 52: 1__ 62: 1__ 72: 1__ 82: 1__ 92: 1__
03: 1__ 13: 1__ 23: 1__ 33: 1__ 43: 1__ 53: 1__ 63: 1__ 73: 1__ 83: 1__ 93: 1__
04: 1__ 14: 1__ 24: 1__ 34: 1__ 44: 1__ 54: 1__ 64: 1__ 74: 1__ 84: 1__ 94: 1__
05: 1__ 15: 1__ 25: 1__ 35: 1__ 45: 1__ 55: 1__ 65: 1__ 75: 1__ 85: 1__ 95: 1__
06: 1__ 16: 1__ 26: 1__ 36: 1__ 46: 1__ 56: 1__ 66: 1__ 76: 1__ 86: 1__ 96: 1__
07: 1__ 17: 1__ 27: 1__ 37: 1__ 47: 1__ 57: 1__ 67: 1__ 77: 1__ 87: 1__ 97: 1__
08: 1__ 18: 1__ 28: 1__ 38: 1__ 48: 1__ 58: 1__ 68: 1__ 78: 1__ 88: 1__ 98: 1__
09: 1__ 19: 1__ 29: 1__ 39: 1__ 49: 1__ 59: 1__ 69: 1__ 79: 1__ 89: 1__ 99: 1__
```

Field descriptions for page 1

CODES

Display-only field showing the desired 100-block, for example 000 through 099 or 900 through 999 based upon the `change rhnpa` command. A separate screen displays for each 100-block.

Code-Pattern Choice Assignments

Valid entries	Usage
1 to **24**	For S87XX Series IP-PNC.

Pattern Choices

There are 12 pattern choices for DEFINITY CSI; there are 24 pattern choices for the S8300/S87XX Servers. Enter the route pattern number you want associated with each code. The pattern choice you list on one screen automatically defaults to the other screens of the same table. If you use one pattern for most of the codes, assign that pattern to choice 1.

Valid entries	Usage
1 to **999** or blank	For S8300/S87XX Servers
1 to **254**	For DEFINITY CSI.

RHNPA TABLE

Display-only field indicating the table number.

Route Pattern

The Route Pattern screen defines the route patterns used by your server running Communication Manager. Each route pattern contains a list of trunk groups that can be used to route the call. The maximum number of route patterns and trunk groups allowed depends on the configuration and memory available in your system.

Use this screen to insert or delete digits so AAR or ARS calls route over different trunk groups. You can convert an AAR number into an international number, and insert an area code in an AAR number to convert an on-network number to a public network number. Also, when a call directly accesses a local central office (CO), if the long-distance carrier provided by your CO is not available, Communication Manager can insert the dial access code for an alternative carrier into the digit string.

Field descriptions for page 1

Figure 269: Route Pattern screen

```
change route-pattern 1                                           Page 1 of X
                       Pattern Number: 1_    Pattern Name:
                                SCCAN? n       Secure SIP? n

                            No.                                    DCS/
  Grp. FRL NPA Pfx Hop Toll Del   Inserted                         QSIG   IXC
   No.         Mrk Lmt List Dgts  Digits                           Intw
1: ___   _ ___  _  __   __   __   ________________________________   n   user
2: ___   _ ___  _  __   __   __   ________________________________   n   user
3: ___   _ ___  _  __   __   __   ________________________________   n   user
4: ___   _ ___  _  __   __   __   ________________________________   n   user
5: ___   _ ___  _  __   __   __   ________________________________   n   user
6: ___   _ ___  _  __   __   __   ________________________________   n   user

    BCC VALUE  TSC CA-TSC   ITC BCIE Service/Feature BAND  No. Numbering LAR
   0 1 2 M 4 W     Request                                Dgts   Format
                                                    Subaddress
 1: y y y y y n  y  none  ___ both ept outwats-bnd____    _    ________  none
 2: y y y y y n  y            rest     _______________    _    ________  next
 3: y y y y y n  y            rest     _______________    _    ________  rehu
 4: y y y y y n  y            rest     _______________    _    ________  none
 5: y y y y y n  y            rest     _______________    _    ________  none
 6: y y y y y n  y            rest     _______________    _    ________  none
```

Band

Enter a number that represents the OUTWATS band number (US only).

WATS is a voice-grade service that provides both voice and low-speed data transmission calls to defined areas (bands) for a flat rate charge.

This field appears when the **Services/Features** field is **outwats-bnd** and when **ISDN-PRI** or **ISDN-BRI Trunks** field is **y** on the System Parameters Customer-Options (Optional Features) screen. **Band** is required by Call-by-Call Service Selection.

BCC Value

Bearer Capability Class (BCC) identifies the type of call appropriate for this trunk group, such as voice calls and different types of data calls. This field appears when the **ISDN-PRI** or **ISDN-BRI Trunks** field is **y** on the System Parameters Customer-Options (Optional Features) screen.

Valid entries	Usage
y/n	Enter **y** in appropriate BCC column (0, 1, 2, 3, 4, or W) if the BCC is valid for the associated route pattern. A trunk group preference can have more than one BCC.

The following table explains BCC values

:

BCC Value	Description
0	Voice-Grade Data and Voice
1	56-kbps Data (Mode 1)
2	64-kbps Data (Mode 2)
M	Multimedia call
4	64-kbps Data (Mode 0)
W	128 to 1984-kbps Data (Wideband)

BCIE (Bearer Capability Information Element)

This field applies to ISDN trunks and appears if ITC is **both**.

Valid entries	Usage
ept (endpoint) **unr** (unrestricted)	Use BCIE to determine how to create the ITC codepoint in the setup message.

CA-TSC Request

Use CA-TSC on ISDN B-channel connections.

Valid entries	Usage
as-needed	The CA-TSC is set up only when needed. This causes a slight delay. Avaya recommends this entry for most situations.
at-setup	The CA-TSC is automatically set up for every B-channel call whether or not it is needed.
none	No CA-TSC is set up. Permits tandeming of NCA-TSC setup requests0.

DCS/QSIG Intw

This field only appears if the **Interworking with DCS** field on the System Parameters Customer-Options (Optional Features) screen is set to **y**.

Valid entries	Usage
y/n	Enter **y** to enable DCS/QSIG Voice Mail Interworking.

FRL

Valid entries	Usage
0 to **7**	Enter the Facility Restriction Level (FRL) associated with the entries on this row (preference). **0** is the least restrictive and **7** is the most restrictive. The calling party's FRL must be greater than or equal to this FRL to access the associated trunk-group.

SECURITY ALERT:
For system security reasons, Avaya recommends using the most restrictive FRL possible.

Grp No

Valid entries	Usage
1 to **666**	Enter the trunk group number associated with this row (preference). For DEFINITY CSI.
1 to **2000**	For S8300/S87XX Servers.

Hop Lmt

Enter the number of hops for each preference. A hop is when a call tandems through a server to another destination. Limiting the number of hops prevents circular hunting, which ties up trunk facilities without ever completing the call. Communication Manager blocks a hop equal to or greater than the number you enter

.

Valid entries	Usage
blank	Indicates that there is no limit to the number of hops for this preference.
1 to **9**	To limit the number of hops if using the tandem hop feature.
1 to **32**	If using the transit feature.

Inserted Digits

Enter the digits you want inserted for routing. Communication Manager can send up to 52 digits. This includes up to 36 digits you can enter here plus up to 18-digits originally dialed. Special symbols count as two digits each.

Valid entries	Usage
0 to 36 digits (**0** to **9**)	Enter the digits you want inserted for routing.
*****	When * is in the route pattern and the outgoing trunk is signaling type "mf", the MFC tone for the "end-of-digits" is sent out to the CO in place of the *.
	1 of 2

Valid entries	Usage
#	When # is in the route pattern and the outgoing trunk is signaling type "mf", the MFC tone for the "end-of-digits" is sent out to the CO in place of the #.
‘,’	Use 2 places. Creates a 1.5 second pause between digits being sent. Do not use as the first character in the string unless absolutely necessary. Misuse can result in some calls, such as Abbreviated Dialing or Last Number Dialed, not completing.
+	Wait for dial tone up to the Off Premises Tone Detection Timer and then send digits or intercept tone based on **Out Pulse Without Tone** y/n on the Feature-Related System Parameters screen.
%	Start End-to-End Signaling.
!	Wait for dial tone without timeout and then send DTMF digits.
&	Wait for ANI (used for Russian pulse trunks)
p	The associated trunk group must be of type **sip**. Enter the single digit **p** for fully qualified E.164 numbers. The p is translated to a + and is prepended to the digit string.
	2 of 2

ITC (Information Transfer Capability)

Use Information Transfer Capability (**ITC**) to identify the type of data transmission or traffic that this routing preference can carry. The ITC applies only to data calls (BCC 1 through 4).

This field must be **unre** or **both** if the **BCC** is **y** and the **BCC** value is **W**

Valid entries	Usage
both	Calls from restricted and unrestricted endpoints can access the route pattern.
rest(ricted)	Calls from restricted endpoints can access the route pattern.
unre(stricted)	Calls from unrestricted endpoints can access the route pattern.

IXC

Inter-Exchange Carrier (**IXC**) identifies the carrier, such as AT&T, used for calls that route via an IXC, and for Call Detail Recording (CDR).

This field appears when the **ISDN-PRI** or **ISDN-BRI Trunks** field is **y** on the System Parameters Customer-Options (Optional Features) screen.

Valid entries	Usage
Valid carrier code	Identifies the carrier for IXC calls
user	For presubscribed carrier. Used when an IXC is not specified.
none	This field must be **none** for non-ISDN trunk groups and for Bellcore NI-2 Operator Service Access. If you need to send an IXC code for a non-ISDN trunk group, enter the IXC code in the Inserted Digits field.

LAR

Enter the routing-preference for Look Ahead Routing. Following are the causes that trigger LAR:

* #3 - No Route to Destination..CV_NRTD
* #6 - channel unacceptable...CV_CU
* #34 - No Circuit or Channel Available..............................CV_NCOCA
* #38 - Network Failure..CV_NETFAIL
* #41 - Temporary Failure...CV_TFAIL
* #42 - Switching Equipment Congestion...........................CV_SEC
* #43 - User Information Discarded...................................CV_UID
* #44 - Requested Circuit/Channel Not Available...............CV_RCCNA
* #47 - Resources Unavailable, Unspecified......................CV_RUU
* #58 - bearer capability not presently available................CV_BCNPA
* #65 - bearer capability not implemented..........................CV_BCNI
* #79 - service/option not implemented, inspect.................CV_SOONIU
* #82 - identified channel does not exist........................... CV_ICDNE
* #102 - recover on timer expiry.......................................CV_ROTE

Valid entries	Usage
next	Go to the next routing preference and attempt the call again.
rehu	Rehunt within the current routing-preference for another trunk to attempt the call again.
none	Look Ahead Routing is not enabled for the preference.

No. Del. Digits

Use this field to modify the dialed number so an AAR or ARS call routes over different trunk groups that terminate in servers/switches with different dial plans.

Valid entries	Usage
0 to **28** or blank	Enter the total number of digits you want the system to delete before it sends the number out on the trunk. Use for calls that route: ● to or through a remote server ● over tie trunks to a private network server ● over Central Office (CO) trunks to the serving CO

No. Dgts Subaddress

Allows a caller to reach a number where the Avaya S8XXX Server's digit processing deletes the dialed number and inserts the listed directory number (LDN). The LDN then is sent to the destination address and the dialed extension is sent in the calling party subaddress information element (IE). At the receiving end, the call terminates to the user indicated by the subaddress number instead of the attendant. Administrable when, on the **System Parameters Customer-Options (Optional Features)** screen, the **ISDN Feature Plus** field is **y**.

Valid entries	Usage
1 to **5** or blank	Enter the number of dialed digits to send in the calling party subaddress IE.

NPA

This entry is not required for AAR.

Valid entries	Usage
3-digit number	Enter the 3-digit Numbering Plan Area (NPA) (or area code) for the terminating endpoint of the trunk group. Call your local telephone company to verify this number if you need help. For WATS trunks, the terminating NPA is the same as the home NPA unless the Local Exchange Carrier requires 10 digits for local NPA calls.
blank	For AAR calls and for tie trunks

Numbering Format

Use this field to specify the numbering format of calls routed over trunk group of the preferred routing pattern. This field applies to ISDN and SIP trunk groups.

Valid entries	Numbering Plan Identifier	Type of Numbering
blank	Derived from the value of the Call Type field on the ARS Analysis screen or the AAR Analysis screen.	Derived from the value of the Call Type field on the ARS Analysis screen or the AAR Analysis screen.
natl-pub	E.164(1)	national(2)
intl-pub	E.164(1)	international(1)
locl-pub	E.164(1)	local/subscriber(4)
pub-unk	E.164(1)	unknown(0)
lev0-pvt	Private Numbering Plan - PNP(9)	local(4)
lev1-pvt	Private Numbering Plan - PNP(9)	Regional Level 1(2)
lev2-pvt	Private Numbering Plan - PNP(9)	Regional Level 2(1)
unk-unk	unknown(0)	unknown(0)

Note:
To gain access to Telcordia Technologies NI-2 Operator Services, type **unk-unk** in the **Inserted Digits** field.

Note:
For Network Call Redirection or Transfer, type **lev0-pvt** in the **Numbering Format** field.

Pattern Name

Enter an alphanumeric name for identification purposes.

Pattern Number

This display-only field shows the route pattern number (**1** to **640**).

Prefix Mark

This entry is not required for AAR. For ARS, enter a number from **0** to **4** or blank.

Prefix Marks set the requirements for sending a prefix digit 1, indicating a long-distance call. Prefix Marks apply to 7- or 10-digit Direct Distance Dialing (DDD) public network calls. A prefix digit 1 is sent only when call type is foreign number plan area (FNPA) or home numbering plan area (HNPA) in the ARS Digit Analysis table.

For a WATS trunk, the Prefix Mark is the same as the local CO trunk.

Valid entries	Usage
0	• Suppress a user-dialed prefix digit 1 for 10-digit FNPA calls. • Leave a user-dialed prefix digit 1 for 7-digit HNPA calls. • Leave a prefix digit 1 on 10-digit calls that are not FNPA or HNPA calls. Do not use **Prefix Mark 0** in those areas where all long-distance calls must be dialed as 1+10 digits. Check with your local network provider.
1	• Send a 1 on 10-digit calls, but not on 7-digit calls. Use Prefix Mark 1 for HNPA calls that require a 1 to indicate long-distance calls.
2	• Send a 1 on all 10-digit and 7-digit long-distance calls. Prefix Mark 2 refers to a Toll Table to define long distance codes.
3	• Send a 1 on all long-distance calls and keep or insert the NPA (area code) so that all long distance calls are 10-digit calls. The NPA is inserted when a user dials a Prefix digit 1 plus 7-digits. Prefix Mark 3 refers to a Toll Table to define long distance codes.
4	• Always suppress a user-dialed Prefix digit 1. Use Prefix Mark 4, for example, when ISDN calls route to a server that rejects calls with a prefix digit 1.
blank	For tie trunks, leave this field blank.

SCCAN

This field appears when **Enhanced EC500** on the System Parameters - Customer Options screen is set to **y**.

Note:
When the **SCCAN** field is set to **y**, non-SCCAN-associated fields are hidden. Only the **Pattern Number**, **Pattern Name**, **SCCAN**, **Secure SIP**, and **Grp No** fields appear.

Valid entries	Usage
y/n	Enter **y** to indicate that this route pattern supports incoming SCCAN calls.

Secure SIP

Valid entries	Usage
y/n	Specify whether the SIP or SIPS prefix will be used, if the call is routed to a SIP Enablement Services (SES) trunk preference. If SES trunks are not specified on the Route Pattern screen, the call will be routed over whatever trunk is specified. Therefore, to ensure a SES TLS connection when such a route pattern is invoked, only SES trunks should be specified. The only instance for entering **y** in this field is when the source provider *requires* a secure SIP protocol. Default is **n**.

Service/Feature

This field appears when **ISDN-PRI** or **ISDN-BRI Trunks** is **y** on the System Parameters Customer-Options (Optional Features) screen.

Enter up to 15 characters to identify the Service/Feature carried by the information element (IE) in a call in this route pattern. This field is required by Call-by-Call Service Selection, and Network Call Redirection and Transfer.

Note:
User-defined service types, defined on the **Network Facilities** screen, can also be used. In addition to pre-defined Services/Features, any user-defined **Facility Type** of **0** (feature), **1** (service), or **3** (outgoing) on the **Network Facilities** screen is allowed. See the description of the **Network Facilities** screen for more information on usage allocation.

Valid entries		
accunet	**multiquest**	**sdn** (Enter to allow Network Call Redirection/Transfer)
i800	**operator**	**sub-operator**
		1 of 2

Valid entries		
inwats	**oper-lds** (operator and lds)	**sub-op-lds** (sub-operator and lds)
lds	**oper-meg** (operator and megacon)	**sub-op-meg** (sub-operator and megacom)
mega800	**oper-sdn** (operator and sdn)	**sub-op-sdn** (sub-operator and sdn)
megacom	**outwats-bnd**	**wats-max-bnd**
		2 of 2

Toll List

This entry is not required for AAR.

Valid entries	Usage
1 to **32** or blank	For ARS, enter the number of the ARS Toll Table associated with the terminating NPA of the trunk group. You must complete this field if **Prefix Mark** is **2** or **3**.

TSC

Set **TSC** to **y** for feature transparency on DCS+ calls and to use QSIG Call Completion.

Valid entries	Usage
y/n	Enter **y** to allow Call-Associated TSCs, and to allow incoming Non-Call-Associated TSC requests to be tandemed out for each preference.

Security-Related System Parameters

Use this screen to determine when Communication Manager reports a security violation. Many of the fields on this screen repeat for each type of security violation. We have explained them once here, but the usage is the same for all. See Security Violations Notifications in *Avaya Aura™ Communication Manager Feature Description and Implementation,* 555-245-205, for more information.

Field descriptions for page 1

Figure 270: Security-Related System Parameters screen

```
change system-parameters security                           Page 1 of x
                     SECURITY-RELATED SYSTEM PARAMETERS

SECURITY VIOLATION NOTIFICATION PARAMETERS

  SVN Login Violation Notification Enabled? y
        Originating Extension: _____ Referral Destination: _____
       Announcement Extension: _____

  SVN Remote Access Violation Notification Enabled? y
        Originating Extension: _____ Referral Destination: _____
       Barrier Code Threshold: 10          Time Interval: 0:03
       Announcement Extension: _____

  SVN Authorization Code Violation Notification Enabled? y
        Originating Extension: _____
         Referral Destination: _____
 Authorization Code Threshold: __         Time Interval: _____
       Announcement Extension: _____
```

SECURITY VIOLATION NOTIFICATION PARAMETERS

SVN Login (Violation Notification, Remote Access, Authorization Code) Enabled

Valid entries	Usage
y/n	Set to **y** if you want Communication Manager to notify you when a login violation occurs. If this field is **y**, the next 5 fields appear so you can establish the parameters for what is considered a security violation.

Announcement Extension

If you enter a value in this field, the server running Communication Manager *calls the referral destination, then plays this announcement upon answer.*

Valid entries	Usage
Valid extension	The announcement extension where SVN violation announcement resides.

Originating Extension

The originating extension initiates the referral call in the event of a security violation. It also sends the appropriate alerting message or display to the referral destination.

Valid entries	Usage
An unassigned extension	If you establish notification for more than one type of security violations, you must assign a different extension to each one. When Communication Manager generates a referral call, *this extension and the type of violation appear on the display at the referral destination.*

Referral Destination

The referral destination receives the referral call when a security violation occurs. The referral destination telephone must have a display, unless the you assign an Announcement Extension.

Valid entries	Usage
An extension	Enter the extension of the telephone, attendant console, or vector directory number (VDN) that you want to receive the referral call for each type of violation. This can be the same extension for all type of violations. If you use a VDN, you must complete the Announcement Extension field. You can also use Call Vectoring Time-of-Day routing to route the referral call to different destinations based on the time of day or the day of the week.

Time Interval

Use this field to enter a time interval for the violation notification.

Valid entries	Usage
0:01 to **7:59**	The range for the time interval is one minute to eight hours. Entered in the screen **x:xx**. For example, if you want the time interval to be one minute, you enter 0:01. If you want the time interval to be seven and one-half hours, you enter 7:30.

SVN Remote Access Violation Notification Enabled

Use the **SVN Remote Access Violation Notification Enabled** and the **SVN Authorization Code Violation Notification Enabled** fields to establish parameters for remote access security violations. A remote access violation occurs if a user enters incorrect barrier codes. You cannot set the system to disable remote access following a security violation unless you have turned these fields on.

SVN Authorization Code Violation Notification Enabled

Use the **SVN Remote Access Violation Notification Enabled** and the **SVN Authorization Code Violation Notification Enabled** fields to establish parameters for remote access security violations. A remote access violation occurs if a user enters incorrect barrier codes. You cannot set the system to disable remote access following a security violation unless you have turned these fields on.

Field descriptions for page 2

Figure 271: Security-Related System Parameters screen (for DEFINITY CSI)

```
change system-parameters security                              Page 2 of x
                     SECURITY-RELATED SYSTEM PARAMETERS

  SECURITY VIOLATION NOTIFICATION PARAMETERS

    SVN Station Security Code Violation Notification Enabled? y
            Originating Extension: _____      Referral Destination: _____
  Station Security Code Threshold: 10                Time Interval: 0:03
           Announcement Extension: _____

  STATION SECURITY CODE VERIFICATION PARAMETERS

                       Minimum Station Security Code Length: 4
    Security Code for Terminal Self Administration Required? y
                      Receive Unencrypted from IP Endpoints? n

  REMOTE MANAGED SERVICES
                                          RMS Feature Enabled? y
                            Port Board Security Notification? y
                   Port Board Security Notification Interval? 60

  ACCESS SECURITY GATEWAY PARAMETERS

        MGR1? n    INADS? n
         EPN? n      NET? n
```

SECURITY VIOLATION NOTIFICATION PARAMETERS

SVN Station Security Code Violation Notification Enabled

Station Security codes are used to validate logins to a particular extension (for example, a home agent using an extender, or two part-time workers using the same telephone, but different extensions, through personal station access.) Enter **y** *here to establish parameters for this.*

STATION SECURITY CODE VERIFICATION PARAMETERS

Minimum Station Security Code Length

This determines the minimum required length of the Station Security Codes that you enter on the Station screen.

Valid entries	Usage
3 to **8**	Longer codes are more secure. If station security codes are used for external access to telecommuting features, the minimum length should be 7 or 8.

Receive Unencrypted from IP Endpoints

Valid entries	Usage
y/n	Enter **y** to allow unencrypted data from IP endpoints. Default is **n**.

Security Code for Terminal Self Administration Required

Specifies if a Personal Station Access code is required to enter the Self-Administration mode.

Valid entries	Usage
y/n	Enter **y** to indicate that a security code is required.

REMOTE MANAGED SERVICES

RMS Feature Enabled

Use this field to enable Remote Managed Services. When you set this field to **y**, the **Port Board Security Notification** and **Port Board Security Notification Interval** fields appear.

Valid entries	Usage
y/n	Enter **y** to enable the Remote Managed Services feature. Default is **n**.

Port Board Security Notification

This field appears when **RMS Feature Enabled** is set to **y**.

Valid entries	Usage
y/n	Enter **y** to enable port board denial of service notification. Default is **n**. When you enter **y** in this field, the **Port Board Security Notification Interval** field appears.

Port Board Security Notification Interval

This field appears when the **RMS Feature Enabled** and **Port Board Security Notification** fields are set to **y**.

Valid entries	Usage
60 to **3600** in increments of 10	Enter the desired interval, in seconds, between port board Denial of Service notifications (traps). Default is **60**. **NOTE:** There is no delay before the first trap is sent. The interval administered in this field applies only to the period *between* the sending of the traps.

ACCESS SECURITY GATEWAY PARAMETERS

These fields appear only if the **Access Security Gateway (ASG)** field on the System Parameters Customer-Options (Optional Features) screen is **y**.

EPN

A direct connection to the Expansion Port Network.

Valid entries	Usage
y/n	Any entry attempt through this port receives a challenge response.

INADS

A direct cable connection to the Initialization and Administration System used to remotely initialize and administer Communication Manager.

Valid entries	Usage
y/n	Any entry attempt through this port receives a challenge response.

MGR1

The direct connect system administration and maintenance access interface located on the processor circuit pack. For more information on the circuit pack, see the *Avaya Aura™ Communication Manager Hardware Description and Reference,* 555-245-207.

Valid entries	Usage
y/n	Any entry attempt through this port receives a challenge response.

NET

A dialed-in (or out) connection to the Network Controller circuit pack. For more information on the circuit pack, see the *Avaya Aura™ Communication Manager Hardware Description and Reference*, 555-245-207.

Valid entries	Usage
y/n	Any entry attempt through this port receives a challenge response.

Translation-ID Number Mismatch Interval (days)

A display-only field for all logins except *init*; only *init* logins can change this field. This field specifies the interval (in days) that the system allows a mismatch between the translation ID stored in the Processor circuit pack(s) and on the flash card. Following expiration of this interval, the ability to execute system administration commands that modify translation data is denied for all logins, except for *init*.

Valid entries	Usage
1 to **90**	Enter a number to indicate the number of days the system allows access to system administration commands.

Service Hours Table

Use this screen to establish signaling group parameters for ISDN-PRI, H.323, ATM, and SIP Enablement Services (SES) trunks. Because these trunk types vary in the types of parameters needed, the fields that appear on this screen change depending on the value of the **Group Type** field. Field descriptions are alphabetized for easier reference.

Field descriptions for page 1

Figure 272: Service Hours Table screen

```
change service-hours-table 4                                    Page   1 of   1
                               SERVICE HOURS TABLE:

                                    Number: 4
                               Description:
        Use time adjustments from location:

         MON           TUE           WED           THU           FRI
     Start  End    Start  End    Start  End    Start  End    Start  End
       :     :       :     :       :     :       :     :       :     :
       :     :       :     :       :     :       :     :       :     :
       :     :       :     :       :     :       :     :       :     :
       :     :       :     :       :     :       :     :       :     :
       :     :       :     :       :     :       :     :       :     :

                              SAT           SUN
                          Start  End    Start  End
                            :     :       :     :
                            :     :       :     :
                            :     :       :     :
                            :     :       :     :
                            :     :       :     :
ESC-x=Cancel Esc-e=Submit Esc-p=Prev Pg Esc-n=Next Pg Esc-h=Help Esc-r=Refresh
```

Description

Provides a description for the table. You can enter a 1 to 27-character alphanumeric table name.The default is blank. Example: Call-ahead Reservations

Number

Displays the table number that you entered on the command line.

Start/End

Defines the range of office hours for each day of the week. Always make sure that the start time is earlier than the end time.

- hour - 0-23
- minute - 0-59

The hour range must be within the specified day, from 00:00 (midnight) until 23:59. If a time range goes past midnight (for example, Friday 19:00 to Saturday 02:00), enter the time in two ranges. Set up the first range as Friday from 19:00 to 23:59 and the second range as Saturday from 00:00 to 01:59.

A time is considered to be in the table from the first second of the start time (for example, 08:00:00). Also, it is still considered to be in the table until the last second of the end time (for example, 17:00:59).

Use time adjustments from location

Points to a field on the **Locations** screen for time zone offset and daylight savings time rule time adjustments.

- The Multiple Locations option must be enabled in order to administer more than one location (locations 2-250).
- You can assign a location to a gateway or to a network region.
- Administer the location where the incoming trunk terminates.

Signaling Group

Use this screen to establish signaling group parameters for ISDN-PRI, H.323, ATM, and SIP Enablement Services (SES) trunks. Because these trunk types vary in the types of parameters needed, the fields that appear on this screen change depending on the value of the **Group Type** field. Field descriptions are alphabetized for easier reference.

Field descriptions for page 1

Figure 273: Signaling Group screen when the Group Type field is atm

```
add signaling-group nnn                                         Page 1 of x
                              SIGNALING GROUP

Group Number  ___          Group Type:  atm___       Name:
                                               Max Number of NCA TSC: ___
                             D-Channel:         Max number of CA TSC:  ___
                                             Trunk Group for NCA TSC:  ___
     Trunk Group for Channel Selection:  ___
    TSC Supplementary Service Protocol: _      Network Call Transfer? n

CIRCUIT PARAMETERS
     Virtual Path Identifier: 0
  Virtual Channel Identifier: 0

              Signaling Mode: isdn-pri            Circuit Type: T1
                   Idle Code: 11111111                 Connect: network
        Interface Companding: mulaw
            Country Protocol: 1
            Protocol Version: d

                                DCP/Analog Bearer Capability:
                                        Interworking Message:
```

Figure 274: Signaling Group screen when the Group Type field is h.323

```
change signaling-group 22                                       Page   1 of   x
                                SIGNALING GROUP

 Group Number: 22             Group Type: h.323
                           Remote Office? c         Max number of NCA TSC: 0
                                     SBS? n          Max number of CA TSC: 0
     IP Video? n                                   Trunk Group for NCA TSC:
       Trunk Group for Channel Selection: 22
      TSC Supplementary Service Protocol: a         Network Call Transfer? n
                         T303 Timer(sec): 10
   H.245 DTMF Signal Tone Duration(msec):
   Near-end Node Name: TofuClan172-1        Far-end Node Name: TacoClan172-1
 Near-end Listen Port: 6022               Far-end Listen Port: 6022
                                       Far-end Network Region: 12
         LRQ Required? n                Calls Share IP Signaling Connection? n
         RRQ Required? n
                                             Bypass If IP Threshold Exceeded? n
                                                      H.235 Annex H Required? n
         DTMF over IP: out-of-band            Direct IP-IP Audio Connections? y
  Link Loss Delay Timer(sec): 90                        IP Audio Hairpinning? y
         Enable Layer 3 Test? n                  Interworking Message: PROGre
 H.323 Outgoing Direct Media? n
```

Figure 275: Signaling Group screen when the Group Type field is isdn-pri

```
add signaling-group n                                       Page 1 of x
                              SIGNALING GROUP
Group Number  ___             Group Type:  isdn-pri
                  Associated Signaling?         Max Number of NCA TSC: ___
                     Primary D-Channel:          Max number of CA TSC:  ___
                                              Trunk Group for NCA TSC:  ___
     Trunk Group for Channel Selection:  ___ X-Mobility/Wireless Type: None
    TSC Supplementary Service Protocol: _
                     ETSI CCBS Support: both directions
```

Figure 276: Signaling Group screen when the Group Type field is sip

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

 Group Number: 6             Group Type: sip
                        Transport Method: tls
  IMS Enabled? n




   Near-end Node Name:                         Far-end Node Name:
 Near-end Listen Port: 5061                  Far-end Listen Port: 5061
                                          Far-end Network Region:
Far-end Domain:

                                             Bypass If IP Threshold Exceeded? n


         DTMF over IP: rtp-payload            Direct IP-IP Audio Connections? y
Session Establishment Timer(min): 3                     IP Audio Hairpinning? n
                                                    Direct IP-IP Early Media? n
H.323 Station Outgoing Direct Media? n            Alternate Route Timer(sec): 6
```

Alternate Route Timer

Appears when the **Group Type** field is **sip**. The default value is 6 seconds.

Valid entries	Usage
2 to 30	The time in seconds Communication Manager waits before trying for an alternate route to establish a call.

Associated Signaling

Appears when the **Group Type** field is **isdn-pri**.

Valid entries	Usage
y	Enter **y** to use associated signaling.
n	Enter **n** to use non-facility associated signaling.

Bypass If IP Threshold Exceeded

Appears when the **Group Type** field is **h.323** or **sip**.

Valid entries	Usage
y/n	Enter **y** to automatically remove from service the trunks assigned to this signaling group when IP transport performance falls below limits administered on the **IP-Options System Parameters** screen.

Calls Share IP Signaling Connection

Appears when the **Group Type** field is **h.323** or **sip**.

Valid entries	Usage
y/n	Enter **y** for inter-connection between servers running Communication Manager. If **y**, then **LRQ Required** must be **n**. This field should be set to **n** if either the local and/or remote server is not running Communication Manager. **Note:** For an H.323 signaling group type, Avaya recommends a value of **y** for **Enable Layer 3 Test** when **Calls Share IP Signaling Connection** is **y** and the far-end is Communication Manager. When both the near and far-end servers are running Communication Manager, the value in this field must be the same for both. When you change the value in this field, the system displays the following message: `Far end Communication Manager Signaling-Group must have same value.`

Circuit Type

Appears when the **Group Type** field is **atm**.

Valid entries	Usage
T1	Results in page 2 displaying 24 channels.
E1	Results in page 2 displaying 31 channels.

Connect

Appears when the **Group Type** field is **atm**. In order to control communications at layers 2 and 3 of the ISDN-PRI protocol, use this field to specify what is on the far end of this link.

Valid entries	Usage
host	Enter **host** when the link connects Communication Manager to a computer.
network	Enter **network** when the link connects Communication Manager to a central office or any other public network switch.
pbx	Enter **pbx** if this link is connected to another switch in a private network. If **pbx** is entered, the **Interface** field appears.

Country Protocol

Appears when the **Group Type** field is **atm**. The entry in this field must match the country protocol used by the far-end server. For connections to a public network, your network service provider can tell you which country protocol they are using. For a list of country codes, see the Country code table on page 886.

Valid entries	Usage
1 to **25**	Enter the country protocol used by the central office at which this link terminates.
etsi	Enter **etsi** if your network service provider uses the protocol of the European Telecommunications Standards Institute (ETSI). Enter **etsi** only if the **Signaling Mode** field is **isdn-pri**.

D Channel

Appears when the **Group Type** field is **atm**. Enter the necessary characters.

Valid entries	Usage
01 to **03** (DEFINITY CSI) or **1** to **64** (S87XX/S8300 Servers)	First and second characters are the cabinet number
A to **E**	Third character is the carrier
0 to **20**	Fourth and fifth character are the slot number
01 to **04** (Analog TIE trunks) **09** to **32**	Six and seventh characters are the circuit number

DCP/Analog Bearer Capability

Appears when the **Group Type** field is **atm** or **h.323**. This field sets the information transfer capability in a bearer capability IE of a setup message to **speech** or **3.1kHz**.

Valid entries	Usage
3.1kHz	Provides 3.1kHz audio encoding in the information transfer capability. This is the default.
speech	Provides speech encoding in the information transfer capability.

Direct IP-IP Audio Connections

Appears when the **Group Type** field is **h.323** or **sip**. Allows direct audio connections between H.323 endpoints. For SIP Enablement Services (SES) trunk groups, this is the value that allows direct audio connections between SES endpoints.

Valid entries	Usage
y/n	Enter to **y** to save on bandwidth resources and improve sound quality of voice over IP (VoIP) transmissions.

DTMF Over IP

Appears when the **Group Type** field is **h.323** or **sip**. Use this field to specify the touchtone signals that are used for dual-tone multifrequency (DTMF) telephone signaling.

Valid entries	Usage
in-band	All G711 and G729 calls pass DTMF **in-band**. DTMF digits encoded within existing RTP media stream for G.711/G.729 calls. G.723 is sent **out-of-band**.
in-band-g711	Only G711 calls pass DTMF in-band.
out-of-band	All IP calls pass DTMF **out-of-band**. For IP trunks, the digits are done with either Keypad IEs or H245 indications. This value is not supported for SIP signaling. This is the default for newly added H.323 signaling groups.
rtp-payload	This is the method specified by RFC1533. This is the default for newly added SIP signaling groups. Support for SIP Enablement Services (SES) trunks requires the default entry of **rtp-payload**.

Enable Layer 3 Test

Appears when the **Group Type** field is **h.323** or **sip**.

Valid entries	Usage
y/n	Enter **y** if you want Communication Manager to run the Layer 3 test that verifies that all connections known at the near-end are recognized at the far-end. The default value is **y** (test enabled) for new Communication Manager installations, **n** for upgrades from previous releases. **Note:** For an H.323 signaling group type, Avaya recommends a value of **y** for **Enable Layer 3 Test** when **Calls Share IP Signaling Connection** is **y** and the far-end is Communication Manager. If this field is administered as **y** (test enabled) and the **Far-end Node Name** field does not have an administered IP address, then you cannot submit the form, and the Layer 3 test aborts.

ETSI CCBS Support

Appears when the **Group Type** field is **isdn-pri** and **TSC Supplementary Service Protocol** is set to **c** for ETSI.

Valid entries	Usage
none	Interface supports neither incoming nor outgoing ETSI CCBS. This is the default.
incoming	Interface supports only incoming ETSI CCBS.
outgoing	Interface supports only outgoing ETSI CCBS.
both directions	Interface supports incoming and outgoing ETSI CCBS. **Note**: When upgrading from a version of Communication Manager that is earlier than 5.1, this is the default.

Far-end Domain

Appears when the **Group Type** field is **sip**. The number of the network region that is assigned to the far-end of the trunk group. For example, to route SES calls within your enterprise, enter the domain assigned to your proxy server. For external SES calling, the domain name could be that of your SES service provider. If blank, the far-end IP address is used.

Valid entries	Usage
Max. 40 character string	Enter the name of the IP domain for which the far-end proxy is responsible (that is, authoritative), if different than the near-end domain. If the domains are the same, leave this blank.
blank	Far-end domain is unspecified. Note that If you leave this field blank, the system might display the following message: "Warning: unspecified far-end IP address is vulnerable to denial of service attacks."

Far-end Listen Port

Appears when the **Group Type** field is **h.323** or **sip**.

Valid entries	Usage
1 to **65535**	Enter the same number as entered in the **Near-end Listen Port** field. Typically, this is the default of **5061** for SIP over TLS.
blank	Far-end listen port is unspecified. Note that If you leave this field blank, the system might display the following message: "Warning: unspecified far-end IP address is vulnerable to denial of service attacks."

Far-end Network Region

Appears when the **Group Type** field is **h.323** or **sip**. The number of the network region that is assigned to the far-end of the trunk group.

Valid entries	Usage
1 to **250**	Enter the network region number that is assigned to the far end of the trunk group. The region is used to obtain the codec set used for negotiation of trunk bearer capability. Leave blank to select the region of the near-end node.
blank	Far-end network region is unspecified. Note that If you leave this field blank, the system might display the following message: "Warning: unspecified far-end IP address is vulnerable to denial of service attacks."

Far-end Node Name

Appears when the value of the entry in the **Group Type** field is **atm** or **sip**. Enter the node name for the far-end Control Lan (C-LAN) IP interface used for trunks assigned to this signaling group. The node name must be administered on the IP Node Names screen.

Valid entries	Usage
Name of an administered IP node.	Describe the far-end node. Note that If you leave this field blank, the system might display the following message: "Warning: unspecified far-end IP address is vulnerable to denial of service attacks."

Tip:
For SIP Enablement Services (SES) signaling groups, if either the node name or port differs for each group, you have different SES signaling connections, and you should administer a maximum of 10 using TLS. If you administer more than 10 TLS signaling connections, and they are all in use at the same time, the results can be unpredictable. Note that if the node names and ports match, you can administer as many identical SES signaling groups using TLS as desired.

Group Number

A display-only field identifying the signaling group.

Group Type

This field describes the type of protocol to be used with the signaling group.

Valid entries	Usage
atm	Use for Asynchronous Transfer Mode signaling trunks
h.323	Use for h.323 protocol or when using SBS signaling trunks.
isdn-pri	Integrated Service Digital Network Primary Rate Interface
sip	For SIP Enablement Services (SES) on the Avaya S8300, S8400, S8500, and S87XX.

H.235 Annex H Required

Appears only for signaling group type **h.323**.

Valid entries	Usage
y/n	Enter **y** to indicate that the Communication Manager server requires the use of H.235 Annex H (now called H.235.5) protocol for authentication during registration. **NOTE:** If this field is set to **y**, then **LRQ Required** on the Station screen must also be set to **y**, or the signaling group will not work. This is because the LRQ is required for the exchanges of authentication data.

H.323 Station Outgoing Direct Media

Appears only when **Group Type** is **h.323** or **sip**, and **Direct IP-IP Audio Conections** is **y**.

Valid entries	Usage
y	When set to **y**, a call from an H.323 station over a trunk that uses this signaling group starts as a direct media call. The IP address and port of the H.323 station are sent as the media and medic control channel addresses in the SETUP/INVITE message. **Note**: If, on an outgoing Direct Media call from an IP (H.323) telephone over this trunk, you attempt to transfer the call, conference another party, or put the call on hold while the call is still in ringing state, the operation fails.
n	The IP address of the MEDPRO board is sent in the SETUP/INVITE Message.

H.245 DTMF Signal Tone Duration(msec)

Valid entries	Usage
80 to **350** or blank	This field specifies the duration of DTMF tones sent in H.245-signal messages when the **DTMF over IP** field is set to **out-of-band** on the Signaling Group screen for IP Trunks. The default Value is blank.

Idle Code

Appears when the **Group Type** field is **atm**. This entry sets the signal sent out over idle DS0 channels. The string must be compatible with the protocol used by the far-end switch/server.

Valid entries	Usage
0, **1**	Enter an 8-digit string.

IMS Enabled

Appears when the Group Type field is sip.

Valid entries	Usage
y	SIP requests that match the domain in the **Far-End Domain** field are accepted. Outgoing SIP messages use a trunk group with signaling set to IMS.
n	The default value is **n**(o).

Interface

This field only appears when the **Connect** field is **pbx**. The **Interface** field controls how your server negotiates glare with the far-end switch.

Valid entries	Usage
Use the following 2 values for private network applications in the U.S.	
network	Enter **network** if your server overrides the other end when glare occurs. If you are connecting your server to a host computer, set this field to **network**.
user	Enter **user** if your server releases the contested circuit and looks for another when glare occurs. If you are connecting your server to a public network, set this field to **user**.
Use the following values for private networks (including QSIG networks) outside the U.S. Entering either of these values causes the **Peer Protocol and Side** fields to appear.	
peer-master	Enter **peer-master** if your switch overrides the other end when glare occurs.
peer-slave	Enter **peer-slave** if your switch releases the contested circuit and looks for another when glare occurs.

Interface Companding

Appears when the **Group Type** field is **atm**. Indicates the companding algorithm expected by the system.The entry in this field must match the companding method used by the far-end switch.

Valid entries	Usage
alaw **mulaw**	Enter **alaw** for E-1 service. Enter **mulaw** for T-1 service.

Interworking Message

Appears when the **Group Type** field is **atm**, **h.323**, or **sip**. This field determines what message Communication Manager sends when an incoming ISDN trunk call interworks (is routed over a non-ISDN trunk group).

Valid entries	Usage
PROGress	Normally select this value. PROGress asks the public network to cut through the B-channel and let the caller hear tones such as ringback or busy tone provided over the non-ISDN trunk.
ALERTing	ALERTing causes the public network in many countries to play ringback tone to the caller. Select this value only if the DS1 is connected to the public network, and it is determined that callers hear silence (rather than ringback or busy tone) when a call incoming over the DS1 interworks to a non-ISDN trunk.

IP Audio Hairpinning

Appears when the **Group Type** field is **h.323** or **sip**. The **IP Audio Hairpinning** field entry allows the option for H.323 and SIP Enablement Services (SES)-enabled endpoints to be connected through the IP circuit pack in the Avaya S8XXX Server, without going through the time division multiplexing (TDM) bus.

Valid entries	Usage
y/n	Type **y** to enable hairpinning for H.323 or SES trunk groups. Default is **n**.

IP Video

Appears for signaling group type **h.323** and **sip**.

Valid entries	Usage
y/n	Enter **y** to enable IP video capability for this signaling group. Default is **n**.

Link Loss Delay Timer (sec)

Use this field to specify how long to hold the call state information in the event of an IP network failure or disruption. Communication Manager preserves calls and starts this timer at the onset of network disruption (signaling socket failure). If the signaling channel recovers before the timer expires, all call state information is preserved and the signaling channel is recovered. If the signaling channel does not recover before the timer expires, the system:

- raises an alarm against the signaling channel
- maintains all connections with the signaling channel
- discards all call state information about the signaling channel

.

Valid entries	Usage
1 to **180**	Enter the number of seconds to delay the reaction of the call controller to a link bounce. Default is **90**.

LRQ Required

Appears when the **Group Type** field is **h.323**. Allows IP trunk availability to be determined on a per call basis. When this option is enabled, a RAS-Location Request (LRQ) message is sent to the far-end gatekeeper prior to each call over the IP trunk. The far-end gatekeeper responds with a RAS-Location Confirm (LCF) message, and the call proceeds.

Note:
If the **H.235 Annex H Required** field on the Signaling Group screen is set to **y**, then **LRQ Required** must also be set to **y**, or the signaling group will not work. This is because the LRQ is required for the exchanges of authentication data.

Valid entries	Usage
y	Enter **y** if **H.235 Annex H Required** is **y**, or if the far-end server is not an Avaya DEFINITY or Avaya S8XXX Server, and requires a location request to obtain a signaling address in its signaling protocol. If this field is set to **y**, **Calls Share IP Signaling Connection** must be **n**.
n	Enter **n** if the far-end server is running Communication Manager, unless **H.235 Annex H Required** is set to **y**.

Max number of CA TSC

Appears when the **Group Type** field is **atm**, **h.323**, or **isdn-pri**.

Valid entries	Usage
0 to **619**	Maximum number of simultaneous call-associated Temporary Signaling Connections that can exist in the signaling group. Typically this is the number of ISDN-PRI trunk group members controlled by this signaling group.

Max number of NCA TSC

Appears when the **Group Type** field is **atm**, **h.323**, or **isdn-pri**.

Valid entries	Usage
0 to **256**	Maximum number of simultaneous non-call-associated Temporary Signaling Connections. The TSCs carry signaling for features not associated with a specific call, for example, signals to turn on Leave Word Calling.

Media Encryption

Appears when the Media Encryption feature is enabled in Communication Manager and the **Group Type** field is **h.323**.

Valid entries	Usage
y/n	Enter **y** to enable encryption for trunk calls assigned to this signaling group. If encryption for the signaling group is not enabled, then trunk calls using this signaling group will not be encrypted regardless of IP Codec Set administration.

Name

Appears when the **Group Type** field is **atm**.

Valid entries	Usage
up to 15 alphanumeric characters	Enter 15 alphanumeric characters for identification. **NOTE:** Avaya BRI stations support ASCII characters only. They do not support non-ASCII characters, such as Eurofont or Kanafont. Therefore, if you use non-ASCII characters in any Communication Manager **Name** field, such characters do not display correctly on a BRI station.

Near-end Listen Port

Appears when **Group Type** is **h.323** or **sip**. Defaults to **5061** for SIP Enablement Services (SES) over TLS.

Valid entries	Usage
1719, 1720, or 5000 to **9999**	Enter an unused port number. The default for SIP is **5061**. Avaya recommends **1720** for h.323 and **1719** if LRQ is **y**.

Near-end Node Name

Appears when the value of the entry in the **Group Type** field is **atm** or **sip**. Enter the node name for the Control Lan (C-LAN) IP interface in this Avaya S8XXX Server. The node name must be administered on the IP Node Names screen and the IP Interfaces screen.

Valid entries	Usage
Name of an administered IP node	Describe the near-end node.

Network Call Transfer

Appears when the **Group Type** field is **atm**.

Valid entries	Usage
y/n	Enter **y** to indicate D-channels are supporting ENCT.

Passphrase

Appears when Media Encryption is enabled or the **H.235 Annex H Required** field is **y**. The passphrase is used for both Media Encryption and authentication. This field cannot be left blank.

Valid entries	Usage
8 to 30 alphanumeric characters	Enter a value for the passphrase used to generate a shared "secret" for symmetric encryption of the media session key. The same passphrase must be assigned to the corresponding signaling groups at both ends of an IP trunk. The passphrase: ● Is case sensitive ● Must contain at least 1 alphabetic and at least 1 numeric ● Valid characters also include letters, numbers, and these symbols: **!&*?;'^(),.:-**

Primary D Channel

Appears when the **Group Type** field is **isdn-pri**.

Valid entries	Usage
Cabinet number (**1** to **44**), Carrier (**A** to **E**), Slot (**00** to **20**), Circuit (**09** to **32**)	Enter the letter or number of the Primary D Channel.

Priority Video

Appears when the **Group Type** field is **h.323** or **sip**. For **sip** signaling groups, this field appears when, on the System-Parameters Customer Options screen, **Multimedia SIP Trunking** is **y**.

Valid entries	Usage
y/n	Enter **y** to specify that incoming video calls have an increased likelihood of receiving bandwidth and are also allocated a larger maximum bandwidth per call. Default is **n**.

Protocol Version

Appears when the **Group Type** field is **atm**.

Valid entries	Usage
a **b** **c** **d**	In countries whose public networks allow multiple layer-3 signaling protocols for ISDN-PRI service, use this field to select the protocol that matches your network service provider's protocol.

RRQ Required

Appears when the **Group Type** field is **h.323**. This field specifies the signaling group that serves as a gateway rather than gatekeeper.

Valid entries	Usage
y/n	Displays **y** if the signaling group serves a remote office (gateway). Displays **n** if the signaling group serves a gatekeeper.

Remote Office

Appears when the **Group Type** field is **h.323**.

Valid entries	Usage
y/n	Enter **y** if the signaling group serves a remote office.

SBS

Appears when the **Group Type** field is set to **h.323**. If you set this to **y**, you must set both the **Trunk Group for NCA TSC** field and the **Trunk Group for Channel Selection** field equal to the signaling group number administered for the SBS trunk group. The **TSC Supplementary Service Protocol** field should always be set to **b** to obtain full QSI.

Valid entries	Usage
y/n	Enter **y** to use SBS signaling trunk groups. The default is **n**.

Session Establishment Timer (min)

Appears when the **Group Type** field is **sip**. This field determines how long the system waits before tearing down a ringing call. The default is **3** minutes.

Valid entries	Usage
3 to **120**	The time in minutes Communication Manager waits before tearing down a ring no answer call.

Signaling Mode

A display-only field that appears when the **Group Type** field is **atm**. This field always sets to **isdn-pri**.

SIP Trunk Port

Valid entries	Usage
y/n	Enter **y**, if you want to install Co-Resident SES application on the Communication Manager system. It enables you to configure SES 5.2 with the same port number configured in the Communication Manager signaling group field. **Note:** If a Communication Manager is configured with a Co-Resident SES application then there must not be any SIP signaling-groups administered with the **Near-end Listen Port** field set to **5061** in order to assure proper operation of the SES application.

T303 Timer (sec)

Use this field to enter the number of seconds the system waits for a response from the far end before invoking Look Ahead Routing. Appears when the **Group Type** field is **h.323**.

Valid entries	Usage
2 to **10**	Enter a number of seconds between 2 and 10. Default is 10.

Transport Method

Appears when the **Group Type** field is **sip**. This field is changeable by all login IDs.

Valid entries	Usage
tcp	Transport is accomplished using Transmission Control Protocol (TCP).
tls	Transport is accomplished using Transport Layer Security (TLS). This is the default.

Trunk Group for Channel Selection

Appears when the **Group Type** field is **atm**, **h.323**, or **isdn-pri**.

Valid entries	Usage
1 to **2000**	For S8300/S87XX Servers.

Trunk Group for NCA TSC

Appears when the **Group Type** field is **atm**, **h.323**, or **isdn-pri**.

Valid entries	Usage
1 to **2000** or blank	For Avaya S8300/S87XX Servers.

TSC Supplementary Service Protocol

Appears when the **Group Type** field is **atm**, **h.323**, or **isdn-pri**. Indicate the supplementary service protocol to use for temporary signaling connections.

Valid entries	Usage
a	AT&T, Bellcore, Nortel. When the **Country Code** field on the DS1 Circuit Pack screen is **1A**, SSA selects AT&T custom supplementary services. When the **Country Code** field on the DS1 Circuit Pack screen is **1B**, SSA selects Bellcore Supplementary Services. When the **Country Code** field on the DS1 Circuit Pack screen is **1C**, SSA selects Nortel Proprietary Supplementary Services.
b	ISO QSIG. Also, use this entry for SBS signaling groups.
c	ETSI (appears only for **Group Type isdn-pri**)
d	ECMA QSIG
e	Allows DCS with rerouting. **DCS with Rerouting** must be **y**, and the **Used for DCS** field on the Trunk Group screen must be **y**.
f	Feature Plus
g	ANSI. Available only if, on the System Parameters Customer-Options (Optional Features) screen, the **ISDN-PRI** or **ISDN-BRI** field is **y** or the **Used for DCS** field is **y**.

Virtual Channel Identifier

Appears when the **Group Type** field is **atm**.

Valid entries	Usage
32 to **1023** or blank	Enter a number between 32 and 1023 or blank.

Virtual Path Identifier

A display-only field that appears when the **Group Type** field is **atm**. This field always sets to **0**.

Valid entries	Usage
32 to **1023** or blank	Enter a number between 32 and 1023 or blank.

X-Mobility/Wireless Type

Appears when the **Group Type** field is **isdn-pri**. This field indicates the type of X-Mobile endpoints allowed.

Valid entries	Usage
DECT	Indicates to Communication Manager that the remote end of the trunk group controlled by the signaling group is a DECT mobility controller. This allows X-Mobility to work over ISDN-PRI trunks between the server/switch and adjunct.
none	

Field descriptions for page 2

This screen appears only when the **Group Type** is **atm**.

Figure 277: Signaling Group screen (when the Group Type field is atm)

```
add signaling-group next                                          Page 2 of x

SIGNALING GROUP

Chan Port Chan Port
  1:       17:
  2:       18:
  3:       19:
  4:       20:
  5:       21:
  6:       22:
  7:       23:
  8:       24:
  9:
 10:
 11:
 12:
 13:
 14:
 15:
 16:
```

The Limit Signaling Group Usage screen is added to allow control of where H.323 trunks are used. This screen appears only when the **Group Type** is **h.323** and **Near-end Node Name** is **procr**.

Figure 278: Limit Signaling Group Usage screen - Enable on Survivable Processor (ESS and LSP) field set to all, ess-all, or none

```
change signaling-group 3                                        Page   2 of   x
                        LIMIT SIGNALING GROUP USAGE

                        Enable on the main Processor(s)? y

          Enable on Survivable Processors (ESS and LSP): all
```

Figure 279: Limit Signaling Group Usage screen - Enable on Survivable Processor (ESS and LSP) field set to selected

```
change signaling-group 3                                        Page   2 of   x
                        LIMIT SIGNALING GROUP USAGE

                        Enable on the main Processor(s)? y

          Enable on Survivable Processors (ESS and LSP): selected

                            Selected Survivable Processor Node Names
                             1:
                             2:
                             3:
                             4:
                             5:
                             6:
                             7:
                             8:
```

Chan Port

Displays when the **Group Type** field is **atm**. If the **Circuit Type** field on page 1 is **T1**, this field displays 24 channels; if you specified **E1**, it displays 31 channels.

You must fill this screen in for ATM signaling groups. This provides two things:

- It allows you to define fractional T1 and fractional E1 facilities, specifying how many and which channels to use.
- It allows you to choose the port numbers to use (port numbers must be unique for all signaling groups on the same ATM board).

The signaling channel (port 16 for an E1 and port 24 for a T1) must be a port between **9** and **32**. A port number used on this screen cannot be used on any other ATM signaling group on the same board.

The channels used must match exactly the channels used on the other end of the signaling group. For example, if your T1 is set up to use channels 1 through 5, 7, and 24 (the signaling channel), the far end must use channels 1 through 5, 7, and 24.

Valid entries	Usage
009 to **256** or blank	Enter the port number for non-signaling channels.

Enable on the main Processor(s)?

Valid entries	Usage
y/**n**	Enter **y** to allow usage of H.323 trunks only on the main server. Default is **y**.

Enable on Survivable Processors (ESS and LSP)

Valid entries	Usage
all, **ess-all**, **none**, **selected**	Enter **y** to allow usage on all survivable processors (LSPs and ESSes). Default is **all**. Enter **ess-all** to allow usage of H.323 trunks on ESSes, but not on LSPs. Enter **none** to block usage of H.323 trunks on survivable processors. If you enter **all**, **ess-all**, or **none**, the screen appears as in Figure 278. Enter **selected** to specify which survivable processors are allowed to use H.323 trunks. If you enter **selected**, additional fields appear to allow you to specify the node names of survivable processors, as shown in Figure 279.

Selected Survivable Processor Node Names

Valid entries	Usage
	Enter the node names of the survivable processors that can use H.323 trunks.

Signaling Group Administered NCA TSC Assignments page

Figure 280: Signaling Group screen (Administered NCA-TSC Assignment Page)

```
                                                          Page 2 of x
                         ADMINISTERED NCA TSC ASSIGNMENT

Service/Feature: _________________   As-needed Inactivity Time-out (min):
  TSC    Local                                                 Adj.     Mach.
 Index    Ext.   Enabled  Established   Dest. Digits    Appl.   Name      ID
   1:    _____     __     _________   _____________   _______ _______   __
   2:    _____     __     _________   _____________   _______ _______   __
   3:    _____     __     _________   _____________   _______ _______   __
   4:    _____     __     _________   _____________   _______ _______   __
   5:    _____     __     _________   _____________   _______ _______   __
   6:    _____     __     _________   _____________   _______ _______   __
   7:    _____     __     _________   _____________   _______ _______   __
   8:    _____     __     _________   _____________   _______ _______   __
   9:    _____     __     _________   _____________   _______ _______   __
  10:    _____     __     _________   _____________   _______ _______   __
  11:    _____     __     _________   _____________   _______ _______   __
  12:    _____     __     _________   _____________   _______ _______   __
  13:    _____     __     _________   _____________   _______ _______   __
  14:    _____     __     _________   _____________   _______ _______   __
  15:    _____     __     _________   _____________   _______ _______   __
```

Appl.

Specifies the application for this administered NCA-TSC.

Valid entries	Usage
audix	Use this for ISDN-PRI D-channel DCS Audix feature.
dcs	Use this for the DCS Over ISDN-PRI D-channel feature.
gateway	Use this when the administered NCA-TSC is used as one end in the gateway channel. If gateway is entered, then the ISDN TSC Gateway Channel Assignments screen must be completed.
masi	Use this when the NCA-TSC is one end of a multimedia application server interface.
qsig-mwi	Use this to convert messages from an administered AUDIX NCA-TSC to a QSIG CISC. If you use this application type, then you must enter a Machine ID between 1 and 20.

As-needed Inactivity Time-out (min)

Valid entries	Usage
10 to **90**, or blank	This field only applies to as-needed NCA-TSCs.

Dest. Digits

Valid entries	Usage
Up to 15 characters **0** to **9**, *****, **#**	Enter the extension of the ISDN interface.

Enabled

Valid entries	Usage
y/n	Enter **y** to enable the administered NCA-TSC.

Established

Used to indicate the strategy for establishing this administered NCA-TSC.

Valid entries	Usage
permanent	Use permanent so that the administered NCA-TSC can be established by either the near end or the far end.
as-needed	Use as-needed so that the administered NCA-TSC will be established the first time the administered NCA-TSC is needed; it can be set up either by the near end or far end switch.

Local Ext

Valid entries	Usage
Extension	Enter the extension of the ISDN interface.

Mach ID

You can enter up to 20 machine IDs.

Valid entries	Usage
1 to **20**	Enter a unique machine ID. The system does not allow you to specify an ID that you already entered on the Processor Channel screen.
1 to **63** for DCS **1** to **99** for Audix **1** to **15** for MASI **1** to **20** for QSIG-MWI blank	For S87XX Series IP-PNC.

Service/Feature

Valid entries	Usage
accunet **i800** **inwats** **lds** **mega800** **megacom** **multiquest** **operator** **sdn** **sub-operator** **wats-max-band**	Enter the service or feature being assigned. In addition to pre-defined Services/Features, any user-defined **Facility Type** of **0** (feature) or **1** (service) on the **Network Facilities** screen is allowed.

SIT Treatment for Call Classification

This screen is available when, on the System Parameters Customer-Options (Optional Features) screen, **ASAI Link Core Capabilities** and **ASAI Link Plus Capabilities** are **y**. Use this screen to specify the treatment of Special Information Tones (SITs) used for Outbound Call Management type calls with USA tone characteristics. The port network TN744 Call Classifier circuit pack ports or H.248 Media Gateway internal tone detector resources in classified mode are used to detect SITs. The classifiers are capable of detecting the following SITs:

- SIT Ineffective Other
- SIT Intercept
- SIT No Circuit
- SIT Reorder
- SIT Vacant Code
- SIT Unknown
- AMD (Answering Machine Detected) Treatment

Field descriptions for page 1

Figure 281: Sit Treatment for Call Classification screen

```
change sit-treatment
                 SIT TREATMENT FOR CALL CLASSIFICATION

               SIT Ineffective Other: dropped
                       SIT Intercept: answered
                      SIT No Circuit: dropped
                         SIT Reorder: dropped
                     SIT Vacant Code: dropped
                         SIT Unknown: dropped

                       AMD Treatment: dropped
                Pause Duration (sec): 0.5
                 Talk Duration (sec): 2.0
```

AMD Treatment

Answering Machine Detected. An ASAI adjunct can request AMD for a call. If Answering Machine is detected, one of two treatments is specified. Default is **dropped**.

Valid entries	Usage
answered	Enter **answered** to specify that these call are classified as answered, and are therefore sent to an agent.
dropped	Enter **dropped** to specify that these calls are classified as not answered, and are therefore not sent to an agent.

AMD Treatment has two separately administrable subfields: **Talk Duration** is for full seconds and **Pause Duration** is for fractions of a second, separated by a display-only decimal point.

Talk Duration : Defaults to 2.0 seconds and allows a range from 0.1 seconds to 5.0 seconds in increments of 0.1 seconds.

Pause Duration : Defaults to 0.5 seconds and allows a range from 0.1 seconds to 2.0 seconds in increments of 0.1 seconds.

Communication Manager looks for voice energy of at least **Talk Duration** seconds. If it finds that much continuous speech, Communication Manager classifies the call as an answering machine. If it finds a pause of duration as long or longer than **Pause Duration** seconds before then, Communication Manager classifies the call as a live person. So the **Talk Duration** timer should be set to a time longer than it takes to say a typical live greeting, for example, “XYZ Corporation,” but shorter than it takes to say a typical answering machine greeting, for example, “We can't answer the phone so please leave a message.” The **Pause Duration** should be set longer than the typical silence between words in an answering machine greeting, but shorter than the typical space between words in a live greeting, for example, “Hello... Hello?.”

SIT Ineffective Other

Sample announcement following this SIT - *You are not required to dial a 1 when calling this number.* Default is **dropped**.

Valid entries	Usage
answered	Enter **answered** to specify that these call are classified as answered, and are therefore sent to an agent.
dropped	Enter **dropped** to specify that these calls are classified as not answered, and are therefore not sent to an agent.

SIT Intercept

Sample announcement following this SIT - *XXX-XXXX has been changed to YYY-YYYY, please make a note of it.* Default is **answered**.

Valid entries	Usage
answered	Enter **answered** to specify that these call are classified as answered, and are therefore sent to an agent.
dropped	Enter **dropped** to specify that these calls are classified as not answered, and are therefore not sent to an agent.

SIT No Circuit

Sample announcement following this SIT - *All circuits are busy, please try to call again later.* Default is **dropped**.

Valid entries	Usage
answered	Enter **answered** to specify that these call are classified as answered, and are therefore sent to an agent.
dropped	Enter **dropped** to specify that these calls are classified as not answered, and are therefore not sent to an agent.

SIT Reorder

Sample announcement following this SIT - *Your call did not go through, please hang up and dial again.* Default is **dropped**.

Valid entries	Usage
answered	Enter **answered** to specify that these call are classified as answered, and are therefore sent to an agent.
dropped	Enter **dropped** to specify that these calls are classified as not answered, and are therefore not sent to an agent.

SIT Unknown

A situation or condition that is unknown to the network is encountered. Default is **dropped**.

Valid entries	Usage
answered	Enter **answered** to specify that these call are classified as answered, and are therefore sent to an agent.
dropped	Enter **dropped** to specify that these calls are classified as not answered, and are therefore not sent to an agent.

SIT Vacant Code

Sample announcement following this SIT - *Your call cannot be completed as dialed, please check the number and dial again.* Default is **dropped**.

Valid entries	Usage
answered	Enter **answered** to specify that these call are classified as answered, and are therefore sent to an agent.
dropped	Enter **dropped** to specify that these calls are classified as not answered, and are therefore not sent to an agent.

Site Data

Use this screen to enter information about buildings, floors and telephone set colors. You must supply values on this screen before you can enter information in the **Site Data** section of the Station screen.

Field descriptions for page 1

Figure 282: Site Data screen

```
change site-data
SITE DATA USER DEFINITION
VALID BUILDING FIELDS



   _________   _________   _________   _________   _________
   _________   _________   _________   _________   _________
   _________   _________   _________   _________   _________
   _________   _________   _________   _________   _________
   _________   _________   _________   _________   _________
```

These pages are available for you to enter descriptive information about the buildings, floors and telephone set colors. You can enter any valid keyboard character. If you want to indicate that a particular floor is in a particular building, you must include this in the floor entry, for example, B301-Fl4.

Station

Use the Station screen to administer individual telephone sets or virtual telephones. This section provides descriptions of all of the fields that can appear on the Station screens. Some of the fields are used for specific telephone types; others are used for all telephone types. The screen examples shown might not show all fields, or might show fields that normally do not appear together; it is not intended to reflect a specific trunk group type. Your own screen might vary from this example according to specific field and system settings. To make it easier to find a specific field description, they are listed in alphabetical order by field name.

Field descriptions for Station screens

Figure 283: Station screen

```
add station next                                              Page 1 of X
                              STATION

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

STATION OPTIONS
      XOIP Endpoint type: auto              Time of Day Lock Table:
              Loss Group: 2           Personalized Ringing Pattern: 3
             Data Module? n                       Message Lamp Ext: 1014
            Speakerphone: 2-way                Mute button enabled? y
        Display Language? English
                   Model:                         Expansion Module?

 Survivable GK Node Name:                        Media Complex Ext:
          Survivable COR:                             IP Softphone? y
   Survivable Trunk Dest?                      Remote Office Phone? y
                                                IP Video Softphone?
                                                          IP Video?

                                               Customizable Labels?
```

Figure 284: Station screen (page 2)

```
add station next                                                  Page   2 of   x
                                     STATION
FEATURE OPTIONS
           LWC Reception: spe           Auto Select Any Idle Appearance? n
          LWC Activation? y                      Coverage Msg Retrieval? y
  LWC Log External Calls? n                                 Auto Answer: none
             CDR Privacy? n                            Data Restriction? n
   Redirect Notification? y                  Idle Appearance Preference? n
 Per Button Ring Control? n                Bridged Idle Line Preference? n
   Bridged Call Alerting? n                    Restrict Last Appearance? y
  Active Station Ringing: single

        H.320 Conversion? n          Per Station CPN - Send Calling Number?
       Service Link Mode: as-needed                   EC500 State: enabled
         Multimedia Mode: basic
                                              Display Client Redirection? n
              AUDIX Name:                    Select Last Used Appearance? n
                                               Coverage After Forwarding? s

                                             Direct IP-IP Audio Connections? y
  Emergency Location Ext: 27000                       IP Audio Hairpinning? n
```

Figure 285: Station screen (page 3)

```
add station next                                                       Page 3 of x
                                    STATION

          Conf/Trans on Primary Appearance? y
Bridged Appearance Origination Restriction? y
            Call Appearance Display Format: loc-param-default
     IP Phone Group ID:
                             ENHANCED CALL FORWARDING
                                      Forwarded Destination        Active
Unconditional For Internal Calls To:                                  n
                  External Calls To:                                  n
         Busy For Internal Calls To:                                  n
                  External Calls To:                                  n
     No Reply For Internal Calls To:                                  n
                  External Calls To:                                  n

           SAC/CF Override? n
```

Figure 286: Station screen (page 4)

```
add station nnnn                                                Page 4 of X
                                    STATION

     SITE DATA
          Room: _______                                     Headset? n
          Jack: ___                                         Speaker? n
         Cable: ___                                        Mounting: d
         Floor: _______                                 Cord Length: 0_
      Building: _______                                   Set Color: _____


ABBREVIATED DIALING
     List1: ________           List2: _________          List3: _________

BUTTON ASSIGNMENTS
 1: call-appr                            7: Team   Ext: 5381231         Rg:n
 2: call-appr                            8: Team   Ext: 5381232         Rg:s
 3: call-appr                            9: Team   Ext: 5385678         Rg:c
 4: _________                           10: Team   Ext: 5385679         Rg:d
 5: _________                           11: Team   Ext: 5385676         Rg:i
 6: _________                           12: Team-auto-ans

voice-mail Number:
```

Figure 287: Station screen (page 5)

```
change station nnnn                                             Page 5 of x
                                      STATION

FEATURE BUTTON ASSIGNMENTS

 9:
10:
11:
12:
13:
14:
15:
16:
17:
18:
19:
20:
21:
22:
23:
24:
```

If the **Expansion Module** field is **y**, an additional page appears.

Figure 288: Station screen (page 6)

```
change station nnnn                                              Page 6 of x
                                  STATION
EXPANSION MODULE BUTTON ASSIGNMENTS

1:                                         13:
2:                                         14:
3:                                         15:
4:                                         16:
5:                                         17:
6:                                         18:
7:                                         19:
8:                                         20:
9:                                         21:
10:                                        22:
11:                                        23:
12:                                        24:
```

Figure 289: SIP Feature Options page

```
change station nnnn                                               Page 5 of x
                                  STATION

SIP Feature Options
         Type of 3PCC Enabled: none
```

The field descriptions for the Station screen are listed alphabetically for easy reference.

1-Step Clearing

Valid entries	Usage
y/n	If set to **y**, the call terminates again at the WCBRI terminal when the user drops from the call.

Abbreviated Dialing List1, List2, List3

You can assign up to 3 abbreviated dialing lists to each telephone.

Valid entries	Usage
enhanced	Allows the telephone user to access the enhanced system abbreviated dialing list.
group	Allows the telephone user to access the specified group abbreviated dialing list. If you enter **group**, you also must enter a group number.
personal	Allows the telephone user to access and program their personal abbreviated dialing list. If you enter **personal**, you also must enter a personal list number.
system	Allows the telephone user to access the system abbreviated dialing list.

Access Code

This field appears when a wireless terminal model number is selected in the **Type** field. The Access Code is a temporary, shorter version of the complete User Authentication Key (UAK) required by the system when the terminal is first put into service. It is then used to automatically generate a unique UAK for that wireless terminal over-the-air.

Valid entries	Usage
5-digit decimal number	Enter the 5-digit access code to place the wireless terminal into service. Default is blank.

Active Station Ringing

Defines how call rings to the telephone when it is off-hook. This field does not affect how calls ring at this telephone when the telephone is on-hook.

Valid entries	Usage
continuous	Enter **continuous** to cause all calls to this telephone to ring continuously.
single	Enter **single** to cause calls to this telephone to receive one ring cycle and then ring silently.

Valid entries	Usage
if-busy-single	Enter **if-busy-single** to cause calls to this telephone to ring continuously when the telephone is off-hook and idle and calls to this telephone to receive one ring cycle and then ring silently when the telephone is off-hook and active.
silent	Enter **silent** to cause all calls to this station to just ring silently.

Adjunct Supervision

Adjunct Supervision appears when the **Type** field is **500**, **2500**, **k2500**, **8110**, **ops**, **ds1fd**, **ds1sa**, **VRU**, **VRUFD**, or **VRUSA**.

Valid entries	Usage
y	Enter **y** if an analog disconnect signal is sent automatically to the port after a call terminates. Analog devices (such as answering machines and speakerphones) use this signal to turn the devices off after a call terminates.
n	Set this field to **n** so hunt group agents are alerted to incoming calls. In a hunt group environment, the disconnect signal blocks the reception of zip tone and incoming call notification by an auto-answer station when a call is queued for the station.

Always Use

This field does not apply to SCCAN wireless telephones, or to extensions administered as type **h.323**.

Valid entries	Usage
y	When this field is **y**: ● The **Remote Softphone Emergency Calls** field is hidden. A softphone can register no matter what emergency call handling settings the user has entered into the softphone. If a softphone dials 911, the **Emergency Location Extension** administered on the Station screen is used. The softphone's user-entered settings are ignored. ● If an IP telephone dials 911, the **Emergency Location Extension** administered on the Station screen is used. ● If a call center agent dials 911, the physical station extension is displayed, overriding the **LoginID for ISDN Display** field on the Agent LoginID screen.
n	For more information, see the description for the **Emergency Location Extension** field on the Station screen. This is the default,

Assigned Member — Ext

The system automatically assigns this extension. This is the extension of the user who has an associated **Data Extension** button and shares the module.

Assigned Member — Name

Display-only field that shows the name associated with the extension shown in the **Assigned Member - Ext** field.

Att. Call Waiting Indication

Attendant call waiting allows attendant-originated or attendant-extended calls to a busy single-line telephone to wait and sends distinctive call-waiting tone to the single-line user. Must be set to **y** when the **Type** field is set to **H.323**. You should not set this field to **y** if the **Data Restriction** field is **y** or the **Switchhook Flash** field is **n**, or if Data Privacy is enabled for the telephone's class of service (COS). If any of these conditions are true, the telephone cannot accept or handle call waiting calls.

Valid entries	Usage
y	Enter **y** to activate Call Waiting (without Caller ID information) for the telephone. Default.
n	Call Waiting is not enabled for the station.
c	Enables the Caller ID Delivery with Call Waiting feature, which displays CID information on for the waiting call. This value can only be entered when the **Type** field is **CallrID**.

Audible Message Waiting

The Audible Message Waiting tone indicates that the user has a waiting message. This field appears only if **Audible Message Waiting** is set to **y** on the System Parameters Customer-Options (Optional Features) screen.

Note that this field does not control the Message Waiting lamp.

Valid entries	Usage
y/n	Enter **y** if you want the telephone user to receive stutter dial tone when they have a waiting message and they go off-hook.

Audix Name

Specifies which AUDIX is associated with the station.

Valid entries	Usage
Names assigned to an AUDIX adjunct	Must contain a user-defined adjunct name that was previously administered on the **IP Node Names** screen.

Auto-A/D

When **Per Button Ring Control** is **y**, this field appears next to the **call-appr** field in the **BUTTON ASSIGNMENTS** section of the Station screen. Use this field to enable automatic abbreviated/delayed ringing for a call appearance.

Valid entries	Usage
y/n	Enter **y** if you want to enable abbreviated/delayed ringing for this call appearance. Default is **n**.

Auto Answer

In EAS environments, the auto answer setting on the Agent LoginID screen can override a station's setting when an agent logs in there.

Note:
For analog stations, if Auto Answer is set to **acd** and the station is off-hook and idle, only the ACD split/skill calls and direct agent calls auto answer; non-ACD calls receive busy treatment. If the station is active on an ACD call and a non-ACD call arrives, the Agent receives call-waiting tone.

Valid entries	Usage
all	Enter all to allow all calls (ACD and non-ACD) terminated to an idle station to be cut through immediately. Does not allow automatic hands-free answer for intercom calls. With non-ACD calls, the set is also rung while the call is cut through. The ring can be prevented by activating the **ringer-off** feature button when, on the Feature-Related System Parameters screen, the **Allow Ringer-off with Auto-Answer** field is **y**.
acd	Enter **acd** to allow only ACD split /skill calls and direct agent calls to auto answer. If this field is set to **acd**, Non-ACD calls terminated to a station ring audibly.
none	Enter **none** to cause all calls terminated to this station to receive an audible ringing treatment.
icom	Enter **icom** to allow a telephone user to answer an intercom call from the same intercom group without pressing the **intercom** button.

Automatic Moves

Automatic Moves allows a DCP telephone to be unplugged from one location and moved to a new location without additional Communication Manager administration. Communication Manager automatically associates the extension to the new port.

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

Valid entries	Usage
always	Enter **always** and the DCP telephone can be moved anytime without additional administration by unplugging from one location and plugging into a new location.
once	Enter **once** and the DCP telephone can be unplugged and plugged into a new location once. After a move, the field is set to done the next time that routine maintenance runs on the DCP telephone. Use **once** when moving a large number of DCP telephones so each extension is removed from the move list. Use **once** to prevent automatic maintenance replacement.
no	Enter **no** to require administration in order to move the DCP telephone.
done	Done is a display-only value. Communication Manager sets the field to done after the telephone is moved and routine maintenance runs on the DCP telephone.
error	Error is a display-only value. Communication Manager sets the field to error, after routine maintenance runs on the DCP telephone, when a non-serialized telephone is set as a movable telephone.

Auto Select Any Idle Appearance

Valid entries	Usage
y/n	Enter **y** to allow automatic selection of any idle appearance for transferred or conferenced calls. Communication Manager first attempts to find an idle appearance that has the same extension number as the call being transferred or conferenced has. If that attempt fails, Communication Manager selects the first idle appearance.

Automatic Selection of DID Numbers

Communication Manager chooses a 2- to 5-digit extension from a predetermined list of numbers and assigns the extension to a hotel room telephone.

Valid entries	Usage
y/n	Enter **y** to use the Automatic Selection of DID Numbers for Guest Rooms feature.

BCC

Appears when **ISDN-PRI** or **ISDN-BRI Trunks** is enabled on the System Parameters Customer-Options (Optional Features) screen. Display-only field set to **0** for stations (that is, indicates voice or voice-grade data).

See Generalized Route Selection in *Avaya Aura™ Communication Manager Feature Description and Implementation,* 555-245-205, for more information on Bearer Capability Classes (BCC) and their ability to provide specialized routing for various types of voice and data calls. The BCC value is used to determine compatibility when non-ISDN facilities are connected to ISDN facilities (ISDN Interworking).

Bearer

This field is useful when Secure Terminal Equipment (STE) telephones are administered as 8510 telephones. This field appears on the BRI Station screen for 8503, 8510, and 8520 stations in Communication Manager 2.1 and 2.2 only. See Secure Terminal Equip on page 835 for **Bearer** field functionality in Communication Manager 3.0 and later.

Valid entries	Usage
speech	Force the Bearer Cap IE to "speech" before a call is delivered to the 85xx BRI station.
3.1khz	Leave the Bearer Cap IE unchanged. Use 3.1khz to let secure calls from Secure Terminal Equipment (STE) telephones to work properly.

Bridged Appearance Origination Restriction

.

Valid entries	Usage
y	Call origination on the bridged appearance is restricted.
n	Call origination ion the bridged appearance is allowed. This is normal behavior, and is the default.

Bridged Call Alerting

Use this field to control how the user is alerted to incoming calls on a bridged appearance.

If **Bridged Call Alerting** is **n** and **Per Button Ring Control** is **n**, audible ringing is suppressed for incoming calls on bridged appearances of another telephone's primary extension.

Valid entries	Usage
y	Enter **y** if you want the bridged appearance to ring when a call arrives at the primary telephone.
n	Enter **n** if you want the bridged appearance to flash but not ring when a call arrives at the primary telephone. This is the default.

Bridged Idle Line Preference

Use this field to specify that the line that the system selects when you go off hook is always an idle call appearance for incoming bridged calls.

Valid entries	Usage
y	If you enter **y**, the user connects to an idle call appearance instead of the ringing call.
n	If you enter **n**, the user connects to the ringing bridged appearance.

Building

Enter a valid building location. See Site Data on page 789 for valid entries.

Note:
Fields in the SITE DATA area of the **Station** screen document information related to the station set installation.

Busy Auto Callback without Flash

Appears on the Station screen for analog telephones, only if the **Without Flash** field in the **ANALOG BUSY AUTO CALLBACK** section of the Feature-Related System Parameters screen is set to **y**. The **Busy Auto Callback without Flash** field then defaults to **y** for all analog telephones that allow Analog Automatic Callback.

Valid entries	Usage
y/n	Enter **y** to provide automatic callback for a calling analog station without flashing the hook.

BUTTON ASSIGNMENTS

Enter the abbreviated software name to assign a feature button. For a list of feature buttons, see *Administering Avaya Aura™ Communication Manager*, 03-300509.

Note:
If you want to use Terminal Translation Initialization (TTI), you must assign a call appearance (**call-appr**) to the first button position. TTI needs the button on the first call appearance to get dial tone.

Cable

You can use this field to identify the cable that connects the telephone jack to the system. You also can enter this information in the Blank column on the Port Assignment Record.

Note:
Fields in the SITE DATA area of the **Station** screen document information related to the station set installation.

Call Appearance Display Format

This field only appears on telephones that support downloadable call appearance buttons, such as the 2420 and 4620 telephones. Bridged call appearances are not affected by this field. Use this field to specify the display format for the station.

Note:
This screen sets the display value for an individual station. To set display values for an entire location, see the **Display Parameters** screen.

Valid entries	Usage
disp-param-default	The system uses the default value from the **Display Parameters** screen that applies to this station.This is the default.
inter-location	The system displays the complete extension on downloadable call appearance buttons.
intra-location	The system displays a shortened or abbreviated version of the extension on downloadable call appearance buttons.

Caller ID Message Waiting Indication

Appears when the **Type** field is **CallrID**. For CallrID type telephones or analog telephones with Caller ID adjuncts only.

Valid entries	Usage
y/n	Enter **y** to allow aliasing of various non-Avaya telephones and adjuncts.

Note:
The Caller ID Message Waiting Indication administration is independent of the administration of LED or NEON-lamp Communication Manager Message Waiting Indication (MWI). For example, it is possible to administer a Caller ID telephone with the **Caller ID Message Waiting Indication** field set to **n** and the **Message Waiting Indicator** field set to **neon**.

Calls Allowed

Appears if the **XMOBILE Type** field is **EC500** and the **Mapping Mode** field is **termination** or **both**. Used to identify the Extension to Cellular call filter type for an XMOBILE station. This field allows an XMOBILE station to function as a bridge and still be restricted.

Valid entries	Usage
internal	External calls are blocked. Internal calls terminate to the XMOBILE station. Attendant-originated and attendant-delivered calls are not delivered
external	Internal calls are blocked. External calls terminate to the XMOBILE station.

Valid entries	Usage
all	All calls terminate to the XMOBILE station.
none	Prevents calls from terminating to the XMOBILE station. Can be used to prevent business-related calls from accruing telephone charges on cellular telephones that are lost, being transferred to a new user, or being disabled for other business reasons.

Note:
Interswitch calls on DCS trunks are treated as internal calls. When the **Calls Allowed** field is set to **internal** or **all**, DCS calls are delivered to the cell telephone. When the **Calls Allowed** field is set to **external** or **none**, DCS calls are not delivered.

Incoming calls from other Extension to Cellular users are internal if office caller ID is enabled for the XMOBILE station associated with the cell telephone. When the **Calls Allowed** field is set to **internal** or **all**, calls from other Extension to Cellular users are delivered. When the **Calls Allowed** field is set to **external** or **none**, calls from other Extension to Cellular users are not delivered.

The calling party receives busy treatment when call filtering blocks calls to a standalone XMOBILE. Calls delivered to standalone XMOBILE stations that are not answered will follow the call coverage or call forwarding paths administered for the standalone XMOBILE.

Call Waiting Indication

This allows user, attendant-originated, and outside calls to a busy single-line telephone to wait and sends a distinctive call-waiting tone to the single-line user. This feature is denied if **Data Restriction** is **y** or **Switchhook Flash** is **n**, or if Data Privacy is active by way of the telephone COS assignment. Must be set to **y** when the **Type** field is set to **H.323**.

Valid entries	Usage
y	Enter **y** to activate Call Waiting (without Caller ID information) for the telephone. Default.
n	Call Waiting is not enabled for the station.
c	Enables the Caller ID Delivery with Call Waiting feature, which displays CID information on for the waiting call. This value can only be entered when the **Type** field is **CallrID**.

CDR Privacy

This option allows digits in the called number field of an outgoing call record to be blanked, on a per-station basis. You administer the number of blocked digits system-wide in the **Privacy - Digits to Hide** field on the CDR System Parameters screen.

Valid entries	Usage
y/n	Enter **y** to enable Call Privacy for each station.

Cell Phone Number

Contains the unformatted cell telephone's published external number. This field can contain a 3-digit area code plus the 7-digit main number. If the same Cell Phone Number is administered on multiple XMOBILE Station screens, then the Dial Prefix associated with each instance of the Cell Phone Number must be the same.

Valid entries	Usage
0 to **9**	Enter 1 to15 digits. Avaya recommends that you enter a full 10-digit Cell Phone Number regardless of whether the cell telephone is local or not. Note that your ARS screen has to be administered to handle this.

Conf/Trans On Primary Appearance

This feature forces the use of a primary appearance when the held call to be conferenced or transferred is a bridge. This is regardless of the **Auto Select Any Idle Appearance** field.

Valid entries	Usage
y/n	Enter **y** to specify that the primary call appearance is always activated for a bridged transfer or conference.

Configuration Set

This field is used to administer the Configuration Set number that contains the call treatment options desired for the XMOBILE station. This field must be administered if:

- The **XMOBILE Type** field is **EC500**.
- The **Mobility Trunk Group** field is a trunk group number and the administered trunk group is non-DECT or non-PHS.
- The **Mobility Trunk Group** field is **aar** or **ars**.

If the **Mobility Trunk Group** field is a trunk group number and the administered trunk group is **DECT** or **PHS**, this field can be left blank.

Valid entries	Usage
1 to **10** or blank	Enter any value corresponding to the appropriate Configuration Set screen.

COR

Enter a Class of Restriction (COR) number to select the desired restriction.

Cord Length

The length of the cord attached to the receiver. This is a free-form entry, and can be in any measurement units.

Note:
Fields in the SITE DATA area of the **Station** screen document information related to the station set installation.

COS

Enter the desired Class of Service (COS) number to select allowed features.

Country Protocol

Enter the protocol that corresponds to your supported initialization and codesets. The Country Protocol must match any previously-administered endpoint on the same port.

Valid entries	Usage
1	United States (Bellcore National ISDN)
2	Australia
etsi	ETSI (Europe)
3	Japan
6	Singapore

Coverage After Forwarding

This field governs whether an unanswered forwarded call is provided coverage treatment.

Valid entries	Usage
y	Coverage treatment is provided after forwarding regardless of the value of the **Coverage After Forwarding** field on the System Parameters - Call Coverage/Call Forwarding screen.
n	No coverage treatment is provided after forwarding regardless of the value of the **Coverage After Forwarding** field on the System Parameters - Call Coverage/Call Forwarding screen.
s(ystem)	Indicates that call processing uses the **Coverage After Forwarding** field on the System Parameters Call Coverage/Call Forwarding screen. To override the system-wide parameter for a given station, set this field to **y** or **n**.

Coverage Msg Retrieval

Applies if the telephone is enabled for LWC Reception.

Valid entries	Usage
y/n	Enter **y** to allow users in the telephone's Coverage Path to retrieve Leave Word Calling (LWC) messages for this telephone.

Coverage Module

Valid entries	Usage
y	Enter **y** to indicate that a coverage module is connected to the station. Once you enter **y**, the system displays an additional page that allows you to assign the buttons for the module.

Coverage Path 1 or Coverage Path 2

Enter a coverage-path number or time-of-day table number from a previously-administered Call Coverage Path screen or Time of Day Coverage Table screen.

Note:
If Modified Misoperation is active (**Misoperation Alerting** is **y** on the Feature-Related System Parameters screen), you must assign a Coverage Path to all stations on Communication Manager.

CRV Length

Only for ASAI stations. Enter **1** or **2** to indicate the length of CRV for each interface.

Custom Selection of VIP DID Numbers

Custom Selection of VIP DID numbers allows you to select the DID number assigned to a room when a guest checks in.

Valid entries	Usage
y/n	Enter **y** to allow you to select the DID number assigned to a room when a guest checks in.

Customizable Labels

Use this field to enable the Increase Text for Feature Buttons feature for this station. This feature expands the text labels associated with Abbreviated Dial buttons from the current five uppercase alphanumeric characters to a maximum of 13 upper and lower case alphanumeric characters.This field allows you to ensure that there will always be sufficient button customization resources to support VIP users. This field appears when **Type** is one of the following:

- 2410 (Release 2 or later)
- 2420 (Release 4 or later)
- 4610 (IP Telephone Release 2.2 or later)
- 4620 (IP Telephone Release 2.2 or later)
- 4621 (IP Telephone Release 2.2 or later)
- 4622 (IP Telephone Release 2.2 or later)
- 4625 (IP Telephone Release 3.1 or later)

Valid entries	Usage
y	Enter **y** to allow the user of this station to program and store feature button labels with up to 13 alphanumeric characters. This is the default.
n	Enter **n** to disable the Increase Text for Feature Buttons feature for this station.

Data Extension

Enter the extension assigned to the data module.

Data Module

Valid entries	Usage
y/n	Enter **y** if this telephone has an associated data module. When set to **y**, a Data Module page is added to the Station screen for defining the data module parameters.

Data Option

Valid entries	Usage
analog **data module** **none**	If a second line on the telephone is administered on the I-2 channel, enter **analog**. Otherwise, enter **data module** if applicable or **none**.

Data Restriction

Data restriction provides permanent protection and cannot be changed by the telephone user. Do not assign a Data Restriction if **Auto Answer** is **all** or **acd**. If **y**, whisper page to this station is denied.

Valid entries	Usage
y/n	Enter **y** to prevent tones, such as call-waiting tones, from interrupting data calls.

Default Dialing Abbreviated Dialing Dial Code

Appears only when the **Special Dialing Option** is set to **default**. Enter a list number associated with the abbreviated dialing list.

When the user goes off-hook for a data call and presses the **Enter** button following the DIAL prompt, the system dials the AD number. The data call originator also can perform data-terminal dialing by specifying a dial string that might or might not contain alphanumeric names.

Dial Prefix

Contains the unformatted sequence of digits or characters that are prepended to the cell telephone's published cell telephone number before dialing. If the same Cell Phone Number is administered on multiple XMOBILE Station screens, then the Dial Prefix associated with each instance of the Cell Phone Number must be the same.

Valid entries	Usage
up to 4 digits: **0** to **9**, *****, **#**	Enter 1 to 4 digits.

Direct IP-IP Audio Connections

Allows direct audio connections between IP endpoints.

Valid entries	Usage
y/n	Enter **y** to save on bandwidth resources and improve sound quality of voice over IP transmissions.

Display Caller ID

Appears when the **Type** field is **CallrID**. For CallrID type telephones or analog telephones with Caller ID adjuncts only.

Valid entries	Usage
y/n	Enter **y** to allow transmission of calling party information to the Caller ID telephone or adjunct.

Display Cartridge

For 7404 D telephones only. Enter **y** to indicate there is a display cartridge associated with the station. This displays an additional page to allow you to assign display buttons for the display cartridge.

Display Client Redirection

Only administrable if **Hospitality** is enabled on the System Parameters Customer-Options (Optional Features) screen. This field affects the telephone display on calls that originated from a station with Client Room Class of Service.

Note:
For stations with an audix station type, AUDIX Voice Power ports, or ports for any other type of messaging that needs display information, **Display Client Redirection** must be set to **y**.

Valid entries	Usage
y	When set to **y**, the redirection information for a call originating from a Client Room and terminating to this station displays.
n	When set to **n**, this station's display does not show the redirection information for all calls originating from a Client Room (even redirected calls) that terminate to this station. Only the client name and extension (or room, depending on what is administered on the **Hospitality** screen) display.

Display Language

Use this field to specify the language in which information is displayed on stations. To view the dial pad letter/number/symbol mapping tables used for the integrated directory, see Telephone Display in *Avaya Aura™ Communication Manager Feature Description and Implementation*, 555-245-205.

Tip:
Time of day is displayed in 24-hour format (00:00 - 23:59) for all languages except **english**, which is displayed in 12-hour format (12:00 a.m. to 11:59 p.m.).To display time in 24-hour format and display messages in English, set the **Display Language** field to **unicode**. When you enter **unicode**, the station displays time in 24-hour format, and if no Unicode file is installed, displays messages in English by default. For more information on Unicode, see *Administering Avaya Aura™ Communication Manager*, 03-300509.

Valid entries	Usage
english	Enter the language you want the user to see on their station display.
french	
italian	
spanish	
	1 of 2

Valid entries	Usage
user-defined	
unicode	**Note:** Unicode display is only available for Unicode-supported telephones. Currently, 4610SW, 4620SW, 4621SW, 4622SW, Sage, Spark, and 9600-series telephones (Avaya one-X Deskphone Edition SIP R2 or later) support Unicode display. Unicode is also an option for DP1020 (aka 2420J) and SP1020 (Toshiba SIP Phone) telephones when **Display Character Set** on the **System Parameters Country-Options** screen is set to **katakana**. To administer Unicode on the SP1020, use the 4624 station type. For more information, see *Administering Avaya Aura™ Communication Manager*, 03-300509.
	2 of 2

Distinctive Audible Alert

Valid entries	Usage
y/n	Enter **y** so the telephone can receive the 3 different types of ringing patterns which identify the type of incoming calls. Distinctive ringing might not work properly for off-premises telephones.

Emergency Location Ext

The **Emergency Location Ext** field defaults to the telephone's extension. This extension identifies the street address or nearby location when an emergency call is made. For more information about the use of this field, see the **Usage** description for the **Remote Softphone Emergency Calls** field later in this section.

Valid entries	Usage
Valid extension	Enter the Emergency Location Extension for this station.

Note:
On the ARS Digit Analysis Table screen, you must administer 911 to be call type **emer** or **alrt** in order for the E911 Emergency feature to work properly.

EMU Login Allowed

Valid entries	Usage
y/n	Enter **y** to allow the station to be used as a visited station by an Enterprise Mobility User (EMU) visitor user. Default is **n**. For more information about Enterprise Mobility User, see *Administering Avaya Aura™ Communication Manager*, 03-300509.

Endpt ID

Appears only if **Endpt Init** is **y**. Enter a unique 2-digit number (**00** to **62**) for this endpoint. Each **Endpt ID** field must have a unique value for each endpoint on the same port.

This field provides for multipoint configuration conformance to the Bellcore Terminal Initialization procedures. In these procedures, a multipoint configuration requires the last 2 digits of the Service Profile Identifier (SPID) be between 00 and 63 and be binary unique for each endpoint.

For WorldClass BRI (WCBRI) data modules only, this field, combined with the SPID, gives the effective SPID administered into the terminal. Bellcore ISDN-1 requires the SPID programmed into the endpoint contain at least 9 digits. For example, if the SPID is **1234**, and **Endpt ID** is **01**, then the SPID administered on the terminal is 000123401. The three leading zeros are necessary to create a 9-digit SPID.

Endpt Init

Endpoint initialization is a procedure, required for multipoint operation, by which User Service Order Profile (USOP) is associated with an endpoint on the ISDN-BRI. This association is made via the SPID, administered into the system, and entered into the ISDN-BRI terminal. For an ISDN-BRI terminal to be operational in a multipoint configuration, both the administered SPID and the SPID programmed into the ISDN-BRI terminal must be the same. Therefore, the SPID of new or reused terminals must be programmed to match the administered SPID value. Appears only if **MIM Support** is **y** and indicates the terminal's endpoint initialization capability.

Valid entries	Usage
y	Enter **y** if the terminal supports Bellcore ISDN-1 terminal initialization procedures.
n	Enter **n** for all other country protocols.

Expansion Module

When this field is **y**, an **Expansion Module Buttons Assignment** page is added to the Station screen for administering buttons for the expansion module.

Valid entries	Usage
y/n	Enter **y** if this telephone has an expansion module. This enables you to administer the buttons for the expansion module.

Extension

Displays the extension for this station—this is either the extension you specified in the station command or the next available extension (if you used **`add station next`**).

For a virtual extension, enter a valid physical extension or a blank. Blank allows an incoming call to the virtual extension to be redirected to the virtual extension's "busy" or "all" coverage path.

> Avaya recommends that you consider the display for emergency notification when you complete the **Name** field on the Station screen. Put the most important identifying information at the beginning of the field. When an emergency call is made and a crisis alert station with a 27-character display is notified, only 17 characters of the **Name** field appear on the first display line, followed by the extension. The second line contains the last 3 characters of the **Name** field, followed by the word “EMERGENCY.” Characters 18 through 24 of the **Name** field do not appear at all.

Feature Module

Enter **y** to indicate the station is connected to a feature module. Entering **y** displays an additional page to allow you to assign feature buttons to the module.

Fixed TEI

This field appears only for ISDN-BRI data modules, NI-BRI telephones, WCBRI data modules, and ASAI links. It indicates that the endpoint has a fixed Terminal Endpoint Identifier (TEI).

The TEI identifies a unique access point within a service. You must administer TEIs for fixed TEI terminals. However, for terminals with the automatic TEI capability, the system dynamically assigns the TEI.

Valid entries	Usage
y/n	Entering **y** displays the **TEI** field. For **ASAI**, enter **y**.

Floor

The floor location of this station.

Note:
Fields in the SITE DATA area of the **Station** screen document information related to the station set installation.

Forwarded Destination

For each of the three types of enhanced call forwarding (**Unconditional**, **Busy**, and **No Reply**), enter the destination extension for both internal and external calls. Valid entries can be up to 18 digits, and the first digit can be an asterisk (*). Blank is also a valid entry. In the **Active** field, indicate whether the specific destination is active (**y**) or inactive (**n**).

H.320 Conversion

Allows H.320 compliant calls made to this telephone to be converted to voice-only. Because the system can only handle a limited number of conversion calls, you might need to limit the number of telephones with H.320 conversion.

Headset

Indicates if the telephone has a headset.

Note:
Fields in the SITE DATA area of the **Station** screen document information related to the station set installation.

Home

This field appears when a wireless terminal model number is selected in the **Type** field.

Valid entries	Usage
y/n	Indicate the roaming status of the wireless user. This field will specify whether the system is the user's home or roaming system. The default is **y**.

HOT LINE DESTINATION — Abbreviated Dialing Dial Code

Appears only when **Special Dialing Option** is **hot-line**.

Use Hot Line Service when very fast service is required and when you use a telephone only for accessing a certain facility. Hot Line Service allows single-line telephone users, by simply lifting the handset, to automatically place a call to a preassigned destination (extension, telephone number, or feature access code).

The Hot Line Service destination number is stored in an Abbreviated Dialing List. When the user goes off-hook on a Data Hot Line call, the system dials the AD number.

A Direct Department Calling (DDC), a Uniform Call Distribution (UCD), a Terminating Extension Group (TEG) extension, or any individual extension within a group can be a Hot Line Service destination. Also, any extension within a DDC group, UDC group, or TEG can have Hot Line Service assigned.

Loudspeaker Paging Access can be used with Hot Line Service to provide automatic access to paging equipment.

HOT LINE DESTINATION — Abbreviated Dialing List Number

Enter the abbreviated dialing list where you stored the hotline destination number.

HOT LINE DESTINATION — Dial Code

Enter the dial code in the specified abbreviated dialing list where you stored the hotline destination number.

Hunt-to Station

Enter the extension the system should hunt to for this telephone when the telephone is busy. This field allows you to create a station hunting chain (by assigning a hunt-to station to a series of telephones).

Idle/Active Ringing (Callmaster)

Defines how call rings to the telephone when it is on-hook. This field applies to CALLMASTER telephones.

Valid entries	Usage
continuous	Enter **continuous** to cause all calls to this telephone to ring continuously.
if-busy-single	Enter **if-busy-single** to cause calls to this telephone to ring continuously when the telephone is off-hook and idle and calls to this telephone to receive one ring cycle and then ring silently when the telephone is off-hook and active.

Valid entries	Usage
silent-if-busy	Enter **silent-if-busy** to cause calls to ring silently when this station is busy.
single	Enter **single** to cause calls to this telephone to receive one ring cycle and then ring silently.

Idle Appearance Preference

Use this field to specify that the selected line for incoming calls is always an idle line.

Valid entries	Usage
y	If you enter **y**, the user connects to an idle call appearance instead of the ringing call.
n	If you enter **n**, the Alerting Appearance Preference is set and the user connects to the ringing call appearance.

Ignore Rotary Digits

If this field is **y**, the short switch-hook flash (50 to 150) from a 2500-type set is ignored.

Valid entries	Usage
y	Enter **y** to indicate that rotary digits from the set should be ignored.
n	Enter **n** to make sure they are not ignored.

IPEI

International Portable Equipment Identifier. This field appears when a wireless terminal model number is selected in the **Type** field.

Valid entries	Usage
9-character hexadecimal number; **0** to **9**, **a** to **f**, **A** to **F**, or blank	Enter the unique ID number of the wireless terminal.

IP Audio Hairpinning

Appears when Group Type is **h.323** or **sip**. Allows IP endpoints to be connected through the IP circuit pack in the server, without going through the time division multiplexing (TDM) bus.

Valid entries	Usage
y/n	Enter **y** to allow IP endpoints to be connected through the IP circuit pack in the server/switch in IP format, without going through the Avaya DEFINITY TDM bus. Default is **n**.

IP Phone Group ID

Appears for H.323 station types.

Valid entries	Usage
0 to **999** or blank	Enter the Group ID number for this station.

IP Softphone

Appears only for DCP station types and IP Telephones.

Valid entries	Usage
y/n	Enter **y** indicate that this extension is either a PC-based multifunction station or part of a telecommuter complex with a call-back audio connection.

IP Video

Appears when one of the following conditions are satisfied:

- when station type is **h.323.**
- when the **IP Softphone?** field is enabled (the station type can support a softphone) and **Multimedia IP SIP Trunking?** is enabled in the system-parameters customer-options screen. In this case, the **IP Video** field is replaced by the **IP Video Softphone** field.

Valid entries	Usage
y/n	Enter **y** to indicate that this extension has IP video capability.

IP Video Softphone

Appears only when **IP Softphone?** is **y**. For more information, see IP Video.

Valid entries	Usage
y/n	Enter **y** to indicate that this extension is a video softphone.

ITC (Information Transfer Capability)

Indicates the type of transmission facilities to be used for ISDN calls originated from this endpoint. The field does not display for voice-only or BRI stations.

Valid entries	Usage
restricted	Either restricted or unrestricted transmission facilities are used to complete the call. A restricted facility is a transmission facility that enforces 1's density digital transmission (that is, a sequence of 8 digital zeros are converted to a sequence of 7 zeros and a digital 1).
unrestricted	Only unrestricted transmission facilities are used to complete the call. An unrestricted facility is a transmission facility that does not enforce 1's density digital transmission (that is, digital information is sent exactly as is).

Jack

Alpha-numeric identification of the jack used for this station.

Note:
Fields in the SITE DATA area of the **Station** screen document information related to the station set installation.

Lock Messages

Valid entries	Usage
y/n	Enter **y** to restrict other users from reading or canceling the voice messages or retrieving messages via Voice Message Retrieval.

Loss Group

This field determines which administered 2-party row in the loss plan applies to each station. Does not appear for stations that do not use loss (for example, x-mobile stations and MASI terminals).

Valid entries	Usage
1 to **17**	Shows the index into the loss plan and tone plans.

LWC Activation

Valid entries	Usage
y/n	Enter **y** to allow internal telephone users to leave short LWC messages for this extension. If the system has hospitality, enter **y** for guest-room telephones if the extension designated to receive failed wakeup messages should receive LWC messages that indicate the wakeup calls failed. Enter **y** if **LWC Reception** is **audix**.

LWC Log External Calls

Appears only where the **LWC Reception** field is available. When external calls are not answered, Communication Manager keeps a record of up to 15 calls (provided information on the caller identification is available) and the telephone's message lamp lights. The telephone set displays the names and numbers of unsuccessful callers.

Valid entries	Usage
y/n	Enter **y** to make unanswered external call logs available to end-users. Each record consists of the latest call attempt date and time.

LWC Reception

Valid entries	Usage
audix	Enter **audix** if the messages are stored on the Audio Information Exchange System.

Valid entries	Usage
none	Enter **none** if LWC messages are not stored.
spe	Enter **spe** if LWC messages are stored in the system or on the switch processor element (spe).

Mapping Mode

Controls the mode of operation in which the cell telephone operates when mapped to this XMOBILE extension. An XMOBILE station can be bridged to a deskset. These restrictions/ modes exist because the COR of a bridge is ignored; instead the principal's COR is used. This field allows an XMOBILE station to function as a bridge and still be restricted.

When a cell telephone is mapped to more than one XMOBILE station, then only one of the mapped XMOBILE station can have **origination** or **both** as its Mapping Mode. Therefore, only one of the XMOBILE stations mapped to the cell telephone number is permitted to originate calls.

Valid entries	Usage
both	The cell telephone can be used to originate and terminate calls from its associated XMOBILE extension. This is the default when the **XMOBILE Type** field is **PHS** or **DECT**.
none	The XMOBILE station is disabled administratively and cannot originate and terminate calls from its associated internal extension.
origination	The cell telephone can be used only to originate calls from its associated internal XMOBILE extension by dialing into the office server running Communication Manager.
termination	The cell telephone can be used only to terminate calls from its associated internal XMOBILE extension. This is the default when the **XMOBILE Type** field is **EC500**.

Map-to Station

This is the physical telephone used for calls to a virtual extension. Do not use an xmobile, xdid or another virtual extension.

Valid entries	Usage
Valid extension	Enter the extension of the physical telephone used for calls to a virtual extension.

Media Complex Ext

When used with Multi-media Call Handling, indicates which extension is assigned to the data module of the multimedia complex. Users can dial this extension to place either a voice or a data call, and voice conversion, coverage, and forwarding apply as if the call were made to the 1-number.

For an IP Telephone or an IP Softphone, this is the extension already administered as an H.323 station type. You can administer this field if the **IP Station** field on the System Parameters Customer-Options (Optional Features) screen is **y**.

Valid entries	Usage
A valid BRI data extension	For MMCH, enter the extension of the data module that is part of this multimedia complex.
H.323 station extension	For 4600 series IP Telephones, enter the corresponding H.323 station. For IP Softphone, enter the corresponding H.323 station. If you enter a value in this field, you can register this station for either a road-warrior or telecommuter/Avaya IP Agent application.
blank	Leave this field blank for single-connect IP applications.

Message Lamp Ext

Enter the extension of the station you want to track with the message waiting lamp.

Message Server Name

Specifies which Message Server is associated with the station.

Valid entries	Usage
Names administered in alphabetical order	Must contain a user-defined adjunct name that was previously administered on the **IP Node Names** screen.

Message Waiting Indicator

This field appears only for ISDN-BRI data modules and for 500, 2500, K2500, 7104A, 6210, 6218, 6220, 8110, H.323 and VRU telephones. (This field is independent of the administration of the Caller ID Message Waiting Indication for CallrID telephones.) Must be set to a value other than **none** when the **Type** field is set to **H.323**.

Valid entries	Usage
led	Enter **led** if the message waiting indicator is a light-emitting diode (LED).
neon	Enter **neon** if the message waiting indicator is a neon indicator. **Note:** The **neon** message waiting indicator is supported only on a small subset of boards, including older US-only boards, such as the TN746 and the TN793. When you select this option, the following warning appears: "WARNING: neon requires specific hardware/admin support." Check the documentation for your board to see if neon is supported. Note that this option is only available if **Analog Ringing Cadence** on the Location Parameters screen is set to **1** (US).
none	No message waiting indicator is selected. This is the default.

MIM Mtce/Mgt

Indicates if the telephone supports MIM Maintenance and Management capabilities other then endpoint initialization. Appears only if **MIM Support** is **y**.

MIM Support (Management Information Message Support)

This field appears only for ISDN-BRI data modules and ASAI. This field supports MIM endpoint initialization (SPID support) and other Maintenance or Management capabilities.

Valid entries	Usage
y	Enter **y** to display Endpt Init and MIM Mtce/Mgt.
n	Enter **n** for **ASAI**.

Mobility Trunk Group

This field associates the XMOBILE station to a trunk.

Valid entries	Usage
2000 (S87XX Server) **99** (DEFINITY CSI, S8300 Server) blank	Enter a valid trunk group number for mobility routing. This trunk group is used for routing. The **Configuration Set** field can be blank if the trunk group is **DECT** or **PHS**. If the trunk group is non-DECT or non-PHS, administer the **Configuration Set** field.
aar	The routing capabilities of Communication Manager is used to direct the call to an ISDN trunk. If no ISDN trunk is available, the call will not be extended out of the Avaya S8XXX Server. It provides ringback to the calling company and might eventually go to coverage.
ars	The routing capabilities of Communication Manager is used to direct the call to an ISDN trunk. If no ISDN trunk is available, the call will not be extended out of the Avaya S8XXX Server. It provides ringback to the calling company and might eventually go to coverage.

Model

This field appears only for NI-BRI telephones.

Valid entries	Usage
L-3 Communication STE	The NI-BRI telephone is a model L-3 Communication STE.
Tone Commander	The NI-BRI telephone is a model 6210 and 6220 Tone Commander.
Other	The NI-BRI telephone is another model (for example, a Nortel 5317T).

Mounting

Indicates whether the station mounting is **d**(esk) or **w**(all).

Note:
Fields in the SITE DATA area of the **Station** screen document information related to the station set installation.

Multimedia Early Answer

Allows you to establish multimedia early answer on a station-by-station basis.

Valid entries	Usage
y/n	If this station receives coverage calls for multimedia complexes, but is not multimedia-capable, enter **y** to ensure that calls are converted and talk path is established before ringing at this station.

Multimedia Mode

There are two multimedia modes, **Basic** and **Enhanced**. Toggling between Basic and Enhanced mode changes the administered mode of a station. The current Multimedia mode of a station is saved when a `save translation` command is executed.

Valid entries	Usage
Basic or **Enhanced**	In the **Basic** mode, users can place voice calls from a multifunction telephone and multimedia calls from a multimedia equipped computer. Voice calls can be answered at the multifunction telephone and multimedia calls alert first at the computer and if unanswered next alert at the voice station if it is administered with **H.320 Conversion** = **y**. In the **Enhanced** mode, voice and multimedia calls originate and are received at the telephone set. The call status is also displayed at the telephone set. An Enhanced mode station allows multimedia calls to take full advantage of most call control features.

Mute Button Enabled

Valid entries	Usage
y/n	Enter **y** to allow the user to use the **Mute** button on this telephone.

MWI Served User Type

Controls the auditing or interrogation of a served user's message waiting indicator (MWI).

Valid entries	Usage
fp-mwi	Use if the station is a served user of an fp-mwi message center.

Valid entries	Usage
qsig-mwi	Use if the station is a served user of a qsig-mwi message center.
blank	Leave blank if you do not want to audit the served user's MWI or if the user is not a served user of either an fp-mwi or qsig-mwi message center.

Name

Enter a name for the person associated with this telephone or data module. The system uses the **Name** field to create the integrated directory.

Note:
For 4610SW, 4620SW, 4621SW, and 4622SW, Sage, Spark, and 9600-series Spice telephones, the **Name** field has an associated optional native name field that is supported by the Unicode language display. The native name field is accessible through the Integrated Management Edit Tools such as Avaya Site Administration (ASA).

Avaya BRI stations support ASCII characters only. They do not support non-ASCII characters, such as Eurofont or Kanafont. Therefore, if you use non-ASCII characters in any Communication Manager **Name** field, such characters do not display correctly on a BRI station.

Off Premises Station

Analog telephones only.

Valid entries	Usage
y	Enter **y** if this telephone is not located in the same building with the system. If you enter **y**, you must complete R Balance Network.
n	Enter **n** if the telephone is located in the same building with the system.

PCOL/TEG Call Alerting

Appears only for 510 telephones.

Valid entries	Usage
y/n	Enter **y** to alert the station for Personal CO Line/Terminating Extension Group calls.

Per Button Ring Control

Valid entries	Usage
y	Enter **y** to allow users to select ring behavior individually for each **call-appr**, **brdg-appr**, or **abrdg-appr** on the station and to enable Automatic Abbreviated and Delayed ring transition for each **call-appr** on the station. Also, enter **y** if you do not want the system to automatically move the line selection to a silently alerting call unless that call was audibly ringing earlier.
n	Enter **n** if you want calls on **call-appr** buttons to always ring the station and calls on **brdg-appr** or **abrdg-appr** buttons to always ring or not ring based on the **Bridged Call Alerting** field value. Also, enter **n** if you want the system to move line selection to a silently alerting call if there is no call audibly ringing the station.

Personalized Ringing Pattern

Personalized Ringing allows users of some telephones to have one of 8 ringing patterns for incoming calls. Users working closely in the same area can each specify a different ringing pattern. This enables the users to distinguish their own ringing telephone from other telephones in the same area. For virtual stations, this field dictates the ringing pattern on its mapped-to physical telephone.

Enter a Personalized Ringing Pattern. (L = 530 Hz, M = 750 Hz, and H = 1060 Hz).

Valid entries	Usage
1	MMM (standard ringing)
2	HHH
3	LLL
4	LHH
5	HHL
6	HLL
7	HLH
8	LHL

Per Station CPN - Send Calling Number

Valid entries	Usage
y	All outgoing calls from the station delivers the Calling Party Number (CPN) information as “Presentation Allowed.”
n	No CPN information is sent for the call.
r	Outgoing non-DCS network calls from the station delivers the Calling Party Number information as “Presentation Restricted.”
blank	The sending of CPN information for calls is controlled by any administration on the outgoing trunk group the calls are carried on.

Port

Enter 7 characters to specify a port, or an x. If this extension is registered as an IP endpoint, this field displays ***sxxxxxx***, where ***xxxxxx*** is the number of previously registered IP stations. For example, if there are 312 IP sets already registered when you register, your extension would get port **s000313**.

Valid entries	Usage
01 to **64**	First and second numbers are the cabinet number
A to **E**	Third character is the carrier
01 to **20**	Fourth and fifth characters are the slot number
01 to **32**	Sixth and seventh characters are the circuit number
x	Indicates that there is no hardware associated with the port assignment since the switch was set up, and the administrator expects that the extension would have a non-IP set. Or, the extension had a non-IP set, and it dissociated. Use **x** for AWOH and CTI stations, as well as for SBS Extensions.
IP	Indicates that there is no hardware associated with the port assignment since the switch was set up, and the administrator expects that the extension would have an IP set. This is automatically entered for certain IP station set types, but can be entered for a DCP set with softphone permissions. This changes to the s00000 type when the set registers.

For DCP sets, the port can only be assigned once. ISDN-BRI provides a multipoint configuration capability that allows a previously assigned port to be specified more than once as follows: 2 stand-alone voice endpoints, 2 stand-alone data endpoints, or 1 integrated voice and data endpoint.

However, for the following cases, the port is assumed to be fully assigned:

- Maximum number of users (currently 2) are assigned on the port.
- One of the users on the port is a fixed TEI station.
- One of the users on the port has B-channel voice and B-channel data capability.
- One of the users on the port has no SPID assigned, which includes telephones that have no SPID initialization capability.
- This field is display-only for H.323 set types and 4600 set types. The system assigns an "IP" when the station is first administered.

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

Note:
To set up paging on an H.248 gateway, connect the paging system to a port on an MM711 and administer the port as an analog station on the Station screen. No entries on the Loudspeaker Paging screen are required.

Precedence Call Waiting

Valid entries	Usage
y/n	Enter **y** to activate Precedence Call Waiting for this station.

R Balance Network

Valid entries	Usage
y	Enter **y** to select the R Balance Capacitor network. In all other cases except for those listed under **n**, enter **y**.
n	Enter **n** to select the standard resistor capacitor network. You must complete this field if **Off-Premise Station** is **y**. Enter **n** when the station port circuit is connected to terminal equipment (such as SLC carriers or impedance compensators) optioned for 600-ohm input impedance and the distance to the equipment from the system is less than 3,000 feet.

Recall Rotary Digit

This field only appears if type is 500 or 2500.

Valid entries	Usage
y/n	Enter **y** to allow the user of a rotary telephone to dial the administered Recall Rotary Digit to receive recall dial tone. This enables the user to perform conference and transfer operations. You establish the Recall Rotary Digit on the **Feature-Related System Parameters** screen.

Redirect Notification

Valid entries	Usage
y/n	Enter **y** to give a half ring at this telephone when calls to this extension are redirected (via Call Forwarding or Call Coverage). Enter **y** if **LWC Reception** is **audix**.

Remote Office Phone

Valid entries	Usage
y/n	Enter **y** to use this station as an endpoint in a remote office configuration.

Remote Softphone Emergency Calls

Use this field to tell Communication Manager how to handle emergency calls from the IP telephone. This field appears when either the **IP Softphone** field or the **Remote Office Station** field is set to **y** on the Station screen.

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

Valid entries	Usage
as-on-local	Type **as-on-local** to achieve the following results: ● If the administrator chooses to leave the **Emergency Location Extension** fields (that correspond to this station's IP address) on the IP Address Mapping screen blank, the value **as-on-local** sends the extension entered in the **Emergency Location Extension** field in the Station screen to the Public Safety Answering Point (PSAP). ● If the administrator populates the IP Address Mapping screen with emergency numbers, the value **as-on-local** functions as follows: – If the **Emergency Location Extension** field in the Station screen is the same as the **Emergency Location Extension** field in the IP Address Mapping screen, the value **as-on-local** sends the extension to the Public Safety Answering Point (PSAP). – If the **Emergency Location Extension** field in the Station screen is different from the **Emergency Location Extension** field in the IP Address Mapping screen, the value **as-on-local** sends the extension in the IP Address Mapping screen to the Public Safety Answering Point (PSAP).
block	Enter **block** to prevent the completion of emergency calls. Use this entry for users who move around but always have a circuit-switched telephone nearby, and for users who are farther away from the Avaya S8XXX Server than an adjacent area code served by the same 911 Tandem office. When users attempt to dial an emergency call from an IP Telephone and the call is blocked, they can dial 911 from a nearby circuit-switched telephone instead.
	1 of 2

Valid entries	Usage
cesid	Enter **cesid** to allow Communication Manager to send the CESID information supplied by the IP Softphone to the PSAP. The end user enters the emergency information into the IP Softphone. Use this entry for IP Softphones with road warrior service that are near enough to the Avaya S8XXX Server that an emergency call routed over the it's trunk reaches the PSAP that covers the server or switch. If the server uses ISDN trunks for emergency calls, the digit string is the telephone number, provided that the number is a local direct-dial number with the local area code, at the physical location of the IP Softphone. If the server uses CAMA trunks for emergency calls, the end user enters a specific digit string for each IP Softphone location, based on advice from the local emergency response personnel.
option	Enter **option** to allow the user to select the option (extension, block, or cesid) that the user selected during registration and the IP Softphone reported. Use this entry for extensions that can be swapped back and forth between IP Softphones and a telephone with a fixed location. The user chooses between **block** and **cesid** on the softphone. A DCP or IP telephone in the office automatically selects **extension**.
	2 of 2

Restrict Last Appearance

Valid entries	Usage
y/n	Enter **y** to restrict the last idle call appearance for incoming priority calls and outgoing call originations only.

Rg

When **Per Button Ring Control** is **y**, this field appears next to the **call-appr** field in the **BUTTON ASSIGNMENTS** section of the Station screen.

Valid entries	Usage
a(bbreviated-ring) d(elayed-ring) n(o-ring) r(ing)	Enter the desired type of automatic abbreviated/delayed ringing for this call appearance. Default is **r**.

Room

Note:
Fields in the SITE DATA area of the **Station** screen document information related to the station set installation.

Valid entries	Usage
Up to 10 characters	To identify the telephone location.
Up to 5 characters	To identify the guest room number, if this station is one of several to be assigned a guest room and the **Display Room Information in Call Display** is **y** on the Hospitality-Related System Parameters screen.

SAC/CF Override

This feature allows the user of a station with a **Team** button administered, who is monitoring another station, to directly reach the monitored station by pushing the **Team** button. This overrides any currently active rerouting (for example, Send All Calls, Call Forwarding) on the monitored station.

Valid entries	Usage
Ask	The system asks if the user wants to follow the rerouting or override it. When the user has the option to decide whether rerouting should take place or not, a message is sent to the station which displays the active rerouting and the number of the forwarded to station. The user of the monitoring station can follow the rerouting by dialing “1,” or by letting the timer that supervises the **Team** button push expire. The user can override the active rerouting by dialing “0,” or by pushing the **Team** button once again.
No	Cannot override rerouting. The station does not have the ability to override the rerouting of a monitored station.
Yes	Can override rerouting. The station has the ability to override the rerouting the monitored station has set, as long as one incoming call appearance is free.

Secure Terminal Equip

This field is useful when Secure Terminal Equipment (STE) telephones are administered as 8510 telephones. This field appears on the BRI Station screen for 8503, 8510, and 8520 stations in Communication Manager 3.0 and later. See Bearer on page 801 for **Bearer** field functionality in Communication Manager 2.1 and 2.2.

Valid entries	Usage
n	Force the Bearer Cap IE to "speech" before a call is delivered to the 85xx BRI station.
y	Leave the Bearer Cap IE unchanged. Use 3.1khz to let secure calls from Secure Terminal Equipment (STE) telephones to work properly.

Security Code

Enter the security code required by users for specific system features and functions, including the following: Personal Station Access, Redirection of Calls Coverage Off-Net, Leave Word Calling, Extended Call Forwarding, Station Lock, Message Retrieval, Terminal Self-Administration, and Demand Printing. The required security code length is determined by **Minimum Security Code Length** on the Feature-Related System Parameters screen.

Select Last Used Appearance

Valid entries	Usage
y	Enter **y** to indicate a station’s line selection is not to be moved from the currently selected line button to a different, non-alerting line button. If you enter **y**, the line selection on an on-hook station only moves from the last used line button to a line button with an audibly alerting call. If there are no alerting calls, the line selection remains on the button last used for a call.
n	Enter **n** so the line selection on an on-hook station with no alerting calls can be moved to a different line button, which might be serving a different extension.

Service Link Mode

The service link is the combined hardware and software multimedia connection between an Enhanced mode complex's H.320 DVC system and a server running Communication Manager which terminates the H.320 protocol. When the user receives or makes a call during a multimedia or IP Softphone or IP Telephone session, a "service link" is established.

Valid entries	Usage
as-needed	Use this setting for most multimedia, IP Softphone, or IP Telephone users. Setting the Service Link Mode to as-needed leaves the service link connected for 10 seconds after the user ends a call so that they can immediately place or take another call. After 10 seconds the link is dropped and a new link would have to be established to place or take another call.
permanent	Use **permanent** for busy call center agents and other users who are constantly placing or receiving multimedia, IP Softphone, or IP Telephone calls. In permanent mode, the service link stays up for the duration of the multimedia, IP Softphone, or IP Telephone application session.

Set Color

Indicates the set color. Valid entries include the following colors: beige, black, blue, brown, burg (burgundy), gray, green, ivory, orng (orange), red, teak, wal (walnut), white, and yel (yellow).

Note:
Fields in the SITE DATA area of the **Station** screen document information related to the station set installation.

Speaker

Indicates whether the station is equipped with a speaker.

Note:
Fields in the SITE DATA area of the **Station** screen document information related to the station set installation.

Speakerphone

This field controls the behavior of speakerphones.

Valid entries	Usage
1-way	Enter **1-way** to indicate that you want the speakerphone to be listen-only.
2-way	Enter **2-way** to indicate that you want the speakerphone to be both talk and listen.
grp-listen	Group Listen works with only 6400-series and 2420/2410 telephones. Group Listen allows a telephone user to talk and listen to another party with the handset or headset while the telephone's 2-way speakerphone is in the listen-only mode. Others in the room can listen, but cannot speak to the other party via the speakerphone. The person talking on the handset acts as the spokesperson for the group. Group Listen provides reduced background noise and improves clarity during a conference call when a group needs to discuss what is being communicated to another party.
none	

Special Dialing Option

This field identifies the type of dialing for calls when this data module originates calls.

Valid entries	Usage
hot-line	Causes the **HOT LINE DESTINATION — Abbreviated Dialing Dial Code** field to appear, for administering a hot line destination. The Hot Line Service destination number is stored in an Abbreviated Dialing List. When the user goes off-hook on a Data Hot Line call, the system dials the AD number.
default	Causes the **Default Dialing Abbreviated Dialing Dial Code** field to appear, for entering a list number associated with the abbreviated dialing list. When the user goes off-hook for a data call and presses the **Enter** button following the DIAL prompt, the system dials the AD number.
blank	For regular (normal) keyboard dialing.

SPID — (Service Profile Identifier)

Enter a variable length parameter. This field appears only if the **Endpt Init** field is **y**.

The SPID is a numeric string, which means that the value of 00 is different from 000. The SPID must be different for all terminals on the BRI and from the Service SPID. The SPID should always be assigned. If the SPID is not assigned for the first BRI on a port, any other BRI assignment to that port are blocked.

Note:
If you have set the **Port** field to **X** for an ISDN-BRI extension and intend to use Terminal Translation Initialization (TTI) to assign the port, then the SPID number must equal the station number.

Survivable COR

This field is for all analog and IP station types. Use this field to set a level of restriction for stations to be used in conjunction with the survivable dial plan in order to limit certain users to only to certain types of calls. The restriction levels are listed in order from most restrictive to least restrictive. Each level assumes the calling ability of the ones above it.

Valid entries	Usage
emergency	This station can only be used to place emergency calls which are defined.
internal	This station can only make intra-switch calls. This is the default.
local	This station can only make calls that are defined as **locl**, **op**, **svc**, or **hnpa** in the Survivable ARS Analysis Table
toll	This station can place any national toll calls which are defined as **fnpa** or **natl** on the Survivable ARS Analysis Table.
unrestricted	This station can place a call to any number defined in the Survivable ARS Analysis Table. Those strings marked as **deny** are also denied to these users as well.

Survivable GK Node Name

Appears when the **Type** field is any H.323 IP phone type, such as **16xx**, **46xx**, or **96xx**. Use this field to indicate the gatekeeper to register with when the gateway unregisters (loses call control) with the main server. The media gateway delivers the gatekeeper list to IP endpoints, allowing them to register and subsequently originate/receive calls from other endpoints in this survivable calling zone.

Valid entries	Usage
valid IP node name or blank	Enter any valid IP node name as administered on the IP Node Names screen to allow the station to be registered as an IP telephone, associated to an H.323 gateway that is capable of supporting a gatekeeper in survivable mode. Default is blank.

Survivable Trunk Dest

This field is for all analog and IP station types. Use this field to designate certain telephones as not being allowed to receive incoming trunk calls when the Media Gateway is in survivable mode.

Valid entries	Usage
y	Enter **y** to allow stations to be incoming trunk destinations while the Media Gateway is running in survivability mode. This is the default.
n	Enter **n** to prevent this station from receiving incoming trunk calls when in survivable mode.

Switchhook Flash

Must be set to **y** when the **Type** field is set to **H.323**.

Valid entries	Usage
y	Enter **y** to allow users to use the switchhook flash function to activate Conference/Transfer/Hold and Call Waiting.
n	Enter **n** to disable the flash function so that when the switchhook is pressed while active on a call, the call drops. If this field is **n**, you must set **Call Waiting Indication** to **n**.

TEI

Appears only when the **Fixed TEI** field is **y**.

Valid entries	Usage
0 to **63**	1 or 2-digit number

Tests

Valid entries	Usage
y	Enter **y** to enable port maintenance tests.
n	If the equipment (dictaphone) connected to the port does not support these tests, you must enter **n**.

Time of Day Lock Table

Use this field to assign the station to a TOD Lock/Unlock table.

Valid entries	Usage
1 to **5**, or blank	The default is blank, indicating no TOD Lock/Unlock feature is active. The assigned table must have administered a valid time interval entry, and the **Table Active** field on the **Time of Day Lock Table** screen must be set to **y**.

TN

Enter the Tenant Partition number. Also, SBS Extensions can be partitioned.

Type

For each station that you want to add to your system, you must specify the type of telephone in the **Type** field. This is how you distinguish between the many different types of telephones.

The following table lists the telephones, virtual telephones, and personal computers that you can administer on Communication Manager. To administer telephones that are not in the table, use the Alias Station screen.

Note:
You cannot change an analog telephone administered with hardware to a virtual extension if **TTI** is **y** on the Feature-Related System Parameters screen. Contact your Avaya representative for more information.

To set up paging on an H.248 gateway, connect the paging system to a port on an MM711 and administer the port as an analog station on the Station screen. No entries on the Loudspeaker Paging screen are required.

Table 6: Telephones

Telephone type	Model	Administer as
Single-line analog	500	500
	2500, 2500 w/ Message Waiting Adjunct	2500
	6210	6210
	6211	6210
	6218	6218
		1 of 8

Table 6: Telephones (continued)

Telephone type	Model	Administer as
	6219	6218
	6220	6220
	6221	6220
CallerID	Analog telephone w/Caller ID	CallrID
	7101A, 7102A	7101A
	7103A Programmable and Original	7103A
	7104A	7104A
	8110	8110
	DS1FD	DS1FD
	7302H, 7303H	7303S
	VRU (voice response unit) with C&D tones	VRU
	VRU without C&D tones	2500
Single-line DS1/DSO (Lineside T1/DS1)	DS1 device without forward disconnect	ops
	VRU with forward disconnect without C&D tones	ds1fd or ds1sa
	VRU with forward disconnect without C&D tones	VRUFD or VRUSA
Terminals	510D	510
	515BCT	515
Multiappearance hybrid	7303S	7303S, 7313H
	7305H	7305S
	7305S	7305S, 7316H, 7317H
	7309H	7309H, 7313H
	7313H	7313H
	7314H	7314H
	7315H	7315H
	7316H	7316H

2 of 8

Table 6: Telephones (continued)

Telephone type	Model	Administer as
	7317H	7317H
Multiappearance digital	2402	2402
	2410	2410
	2420	2420
	6402	6402
	6402D	6402D
	6408	6408
	6408+	6408+
	6408D	6408D
	6408D+	6408D+
	6416D+	6416D+
	6424D+	6424D+
	7401D	7401D
	7401+	7401+
	7403D	7403D
Multiappearance digital	7404D	7404D
	7405D	7405D
	7406D	7406D
	7406+	7406+
	7407D	7407D
	7407+	7407+
	7410D	7410D
	7410+	7410+
	7434D	7434D
	7444D	7444D
	8403B	8403B
	8405B	8405B
		3 of 8

Table 6: Telephones (continued)

Telephone type	Model	Administer as
	8405B+	8405B+
	8405D	8405D
	8405D+	8405D+
	8410B	8410B
	8410D	8410D
	8411B	8411B
	8411D	8411D
	8434D	8434D
	CALLMASTER I	602A1
	CALLMASTER II, III, IV	603A1, 603D1, 603E1, 603F1
	CALLMASTER VI	606A1
	IDT1	7403D
	IDT2	7406D
IP Telephone	4601+ **Note:** When adding a new 4601 IP telephone, you must use the 4601+ station type. This station type enables the Automatic Callback feature. When making a change to an existing 4601, you receive a warning message, stating that you should upgrade to the 4601+ station type in order to access the Automatic Callback feature.	4601+
	4602+ **Note:** When adding a new 4602 IP telephone, you must use the 4602+ station type. This station type enables the Automatic Callback feature. When making a change to an existing 4602, you receive a warning message, stating that you should upgrade to the 4602+ station type in order to access the Automatic Callback feature.	4602+
	4606	4606
	4610	4610
		4 of 8

Table 6: Telephones (continued)

Telephone type	Model	Administer as
	4612	4612
	4620SW IP (G3.5 hardware) **Note:** Effective December 5, 2005, Avaya is no longer make 4620 IP telephones commercially available. The 4621SW IP telephone is an appropriate replacement. The 4621SW IP telephone, generally available since May 2005, offers the same functionality as the 4620 and adds a backlit display.	4620
	4621	4621
	4622	4622
	4624	4624
	4625	4625
	4690	4690
IP Telephone	9610	9610 **Note:** If your version of Communication Manager is earlier than version 4.0, administer as 4606.
	9620	9620 **Note:** If your version of Communication Manager is earlier than version 4.0, administer as 4610.
	9630	9630 **Note:** If your version of Communication Manager is earlier than version 4.0, administer as 4620.
		5 of 8

Table 6: Telephones (continued)

Telephone type	Model	Administer as
	9640	9640 **Note:** If your version of Communication Manager is earlier than version 4.0, administer as 4620.
	9650	9650 **Note:** If your version of Communication Manager is earlier than version 4.0, administer as 4620.
SIP IP Telephone	• 4602SIP with SIP firmware • 4610SIP with SIP firmware • 4620SIP with SIP firmware • 4620SIPCC (Call Center) • Avaya one-X (tm) Deskphone 9620, 9630, 9630G 9640, 9640G with SIP firmware • SIP Softphone/Avaya one-X Desktop • 1616CC (Call Center) • Toshiba SP-1020A **Note**: Any model telephone that has SIP firmware and is being used for SIP networking must be administered as a 4620SIP telephone.	4620SIP
IP SoftPhone	Road-warrior application	H.323 or DCP type
	Native H.323	H.323
	Single-connect	H.323 or DCP type
ISDN-BRI station	—	asai
	Any NI-BRI (N1 and N2) telephone	NI-BRI
		6 of 8

Table 6: Telephones (continued)

Telephone type	Model	Administer as
	7505D	7505D
	7506D	7506D
	7507D	7507D
	8503D	8503D
	8510T	8510T
	8520T	8520T
Personal computer	6300/7300	PC
(voice/data)	6538/9	Constellation
Test Line	ATMS	105TL
No hardware assigned at the time of administration.		XDID (use when Communication Manager later assigns a DID number to this station) XDIDVIP (use when the administrator later assigns a DID number to this station) virtual (use to map this and other extensions to one physical telephone)
Key telephone system interface	—	K2500
ASAI	asai link computer telephony adjunct link	asai adjlk
AWOH	any digital set CTI station	same as "Multiappearance Digital" see table above CTI
CTI	CTI station	CTI
		7 of 8

Table 6: Telephones (continued)

Telephone type	Model	Administer as
XMOBILE	EC500, DECT, PHS	XMOBILE
ISDN-BRI data module	7500	7500
SBS Extension	SBS test extension (no hardware)	sbs
		8 of 8

Type of 3PCC Enabled

Valid entries	Usage
Avaya or **none**	Default is none.

Voice mail Number

This field supports the voice mail retrieval feature as a fixed feature button on type 24xx, 46xx, and 96xx telephones. When you enter a number in this field, the telephone's fixed voice mail retrieval button acts an autodial button, dialing the number entered in this field to access voice mail.

Note:
If this field is left blank, the telephone's fixed voice mail retrieval button acts as a "transfer to voice mail" button, which only works for INTUITY Audix or QSIG-integrated voice mail systems. Avaya recommends that for INTUITY Audix and QSIG-integrated voice mail systems, this field should be left blank. For DEFINITY AUDIX and non-QSIG integrated voice mail systems, this field should be filled in with the appropriate number.

Valid entries	Usage
digits (1 to 9) * # ~p (pause) ~w/~ (wait) ~m (mark) ~s (suppress)	Enter the complete voice mail number up to 15 digits.

XID

Appears only for an ISDN-BRI data module or an ASAI link. Used to identify Layer 2 XID testing capability.

XMOBILE Type

When the **Type** field is **XMOBILE**, the **Mobility Trunk Group** field must be administered.

Valid entries	Usage
DECT	For the DECT Access System or the AGCS (ROAMEO) IS-136 (TDMA cellular).
EC500	For any public cellular networks.
PHS	For the DENSO 300M.

XOIP Endpoint type

This field appears when a valid analog station is entered in the **Type** field. Use this field for older or non-standard external equipment such as modems, fax, and TTY devices, which are not easily recognized by VoIP resources within Communication Manager. By identifying this external equipment through administration, VoIP firmware can determine whether to immediately attempt to put a call in pass-through mode, or allow the system to handle it normally. Supported station types are 500, 2500, K2500 and CallrID.

Valid entries	Usage
auto **modem** **fax** **tty**	Enter the type of analog endpoint for this station. Default is **auto**. **Note**: This field is intended for exception cases only. For the majority of stations, use the default setting of auto.

Stations With Off-PBX Telephone Integration

Use the Stations with Off-PBX Telephone Integration screen to map an office phone to a cell phone through the Extension to Cellular feature. The office phone can be a standard office number or an administration without hardware (AWOH) station.

For more information on Extension to Cellular, see *Avaya Aura™ Communication Manager Feature Description and Implementation*, 555-245-205.

Field descriptions for page 1

Figure 290: Stations with Off-PBX Telephone Integration screen, page 1

```
add off-pbx-telephone station-mapping                          Page   1 of   x
                    STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

 Station        Application Dial   CC  Phone Number    Trunk       Config  Dual
 Extension                  Prefix                     Selection   Set     Mode
                                 -
                                 -
                                 -
                                 -
                                 -
                                 -
                                 -
                                 -
                                 -
                                 -
                                 -
                                 -
                                 -
                                 -
                                 -
                                 -
```

Command parameters

Action	Object	Qualifier
add	**off-pbx-telephone station-mapping**	
change	**off-pbx-telephone station-mapping**	***<station extension>***
display	**off-pbx-telephone station-mapping**	***<station extension>***
list	**off-pbx-telephone station-mapping**	***<variable>***

The **add off-pbx-telephone station-mapping** command displays the blank **Stations with Off-PBX Integration** screens. You can add up to sixteen associations between an office phone and an external phone.

The **`change off-pbx-telephone station-mapping`** ***`<station extension>`*** command displays the **Stations with Off-PBX Integration** screens. You can change the associations between office telephones and external telephones. The first line on the screen contains the information for the station extension that you entered as the command variable. You can also add additional associations in this screen.

The **`display off-pbx-telephone station-mapping`** ***`<station extension>`*** command displays the **Stations with Off-PBX Integration** screens. The ***`<station extension> variable is not mandatory`*** These screens list up to sixteen entries, starting with the station extension you entered as the command variable. If this extension is not administered for an off-PBX, the display starts with the next administered off-PBX extension in numerical order.

The **`list off-pbx-telephone station-mapping`** ***`<variable>`*** command information about the association between an office phone and an off-PBX phone. The command variable specifies the office phone number or numbers of interest. The ***`<variable>`*** can be:

- a complete phone number
- a partial phone number followed by an asterisk, which is a “wildcard” character
- blank

Station Extension

The **Station Extension** field is an administered extension in your dial plan. This number is the extension of the office phone.

Valid entries	Usage
a valid number in your dial plan	Type an extension number of the office phone. Default is blank.

Application

Indicate the type of off-PBX application that is associated with the office phone. You can assign more than one application to an office phone.

Valid entries	Usage
blank	Default is blank.
CSP	cell phone with Extension to Cellular provided by the cellular service provider
EC500	cell phone with Extension to Cellular
HEMU	Home Enterprise Mobility User

Valid entries	Usage
OPS	SIP Enablement Services (SES)-enabled phone
PBFMC	Public Fixed Mobile Convergence
PVFMC	Private Fixed Mobile Convergence
SCCAN	wireless SIP Enablement Services (SES) phone and cell phone
VEMU	Visited Enterprise Mobility User
VIEMU	Visited Initial Enterprise Mobility User

CC

Valid entries	Usage
up to three digits using **0**–**9**, or blank	Enter the Country Code associated with the extension. Multiple entries that use the same phone number must also have the same Country Code, including entries on XMOBILE Station screens. Country Code changes made to existing stations or XMOBILE entries are applied to all instances of the phone number. SAFE (Self-Administered Feature Access Code for EC500) is not recommended on an extension that has an administered Country Code. Origination mapping can occur with or without a country code. Default is blank.

Dial Prefix

The system prepends the **Dial Prefix** to the off-PBX phone number before dialing the off-PBX phone. The system deletes the dial prefix when a user enters their cell phone number using the Self Administration Feature (SAFE) access code. You must set the routing tables properly so that the dial prefix "1" is not necessary for correct routing.

Valid entries	Usage
blank **0**–**9**, *****, **#**	Type up to four digits, including “*****” or “**#**”. If included, “*****” or “**#**” must be in the first digit position. Enter a "1" if the phone number is long-distance. Enter "011" if the phone number is international. Default is blank.

Phone Number

Enter the phone number of the off-PBX phone.

Valid entries	Usage
0–9	Enter the phone number of the off-PBX phone. May be blank for the first EC500, CSP or PBFMC phones administered. May be blank for EC500, CSP, PBFMC, which support SAFE (Self-Administered Feature Access Code for EC500). Default is blank.

Trunk Selection

Valid entries	Usage
ars **aar** **trunk group number** **ext**	Indicates the trunk group selection method used for the outgoing call to Session Manager only if the application field is set to ops. If the application field is set to any other value, for example EC500, then the trunk selected will not be Session Manager. It will be some other destination, for example, to a cell phone in the PSTN. In the half-call model, indicates a different trunk group than the one that eventually connects the call to the telephone network. For the one-X® application type: ars: typically used for off-PBX telephones. ext: typically used for on-PBX telephones.

Configuration Set

Use the **Configuration Set** field to administer the Configuration Set number. This number contains the desired call treatment options for the Extension to Cellular station. Ninety-nine Configuration Sets exist.

The SCCAN application requires two different configuration sets selected for each station. The first set is the value for the WLAN followed by a slash. The second is the value for the cellular network.

Valid entries	Usage
1–99	Type the number of the Configuration Set. Shows blank for Enterprise Mobility User (EMU). Default is blank.

Field descriptions for page 2

Finish the administration steps to map an office phone to an off-PBX phone on the second page of the Stations with Off-PBX Telephone Integration screen (Figure 291: Stations with Off-PBX Telephone Integration screen, page 2 on page 853). The information you entered in the first page appears as display-only information on the second page.

Figure 291: Stations with Off-PBX Telephone Integration screen, page 2

```
add off-pbx-telephone station-mapping                            Page   2 of   x
                  STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

 Station       Appl    Call         Mapping    Calls        Bridged       Location
 Extension     Name    Limit        Mode       Allowed      Calls
```

Call Limit

Valid entries	Usage
1–10	Set the maximum number of Extension to Cellular (EC500) calls that can be active simultaneously at a single station. Default is **2** for EC500, CSP, OPS, PBFMC, PVFMC.

Mapping Mode

Enter the mode of operation for the Extension to Cellular cell phone. Use these modes to control the degree of integration between the cell phone and the office phone. The modes are valid for Extension to Cellular calls only. For each office phone, you can only assign one cell phone as the **origination** mode. You cannot assign a cell phone as either the **origination** or **both** mode more than once.

Valid entries	Usage
both	Default = **both** when the **Phone Number** field was previously administered for another extension with a **Mapping Mode** of **termination or none**. Default = **termination** when the **Phone Number** field was previously administered with a **Mapping Mode** of **origination or both**. In the both mode, users can originate and receive calls from the office phone with the cell phone.
termination	In **termination** mode, users can only use their Extension to Cellular cell phone to receive calls from the associated office phone. Users cannot use the cell phone to originate calls from the associated office phone. Calls originating from the cell phone independent of the office phone are independent of Extension to Cellular and behave exactly as before enabling Extension to Cellular.
origination	In **origination** mode, users can only originate Extension to Cellular cell phone calls from the associated office phone. Users cannot use the cell phone to receive calls from the associated office phone.
none	In the **none** mode, users cannot originate or receive calls from the office phone with the cell phone.

Calls Allowed

Identify the call filter type for an Extension to Cellular station. The **Calls Allowed** values filter the type of calls to the office phone that a user can receive on an Extension to Cellular cell phone.

Valid entries	Usage
all	Default is **all**. The cell phone receives both internal and external calls.
internal	The cell phone receives only internal calls.
external	The cell phone receives only external calls.
none	The cell phone does not receive any calls made to the associated office phone.

Bridged Calls

Use the **Bridged Calls** field to determine if bridged call appearances extend to the Extension to Cellular cell phone. The valid entry definitions are the same as the **Mapping Mode** field entries.

Valid entries	Usage
both **termination** **origination** **none**	Default for OPS and PVFMC applications is **none**. Default for other applications is **both**.

Location

Valid entries	Usage
blank **1**–**255**	Set the **Location** value for each OPS, PBFMC, or PVFMC application that you have administered on Page 1. You can find the **Location** value on the Locations screen (`change locations`). ● For a DCP set, enter the location of the media gateway. ● For an IP set, enter the location of the network region for the set. ● If blank, use the location of the incoming trunk. The default for this field is blank.

Survivable ARS Analysis Table

Communication Manager compares dialed numbers with the dialed strings in this table and determines the route pattern for the number.

Field descriptions for page 1

Figure 292: Survivable ARA Analysis screen

```
change survivable-ars-analysis                                Page 1 of X

                             SURVIVABLE ARS ANALYSIS TABLE

                              Total        Call                 Trunk
Dialed String                 Length       Type       Deny      Grp No
__________________________      __         ____       _____     _____
__________________________      __         ____       _____     _____
__________________________      __         ____       _____     _____
__________________________      __         ____       _____     _____
__________________________      __         ____       _____     _____
__________________________      __         ____       _____     _____
__________________________      __         ____       _____     _____
__________________________      __         ____       _____     _____
__________________________      __         ____       _____     _____
__________________________      __         ____       _____     _____
__________________________      __         ____       _____     _____
__________________________      __         ____       _____     _____
__________________________      __         ____       _____     _____
__________________________      __         ____       _____     _____
__________________________      __         ____       _____     _____
__________________________      __         ____       _____     _____
```

Call Type

Enter the call type to be used for this dialed string. This field cannot be left blank if the **Dialed String** field is not blank.

Valid entries	Usage
emer **fnpa** **hnpa** **intl** **iop** **locl** **natl** **op** **svc**	The different call types and its usage are in the following: ● **emer**—only emergency call ● **fnpa**—(Foreign Number Plan Area) 10-digit North American Numbering Plan (NANP) call (11 digits with Prefix Digit “1”) ● **hnpa**—(Home Number Plan Area) 7-digit NANP call ● **intl**—public-network international number call ● **iop**—international operator assisted call ● **locl**—public-network local number call ● **natl**—non-NANP call ● **op**—operator assisted call ● **svc**—service call The default value is **emer**.

Deny

The system denies a dialed string that does not match an entered pattern. If there are no available trunks when a match is performed for the given route option, then the user receives a trunk busy indication.

Valid entries	Usage
y/n	Indicate whether or not the dialed string should be blocked.

Dialed String

Valid entries	Usage
up to 18 characters, **0-9**, *****, **x**, or **X**	Enter the dialed string digits to be analyzed.

Total Length

This field defines the minimum number of digits required to validate the route. The minimum value when the dial string is populated is the length of the dialed string entry with a maximum value up to **28**. This field cannot be left blank if the **Dialed String** field is not blank.

Valid entries	Usage
0 to **28**	Enter the minimum number of digits required to validate the route. Default is blank.

Trunk Grp No

Valid entries	Usage
Valid trunk group number	Enter the trunk group number that specifies the destination route for the dial plan analysis of this dialed string.

Survivable Processor

Use the Survivable Processor screen to administer a Local Survivable Processor (LSP) and ESS server, or to control use of the Processor Ethernet interface on the LSP or ESS. Before administering this screen, you must first assign node names for each LSP and ESS server on the **IP Node Names** screen.

While this screen is administered on the active main server, the information entered applies only to LSPs and ESSs and does not apply to main servers. When translations are copied to an LSP or ESS, the LSP/ESS replaces like translations for the main server with the overrides administered on the Survivable Processor screens. That is, use the Survivable Processor screen to administer overrides against adjunct links that have already been administered for the main server(s). For more information on Processor Ethernet, see *Administering Avaya Aura™ Communication Manager*, 03-300509. For more information about ESS, see *Using Avaya Enterprise Survivable Servers (ESS)*, 03-300428.

Note:
The ESS server is first administered on the **IP Node Names** screen and then on the **System Parameters – Port Networks** screen before it is administered on the Survivable Processor screen. The information from the **System Parameters – Port Networks** screen is automatically copied to the Survivable Processor screen. You do not need to use the Survivable Processor screen for an ESS server unless one of the supported adjuncts connects to the PE interface of the ESS server.

Field descriptions for page 1

The first page of the Survivable Processor screen displays the Processor Ethernet interface information for the LSP or the ESS server. The information includes the node name, the IP address, the server type, the cluster ID, and the network region. The only administrable field on this page is the **Network Region** field. The following figures show several ways that page 1 of the Survivable Processor screen might appear. The descriptions that follow the figures include all fields shown in all variations of Page 1.

Field descriptions for page 1

Figure 293: Survivable Processor screen - adding an LSP

```
add survivable-processor punsrvrl                              Page   1 of   x
                    SURVIVABLE PROCESSOR

Type: lsp                                 Processor Ethernet Network Region:


          Node Name: lsp-4
         IP Address: 135.9.9.4
```

Figure 294: Survivable Processor screen - adding a simple ESS server

```
add survivable-processor punsrvrs                                  Page   1 of   x
                    SURVIVABLE PROCESSOR

Type: simplex-ess      Cluster ID:       Processor Ethernet Network Region: 1
                        Community: 1         Enable PE for H.323 Endpoints? n
                                              Enable PE for H.248 Gateways? n


SERVER A
          Server ID:
          Node Name: ess-smp
         IP Address: 135.9.9.4






PORT NETWORK PARAMETERS
                  Community Size: all      System Preferred: y
                  Priority Score: 1         Local Preferred: n
                                                 Local Only: n
```

Figure 295: Survivable Processor screen - adding a duplicated ESS server

```
add survivable-processor punsrvrd                                   Page   1 of   x
                    SURVIVABLE PROCESSOR

Type: duplex-ess       Cluster ID:       Processor Ethernet Network Region: 1
                        Community: 1         Enable PE for H.323 Endpoints? n
ACTIVE SERVER                                 Enable PE for H.248 Gateways? n
          Node Name: ess-d
         IP Address: 135.9.9.4
SERVER A
          Server ID:
          Node Name:
         IP Address:
SERVER B
          Server ID:
          Node Name:
         IP Address:

PORT NETWORK PARAMETERS
                  Community Size: all      System Preferred: y
                  Priority Score: 1         Local Preferred: n
                                                 Local Only: n
```

Cluster ID

Valid entries	Usage
1 to **999**, blank	Enter the Cluster ID (the Module ID from the Communication Manager license file) for the ESS server. The Cluster ID corresponds to the Module ID from the license file of the ESS server. **Note:** If you do not know the Module ID for the ESS server, use the `statuslicense -v` shell command on the ESS server. The Module ID displays in the **RFA Module ID** field.

Community

Valid entries	Usage
	A community is a virtual group consisting of an ESS server and one or more Port Networks. Assigning an ESS server to a community associates the ESS server with the IPSI(s) in the Port Network(s) for that community. The IPSI(s) are assigned to communities on the System Parameters Port Network screen. The association effects how the ESS server is prioritized for the IPSI in that community, if the ESS server is administered with a **Local Preferred** or **Local Only** preference. The Community number for an S8400 ESS server must be set to 2 or greater and must be unique. **Note:** It is possible to administer an ESS server as having no preferences and just a priority score. If all ESS servers were administered in this fashion, the IPSI would prioritize each ESS server based on its priority score only.

Enable PE for H.323 Endpoints?

Valid entries	Usage
y/**n**	Enter **y** to allow the PE interface of the ESS server to be used for H.323 devices such as phones. If you enter **n**, the ESS Node Name may not appear in the Alternate Gatekeeper (Backup Server) List on the IP Network Regions screen. If you enter **y** and you administer the ESS node name on the IP Network Regions screen, the AGL list for IP endpoints will include the ESS PE. When you run the **`display ip-interface procr`** command on the ESS server, the **Allow H.323 Endpoints?** field in that screen displays the value that you enter.

Enable PE for H.248 Gateways?

Valid entries	Usage
y/**n**	Enter **y** to allow the PE interface of the ESS server to be used for gateways. When you run the **`display ip-interface procr`** command on the ESS server, the **Allow H.248 Gateways?** field in that screen displays the value that you enter.

IP Address

e

Valid entries	Usage
	(display-only) shows the IP address that corresponds to the node name you entered. There are three IP addresses, one for each node name if the survivable processor is a duplicated ESS.

Node Name

Valid entries	Usage
Character string (up to 15 characters max.)	(display-only) shows the name used to identify this server. You enter node names through the IP Node Names screen. If the survivable processor is duplicated, there are three node names, one each for the duplicated server pair and one for the server that is active at a given point of time. The IP address of the active server is known as the IP-Alias address.

Processor Ethernet Network Region

Valid entries	Usage
1 to **250**	Enter the network region in which the PE interface of the LSP or ESS resides.

Type

Valid entries	Usage
lsp, simplex_ess, duplex_ess	Enter the survivable processor type.

Active Server

Node Name

Valid entries	Usage
valid node name administered on the IP Node Names screen	(display only) This field is displayed only for duplicated ESS servers. The node name entered at the command line is displayed.

IP Address

Valid entries	Usage
	(display only) This field is displayed only for duplicated ESS servers. The IP address corresponding to the node name entered at the command line is displayed.

Server A

Server ID

Valid entries	Usage
1 to **256**, blank	Server A ID corresponds to the Server ID configured using the **Set Identities** page under **Configure Server** on the SMI GUI of the ESS server. The administration on the main server and the configuration on the ESS server must match for the ESS server to register to the main server.

Node Name

Valid entries	Usage
valid node name administered on the IP Node Names screen	(display only) For LSP or single ESS server, the node name is displayed. For duplicated ESS servers, enter the node name for Server A.

IP Address

Valid entries	Usage
	(display only) The IP address corresponding to the node name for Server A is displayed.

Server B

Server ID

Valid entries	Usage
1 to **256**, blank	(display only) For duplicated ESS servers, the node name of Server B is displayed.

Node Name

Valid entries	Usage
valid node name administered on the IP Node Names screen	For duplicated ESS servers, enter the node name for Server B.

IP Address

Valid entries	Usage
	(display only) For duplicated ESS servers, the IP address corresponding to the node name for Server B is displayed.

PORT NETWORK PARAMETERS

Community Size

Valid entries	Usage
all, **Sngl_PN**	Default is **all**. For an S8400 ESS server, the value must be **Sngl_PN**.

Local Only

Valid entries	Usage
	Use this option when you want the ESS server to accept the request for service from an IPSI, only if the IPSIs is located in the same community as the ESS server. Default is **n**. For S8400 ESS server, this defaults to **y** and cannot be changed.

Local Preferred

Valid entries	Usage
	Use this option when you want the ESS server to accept the request for service from IPSIs co-located in the same geographical region, WAN/LAN segment, district, or business unit. Default is **n**. When the Community size is set to **Sngl_PN** (for an S8400 ESS server) this field defaults to **n** and cannot be changed.

Priority Score

Valid entries	Usage
1 to **100**	Enter the Priority Score for this ESS server. Default is **1**.

System Preferred

Valid entries	Usage
y/n	Use this option when the goal is to keep as much of the system network intact as possible, allowing one ESS server to replace the Main server. If this field is set to **y**, then **Local Preferred** and **Local Only** default to **n** and cannot be changed. If this field is **n**, then **Local Preferred** and **Local Only** can be either **y** or **n**. Default is **y**. For S8400 ESS server, this defaults to **n** and cannot be changed.

Survivable Processor - Processor Channels page

Use this page of the Survivable Processor screen to administer processor channels.

Figure 296: Survivable Processor - Processor Channels screen

```
                                                              page 2 of x

                        SURVIVABLE PROCESSOR - PROCESSOR CHANNELS

Proc                          Interface           Destination      Session
Chan   Enable   Appl.   Mode  Link/Chan             Node         Port   Local/Remote

 1       i      mis      s     9   5001         cmshost           0        1      1
 1       i      ccr      s     10  5002         ccrhost1          0        1      1
```

Appl

This display-only field identifies the server application type or adjunct connection used on this channel.

Valid entries	Usage
mis	CMS channel assignments.
ccr	CCR channel assignments.

Destination Node

This field identifies the server or adjunct at the far end of this link. This field changes to display-only when you enter **i(nherit)** in the **Enable** field, showing the value administered on the main server.

Valid entries	Usage
valid administered node name	Enter an adjunct name, server name, far end IP address, node name, or leave blank for services local to this Avaya S8XXX Server.

Destination Port

This field specifies the far-end port number of this link. This field changes to display-only when you enter **i(nherit)** in the **Enable** field, showing the value administered on the main server.

Valid entries	Usage
0, **5000** to **64500**	Enter the number of the destination port. An entry of **0** means any port can be used.

Enable

Use this field to specify how data for this processor channel is transferred to the survivable processor.

Valid entries	Usage
i(nherit)	Enter **i(nherit)** to indicate that this link is to be inherited by the LSP or ESS server. When you enter **i(nherit)**, all fields in the row for this processor channel change to display-only. The survivable processor inherits this processor channel just as it is administered on the main server. You generally use the **i(nherit)** option in the following situations: ● The main server connects to the adjuncts using a CLAN and you want the ESS server to use the same connectivity ● The main server connects to the adjuncts using the PE interface of the main server, and you want the LSP or ESS server to connect to the adjunct using its PE interface.
n(o)	This processor channel is disabled on the LSP or the ESS server. The survivable processor does not use this channel. This is the default.
o(verwrite)	The survivable processor overwrites the processor channel information sent in the file sync from the main server. The **o(verwrite)** option The **o(verwrite)** option allows the administered adjunct attributes to be modified uniquely for each individual LSP or ESS server. Use the remaining editable fields to enter the processor channel information for the LSP or ESS server.

Interface Channel

This field identifies the channel number or the TCP/IP listen port channel to carry this processor (virtual) channel. For TCP/IP, interface channel numbers are in the range **5000** to **64500**. Avaya recommends the value **5001** for CMS, and **5003** for DCS. This field changes to display-only when you enter **i(nherit)** in the **Enable** field, showing the value administered on the main server.

Valid entries	Usage
0, **5000** to **64500**	For **ethernet** or **ppp**. The channel number **0** means any port can be used.

Interface Link

This field identifies the communication interface link carrying this processor (virtual) channel. A **p** in this field indicates that the link is the Processor Ethernet interface. Otherwise, the CLAN link number is used. When you enter **o(verwrite)** in the **Enable** field, the value of this field changes to **p** (processor).

Mode

Indicate whether the IP session is passive (client) or active (server). This field changes to display-only when you enter **i(nherit)** in the **Enable** field, showing the value administered on the main server.

Valid entries	Usage
c(lient)	Indicate whether the IP session is passive **c(lient)** or active **s(erver)**.
s(erver)	

Proc Chan

This display-only field shows the processor channel number used for this link when it was administered on the **Processor Channel Assignment** screen.

Session - Local/Remote

Local and Remote Session numbers must be consistent between endpoints. These fields change to display-only when you enter **i(nherit)** in the **Enable** field, showing the values administered on the main server.

Valid entries	Usage
1 to **256** (si) **1** to **384** (r) or blank	For each connection, the Local Session number on this Avaya S8XXX Server must equal the Remote Session number on the remote server and vice versa. It is allowed, and sometimes convenient, to use the same number for the Local and Remote Session numbers for two or more connections.

Survivable Processor - IP Services page

Use this page of the Survivable Processor screen when an AESVCS or a CDR connects to the LSP or ESS server that was identified on the Survivable Processor - Processor Ethernet screen. If the AESVCS or the CDR is administered on the IP Services screen, it automatically appears on the Survivable Processor - IP-Services screen.

Figure 297: Survivable Processor - IP-Services screen

```
                                                                page 3 of x

                    SURVIVABLE PROCESSOR - IP-SERVICES

Service    Enabled   Store    Local        Local    Remote        Remote
 Type                to dsk    Node         Port     Node          Port
 CDR1         o        y     gert_clan6       0      cdr_1          9003
 CDR2         i        y     gert_clan1       0      cdr_rsp        9000
```

Enabled

Use this field to specify how data for each specified service type is transferred to the survivable processor.

Valid entries	Usage
i(nherit)	Enter **i(nherit)** to indicate that this link is to be inherited by the LSP or ESS server. When you enter **i(nherit)**, all fields in the row for this service type change to display-only. The survivable processor inherits this service type just as it is administered on the main server. You generally use the **i(nherit)** option in the following situations: ● The main server connects to the adjuncts using a CLAN and you want the ESS server to use the same connectivity ● The main server connects to the adjuncts using the main server's PE interface and you want the LSP or ESS server to connect to the adjunct using its PE interface
n(o)	This IP services link is disabled on the LSP or the ESS server. This is the default.
o(verwrite)	Enter **o(verwrite)** to overwrite the processor channel information sent in the file sync from the main server. The overwrite option allows the administered CDR or AE Services attributes to be modified uniquely for each individual LSP or ESS server. Entering **o(verwrite)** causes the **Local Node** field to change to **procr**. Use the remaining editable fields to enter the information for the LSP or ESS server.

Local Node

This display-only field contains the node name as defined on the **IP Node Names** screen.

Local Port

This display-only field contains the originating port number. For client applications such as Call Detail Recording (CDR), this field defaults to **0**.

Remote Node

This field becomes editable when you enter **o(verwrite)** in the **Enable** field. Specify the name at the far end of the link for the CDR. The remote node should not be defined as a link on the **IP Interfaces** or **Data Module** screens. This field does not apply for AESVCS.

Remote Port

This field becomes editable when you enter **o(verwrite)** in the **Enable** field. Specify the port number of the destination. Valid entries range from **5000** to **65500** for CDR or AESVCS. The remote port number must match the port administered on the CDR or AESVCS server.

Service Type

This field is display-only and reflects the value administered in the **Service Type** field on the IP Services screen. Valid entries include **CDR1 or CDR2**, and **AESVCS**.

Note:
For service type **CDR1** and **CDR2**, if the **Enable** field on the Survivable Processor - IP Services screen is **n**, the corresponding CDR1/CDR2 entry (for the Primary or Secondary CDR link) is removed from the System Parameters CDR screen on the LSP or the ESS server. The primary must be up and running before adding the secondary. The secondary must be removed first before removing the primary. On duplicated ESS, for CDR1/CDR2/AESVCS, the **Enabled** field defaults to **i**, and the rest of the fields are display-only.

Store to dsk

Use this field to enable or disable the storage of the CDR data on the local hard drive of the LSP or ESS. This column only pertains to rows which have the **Service Type** set to **CDR1** or **CDR2**.

Valid entries	Usage
y/n	Enter **y** to enable storage of CDR data on the local hard drive of an LSP or ESS.

Survivable Processor - IP-Services - Session Layer Timers page

This page appears if **CDR1** or **CDR2** is administered on the SURVIVABLE PROCESSOR - IP-SERVICES page of the Survivable Processor screen. These fields are only administrable if the **Enable** field on that page is set to **o(verwrite)**.

Figure 298: Survivable Processor - IP-Services - Session Layer Timers screen

```
                                                          page 4 of x

            SURVIVABLE PROCESSOR - IP-SERVICES - Session Layer Timers

Service     Reliable     Packet Resp     Session Connect    SPDU     Connectivity
 Type       Protocol       Timer          Message Cntr      Cntr         Time

 CDR1         n             30                3               3            60
 CDR2         y             30                3               3            60
```

Connectivity Time

Use this field to set the amount of time that the link can be idle before Communication Manager sends a connectivity message to ensure the link is still up. This field is only administrable if the **Enable** field on the SURVIVABLE PROCESSOR - IP-SERVICES page of the Survivable Processor screen is set to **o(verwrite)**.

Valid entries	Usage
1 to **255**	Enter the desired interval in seconds. Default is **60**.

Packet Resp Timer

Use this field to specify the number of seconds to wait from the time a packet is sent until a response (acknowledgement) is received from the far-end, before trying to resend the packet. This field is only administrable if the **Enable** field on the SURVIVABLE PROCESSOR - IP-SERVICES page of the Survivable Processor screen is set to **o(verwrite)**.

Valid entries	Usage
1 to **255**	Enter the desired interval in seconds. Default is **30**.

Reliable Protocol

Use this field to indicate whether you want to use a reliable protocol over this link. This field is only administrable if the **Enable** field on the SURVIVABLE PROCESSOR - IP-SERVICES page of the Survivable Processor screen is set to **o(verwrite)**.

Valid entries	Usage
y/n	Enter **y** to indicate that you want to Use reliable protocol if the adjunct on the far end of the link supports it. Default is **y**.

Service Type

This field is display-only and corresponds to the **Service Type** entry on the SURVIVABLE PROCESSOR - IP-SERVICES page of the Survivable Processor screen.

Session Connect Message Cntr

The Session Connect Message counter indicates the number of times Communication Manager tries to establish a connection with the far-end adjunct.

Valid entries	Usage
1 to **5**	Enter the desired number of connection attempts.

SPDU Cntr

The Session Protocol Data Unit (SPDU) counter indicates the number of times Communication Manager transmits a unit of protocol data before generating an error.

Valid entries	Usage
1 to **5**	Enter the desired number of transmission attempts.

System Capacity

The **System Capacity** screen (command `display capacity`) is described in *Maintenance Commands for Avaya Aura™ Communication Manager, Media Gateways and Servers*, 03-300431. Detailed system capacity information can be found in *Avaya Aura™ Communication Manager System Capacities Table*, 03-300511.

System Configuration

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

Figure 299: System Configuration screen

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

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

System Parameters Call Coverage/Call Forwarding

This screen sets the system-wide parameters for call coverage and call forwarding.

Field descriptions for page 1

Figure 300: System-Parameters — Call Coverage/Call Forwarding screen

```
change system-parameters coverage-forwarding                    Page   1 of   2
                SYSTEM PARAMETERS CALL COVERAGE / CALL FORWARDING

CALL COVERAGE/FORWARDING PARAMETERS

          Local Cvg Subsequent Redirection/CFWD No Ans Interval (rings): 2
        Off-Net Cvg Subsequent Redirection/CFWD No Ans Interval (rings): 2
                          Coverage - Caller Response Interval (seconds): 4
     Threshold for Blocking Off-Net Redirection of Incoming Trunk Calls: 1
                          Location for Covered and Forwarded Calls: called
                        PGN/TN/COR for Covered and Forwarded Calls: caller
                     COR/FRL check for Covered and Forwarded Calls? n
COVERAGE
                                        Keep Held SBA at Coverage Point? y
       External Coverage Treatment for Transferred Incoming Trunk Calls? n
        Immediate Redirection on Receipt of PROGRESS Inband Information? n
                                              Maintain SBA At Principal? y

                                           Station Hunt Before Coverage? n
FORWARDING
                                                  Call Forward Override? n
                                              Coverage After Forwarding? y
```

CALL COVERAGE / FORWARDING PARAMETERS

Coverage - Caller Response Interval (seconds)

The time in seconds the caller (internal caller only) has before a call redirects to the called party's first coverage point. The calling party either can hang up, use Leave Word Calling, or press the **Go to Cover** button during this time interval.

Valid entries	Usage
0 to **10**	Enter the time in seconds the caller (internal caller only) has before a call redirects to the called party's first coverage point.

Local Cvg Subsequent Redirection/CFWD No Ans Interval (rings)

This field specifies:

- the number of rings applied at a local coverage point before a call redirects to the next coverage point
- the number of rings applied at the principal before a call forwards when Call Forwarding Busy/Don't Answer is activated

Valid entries	Usage
1 to **99**	Enter the desired number of rings. **Note:** When ringing local destinations (that is, in an office environment), a short interval often is appropriate because the intended party either is near the telephone or not present. However, when ringing off-net locations, you cannot assume how near the intended party is to the telephone. If the call is left at an off-net destination for only a short interval, the call can be redirected to the next destination before the intended party has any real chance of answering the call.

Location for Covered and Forwarded Calls

Valid entries	Usage

called	Default.
	• If the called party is registered or in-service, coverage and forwarding use the called party's physical phone's location number.
	• If the called party is AWOH (x-port) or unregistered, coverage and forwarding use a location based on the type of set.
	  • IP set: location 1
	  • DCP set: location 0 (**all**)
	• When the forwarding or coverage destination is to UDP instead of to an external destination starting with the ARS FAC, routing is always based on the caller's physical phone's location regardless of how this field is administered.

caller	Coverage and forwarding use the caller's physical phone's location number.

Off-Net Cvg Subsequent Redirection/CFWD No Ans Interval (rings)

This field specifies:

- the number of rings applied at an off-net coverage point before a call is redirected to the next coverage point

- the number of rings applied at an off-net forwarded-to destination before the call is redirected to coverage.

Valid entries	Usage
1 to **99**	Enter the desired number of rings. **Note:** When ringing local destinations (that is, in an office environment), a short interval often is appropriate because the intended party either is near the telephone or not present. However, when ringing off-net locations, you cannot assume how near the intended party is to the telephone. If the call is left at an off-net destination for only a short interval, the call can be redirected to the next destination before the intended party has any real chance of answering the call.

Note:
When ringing local destinations (that is, in an office environment), a short interval often is appropriate because the intended party either is near the telephone or not present. However, when ringing off-net locations, you cannot assume how near the intended party is to the telephone. If the call is left at an off-net destination for only a short interval, the call can be redirected to the next destination before the intended party has any real chance of answering the call.

Threshold for Blocking Off-Net Redirection of Incoming Trunk Calls

This field applies for those occasions when an incoming call to a station redirects off-net. At that time, the Call Forward timer activates to block any further incoming calls to that station from being redirected off-net until the timer expires.

Valid entries	Usage
1 to **7**	The number of allowed calls to be routed off-net before blocking commences.
n (all)	Call processing never activates the Call Forward timer. Therefore, any number of calls to a principal can be redirected off-net.

COVERAGE

External Coverage Treatment for Transferred Incoming Trunk Calls

This field governs how an transferred incoming trunk call is handled if the call redirects to coverage.

Valid entries	Usage
y	Enter **y** to allow external coverage treatment for incoming trunk calls that redirect to coverage.
n	Enter **n** to allow internal coverage treatment for incoming trunk calls that redirect to coverage

Immediate Redirection on Receipt of PROGRESS Inband Information

This field appears only if one of the following is true:

- The **Coverage of Calls Redirected Off-Net Enabled** field on the System Parameters Coverage/Forwarding screen is **y**.
- The **Value-Added Avaya (VALU)** field on the System Parameters Customer-Options (Optional Features), Page 6, screen is **y**.

This field pertains only to CCRON and QSIG VALU coverage calls redirected over end-to-end ISDN facilities. Some cellular telephone providers send an ISDN PROGRESS message with the **Progress Indicator** field set to 'inband information' when a cellular telephone is unavailable to receive a call. In these circumstances, the message indicates that an announcement is being played to the originating party and we should move the call immediately to the next available coverage point.

However, a PROGRESS message with a Progress Indicator of 'inband information' might be received for reasons other than an unavailable cellular telephone. In this case, you probably do not want to redirect the call to the next coverage point.

There is no way for Communication Manager to know the actual intent of such a PROGRESS message, yet you might choose how the system should handle this message. It is essentially an educated, but blind, choice and you should be aware that there will be instances when this choice leads to inappropriate call handling.

Communication Manager queries this field on receipt of a qualifying PROGRESS message and acts according to your instruction on how to treat it.

As a guide, users in European countries following the ETSI standard and redirecting to GSM cellular telephones might want to consider setting this field to **y**.

In the United States, PROGRESS messages with the **Progress Indicator** field set to 'inband information' are sent for a variety of reasons not associated with unavailable cellular telephones and you should set this field to **n**.

Valid entries	Usage
y	Immediately redirect an off-net coverage/forwarded call to the next coverage point.
n	Do not immediately redirect an off-net coverage/forwarded call to the next coverage point.

Keep Held SBA at Coverage Point

This field governs how a covering user who has placed an answered coverage call on hold is treated if the original principal bridges onto the call.

Valid entries	Usage
y	Keeps the coverage party on the call. The coverage party remains on hold, but might enter the call along with the principal and the calling party.
n	Drops the coverage party from the call.

Maintain SBA At Principal

Allows a user to maintain a simulated bridged appearance when a call redirects to coverage.

Valid entries	Usage
y	Enter **y** to maintain a simulated bridged appearance (SBA) on the principal's telephone when a call redirects to coverage. DCS with rerouting will not be attempted after coverage.
n	When set to **n**, no SBA is maintained on the principal's telephone. DCS with rerouting will be attempted, and if successful, the principal will lose the bridged appearance and the ability to bridge onto the coverage call.

QSIG VALU Coverage Overrides QSIG Diversion with Rerouting

This field appears if, on the System Parameters Customer-Options (Optional Features) screen, the **Basic Supplementary Services** and **Supplementary Services with Rerouting** fields are both set to **y**.

This field specifies whether, with both QSIG Diversion with Rerouting and QSIG VALU turned on, the Coverage After Forwarding option on the Station screen will work for a user for calls that go to remote coverage. Normally, with QSIG Diversion with Rerouting turned on, the local system passes control of a forwarded call to the remote QSIG server on which the forwarding destination resides. In this case, the forwarded call cannot return to coverage for the user who originally received the call, even when the remote destination is busy or does not answer.

However, you can enter **y** in this field to have QSIG VALU call coverage take precedence. In this case, if the **Coverage After Forwarding** option on the Station screen is enabled for a user, then QSIG Diversion with Rerouting is not be attempted.

Valid entries	Usage
y/n	Enter **y** to allow Coverage After Forwarding to work when it is activated on a user's Station screen and Diversion with Rerouting is also turned on.

Station Hunt Before Coverage

This field allows you to choose whether a call to a busy station performs station hunting before going to coverage.

Valid entries	Usage
y/n	Enter **y** to use Station Hunt Before Coverage.

FORWARDING

Call Forward Override

This field specifies how to treat a call from a forwarded-to party to the forwarded-from party.

Valid entries	Usage
y	Overrides the Call Forwarding feature by allowing a forwarded-to station to complete a call to the forwarded-from station.
n	Directs the system to forward calls to a station even when they are from the forwarded-to party.

Coverage After Forwarding

This field governs whether an unanswered forwarded call is provided coverage treatment.

Valid entries	Usage
y	Coverage treatment is provided to unanswered forwarded calls.
n	No coverage treatment is provided to unanswered forwarded calls. The call remains at the forwarded-to destination.

For information on how **Coverage After Forwarding** works for QSIG calls, see the *Avaya Aura™ Communication Manager Feature Description and Implementation*, 555-245-205.

Field descriptions for page 2

Figure 301: System-Parameters Coverage-Forwarding screen

```
change system-parameters coverage-forwarding                     page 2 of x

           SYSTEM PARAMETERS -- CALL COVERAGE / CALL FORWARDING

COVERAGE OF CALLS REDIRECTED OFF-NET (CCRON)
                            Coverage Of Calls Redirected Off-Net Enabled? y
      Activate Answer Detection (Preserves SBA) On Final CCRON Cvg Point? y
                                       Ignore Network Answer Supervision? y
                      Disable call classifier for CCRON over ISDN trunks? n
                      Disable call classifier for CCRON over SIP trunks?  n

CHAINED CALL FORWARDING
                                Maximum Number Of Call Forwarding Hops: 3
                Station Coverage Path For Coverage After Forwarding: principal
```

COVERAGE OF CALLS REDIRECTED OFF-NET (CCRON)

Activate Answer Detection (Preserves SBA) On Final CCRON Cvg Point

This field appears only if the **Coverage of Calls Redirected Off-Net Enabled** field on this screen is **y**.

When the system redirects a call off-net at the final coverage point in a coverage path, the system can apply no further coverage treatment even if the call is unanswered. The only reason for activating answer detection on such a call is to maintain the simulated bridged appearance (SBA) on the principal's telephone that allows the principal to answer or bridge onto the call. However, when the system monitors the call through use of a call classifier port, there is an inherent cut-through delay following the detection of answer at the far end. This field has no consequence when the off-net call is carried end-to-end by ISDN facilities; the SBA is maintained and there is no cut-through delay.

Valid entries	Usage
y	Directs the system to maintain a simulated bridged appearance on the principal when redirecting to a final off-net coverage point.
n	Allows the system to drop the SBA on the principal's telephone when the call redirects off-net at the last coverage point, eliminating the cut-through delay inherent in CCRON calls, but sacrificing the principal's ability to answer the call.

Coverage Of Calls Redirected Off-Net Enabled

This field allows you to enable/disable the Coverage of Calls Redirected Off-Net (CCRON) feature. This field provides a quick means of disabling this feature if the demand on the call classifier port resources degrades other services provided by Communication Manager.

To set this field to **y**, first set the **Coverage Of Calls Redirected Off-net** field on the System Parameters Customer-Options (Optional Features) screen to **y**. The **Coverage of Calls Redirected Off-Net** field on this screen must be **y** to administer this field.

Valid entries	Usage
y	Communication Manager monitors off-net coverage/forwarded calls and provides further coverage treatment for unanswered calls.
n	Communication Manager does not monitor off-net coverage/forwarded calls. No further coverage treatment is provided if such calls are unanswered.

Disable call classifier for CCRON over ISDN trunks

When a CCRON call routes offnet over ISDN end-to-end facilities, no call classifier is attached to the call. If, subsequently during the call, an ISDN PROGRESS or ALERT message is received that indicates that interworking has occurred, a call classifier is normally attached to the call and assumes precedences over ISDN trunk signalling. This field provides a customer the means of directing Communication Manager to dispense with the call classifier on interworked calls and rely on the ISDN trunk signalling messages.

Valid entries	Usage
y	Use **y** to disable the call classifier for CCRON calls over interworked trunk facilities.
n	Use **n** to enable the call classifier for CCRON calls over interworked trunk facilities.

Disable call classifier for CCRON over SIP Enablement Services (SES) trunks

This field provides a customer the means of directing Communication Manager to dispense with the call classifier on interworked calls and rely on the SIP Enablement Services (SES) trunk signalling messages.

Valid entries	Usage
y	Use **y** to disable the call classifier for CCRON calls over interworked trunk facilities.
n	Use **n** to enable the call classifier for CCRON calls over interworked trunk facilities.

Ignore Network Answer Supervision

This field appears only if the **Coverage of Calls Redirected Off-Net Enabled** field on this screen is **y**.

CCRON might use a call classifier port to determine whether an off-net coverage or forwarded call has been answered, discarding other information that might indicate an answered state. However, some customers pay the operating company to provide network answer supervision on their trunks and desire that CCRON not discard that information. This service can be preserved by setting this field to **n**.

On the other hand, beware when you tandem a call over a tie trunk through another server node from where it redirects to the public network over non-ISDN facilities. If the trunk on the far-end node sends a timed answer supervision, that might get tandemed back to the originating node as a network answer. In such a scenario, the originating server interprets the call as answered, leading to some undesirable behavior. To avoid these calls from mistakenly be treated as answered, set this field to **y**. An unfortunate consequence is that a short cut-through delay that is inherent to call classification is introduced when the call is answered.

Valid entries	Usage
y	Ignore network answer supervision and rely on the call classifier to determine when a call is answered.
n	Treat network answer supervision as a true answer.

CHAINED CALL FORWARDING

Maximum Number Of Call Forwarding Hops

This field appears only if the **Chained Call Forwarding** field on the System-Parameters Features screen is **y**.

Valid entries	Usage
3 to 10	Enter the number of hops allowed in the forwarding chain.

Station Coverage Path For Coverage After Forwarding

This field appears only if the **Chained Call Forwarding** field on the System-Parameters Features screen is **y**.

Valid entries	Usage
	Specify the coverage path that you want the call to follow.

System Parameters CDR (Call Detail Recording)

See CDR System Parameters.

System Parameters Country-Options

This screen implements parameters associated with certain international (including North American) call characteristics. You cannot change this screen. See your Avaya technical support representative if you want to modify any of the values here.

The following table shows the country codes that are used in Communication Manager. The Country Code is used by various fields and screens throughout the system.

Country code table

Code	Country	Ringing Signal Voltage, Frequency, and Cadence
1	United States, Canada, Korea, India	300v peak to peak, < 200v peak to ground; < 70 Hz; < 5s on > 1s off Korea: 20 Hz, 75 to 85 Volts (AC), Cadence: 1 sec on, 2 sec off
2	Australia, New Zealand	75 +/- 20 VRMS superimposed on 48 V dc at 14.5 to 55 Hz with cadence 400ms on, 200ms off, 400ms on, 2000ms off New Zealand: Ringing voltage at the customer's premises not less than 38 V rms (25Hz) on top of 50V d.c; 20 Hz; 400ms on, 200ms off, 400ms on, 2000ms off
3	Japan	75 VRMS(75-10VRMS <= x <= 75+8VRMS), 15-20 Hz and cadence of 1second on and 2 seconds off is required
4	Italy	20 to 50 Hz, 26 to 80 Volts rms superimposed on 48 V dc, 1 sec on, 4 sec off ETSI countries: 30 Volts rms, superimposed on a DC voltage of 50 Volts, 25 or 50 hz, cadence of 1 sec on, 5 sec off
5	Netherlands	25 Hz, 35 to 90 Volts rms superimposed on 66 V dc, 1 sec on, 4 sec off. Note that 50 Hz is recommended, and another cadence may be 0.4 sec on, 0.2 sec off, 0.4 sec on, 4 sec off ETSI countries: 30 Volts rms, superimposed on a DC voltage of 50 Volts, 25 or 50 hz, cadence of 1 sec on, 5 sec off

1 of 4

Code	Country	Ringing Signal Voltage, Frequency, and Cadence
6	Singapore	75V at 24Hz with a cadence of 0.4 seconds on, 0.2 seconds off, 0.4 seconds on and 2.0 seconds off.
7	Mexico	25 Hz, 70 +/- 20 Vrms superimposed on 48Vdc Cadence 1 sec on, 4 sec off, flashhook is 100 ms
8	Belgium, Luxembourg, Korea	25 Hz, 25 to 75 Volts rms superimposed on 48 V dc, 1 sec on, 3 sec off Korea: 20 Hz, 75 to 85 Volts (AC), Cadence: 1 sec on, 2 sec off ETSI countries: 30 Volts rms, superimposed on a DC voltage of 50 Volts, 25 or 50 hz, cadence of 1 sec on, 5 sec off
9	Saudi Arabia	
10	United Kingdom	U.K.: 15 to 26.25 Hz, 25 to 100 Volts rms superimposed on 48 V dc, 0.35 on, 0.22 off then start in at any point in: 0.4 sec on, 0.2 sec off, 0.4 sec on, 2 sec off. Note 1: 48v DC may be present during the whole cadence or may be confined to silent periods. Note 2: Some exchanges provide a facility known as immediate ring; in this case an initial burst of ringing 20 msec to 1 sec in length immediately precedes switching to any point in the normal ringing cycle. Ireland: 25 Hz, 30 to 90 Volts rms superimposed on 50 V dc, 0.4 sec on, 0.2 sec off, 0.4 sec on, 2 sec off another possible cadence is 0.375 sec on, 0.250 sec off, 0.375 sec on, 2 sec off. ETSI countries: 30 Volts rms, superimposed on a DC voltage of 50 Volts, 25 or 50 hz, cadence of 1 sec on, 5 sec off
11	Spain	20 to 30 Hz, 35 to 75 Volts rms superimposed on 48 V dc, 1 to 1.5 sec on, 3 sec off ETSI countries: 30 Volts rms, superimposed on a DC voltage of 50 Volts, 25 or 50 hz, cadence of 1 sec on, 5 sec off
12	France	50 Hz, 28 to 90 Volts rms superimposed on 0.45 to 54 V dc, 1.5 sec on, 3.5 sec off ETSI countries: 30 Volts rms, superimposed on a DC voltage of 50 Volts, 25 or 50 hz, cadence of 1 sec on, 5 sec off
		2 of 4

Code	Country	Ringing Signal Voltage, Frequency, and Cadence
13	Germany	Germany: 25 Hz, 32 to 75 Volts rms superimposed on 0 to 85 V dc, 1 sec on, 4 sec off Austria: 40 to 55 Hz, 25 to 60 Volts rms superimposed on 20 to 60 V dc, 1 sec on, 5 sec off +/- 20% ETSI countries: 30 Volts rms, superimposed on a DC voltage of 50 Volts, 25 or 50 hz, cadence of 1 sec on, 5 sec off
14	Czech Republic, Slovakia	
15	Russia (CIS)	25+\-2 Hz, 95+\-5 Volts eff, local call cadence: first ring 0.3-4.5 sec then 1 second on 4 seconds Off toll automatic cadence 1 sec On 2 sec Off toll operator: manual sending
16	Argentina	25Hz; 75 Vrms superimposed on 48 Vdc; 1s on 4s off
17	Greece	
18	China	25Hz +/- 3Hz; 75 +/- 15 Vrms; Harmonic Distortion <= 10%; 1 sec ON, 4 secs OFF
19	Hong Kong	75 +/- 20 VRMS superimposed on -40 to -48 V dc at 25 Hz +/- 10% with cadence 0.4 s on, 0.2 s off, 0.4 s on, 3.0 s off
20	Thailand	
21	Macedonia	
22	Poland	
23	Brazil	25Hz +/-2.5Hz; minimum of 40 Vrms; 1s on, 4s off for equipment supporting up to six trunks only otherwise 25Hz +/-2.5Hz; minimum of 70+/-15 Vrms at a continuous emitting condition under no load, overlapping a DC level.
		3 of 4

Code	Country	Ringing Signal Voltage, Frequency, and Cadence
24	Nordic	Finland: 25 Hz, 35 to 75 Volts rms superimposed on 44 to 58 V dc, 1 sec on, 4 sec off 25 Hz, 40 to 120 Volts rms superimposed on 44 to 56 V dc, 0,75 on, 7,5 off +/- 20 % 25 Hz, 28 to 90 Volts rms superimposed on 24 to 60 V dc, 1 sec on, 4 sec off 25 and 50 Hz, 30 to 90 Volts rms superimposed on 33 to 60 V dc, 1 sec off, 5 sec off ETSI countries: 30 Volts rms, superimposed on a DC voltage of 50 Volts, 25 or 50 hz, cadence of 1 sec on, 5 sec off
25	South Africa	

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Field descriptions for page 1

Figure 302: System Parameters Country-Options screen

```
change system-parameters country-options                        Page 1 of x

                        SYSTEM PARAMETERS COUNTRY-OPTIONS

                       Set Layer 1 timer T1 to 30 seconds? n
                                    Display Character Set? Ukrainian
                              Directory Search Sort Order: Cyrillic
                                   Howler Tone After Busy? n
                     Disconnect on No Answer by Call Type? n
 Enable Busy Tone Disconnect for Analog Loop-start Trunks? n

TONE DETECTION PARAMETERS
        : 5       dial tone validation timer (sec): 60
          Interdigit Pause: short
```

Dial Tone Validation Timer (sec)

This field appears only when **Tone Detection Mode** is equal to **4** or **5**. (Valid with TN420C or later Tone Detector circuit pack.).

Valid entries	Usage
0 to **6375** in increments of 25	Displays number of milliseconds that the dial tone validation routine will use to sample transmissions.

Directory Search Sort Order

This field appears only when **Display Character Set** on the System Parameters Country-Options screen is set to **Cyrillic** or **Ukrainian**.

Tip:
You can toggle to the unadministered value for a single search session by first pressing the pound key (#) on your telephone dial pad. Subsequent sessions return to the administered value.

Valid entries	Usage
Cyrillic	Cyrillic Collation is used for integrated directory name search and result sorting. This is the default value.
Roman	Eurofont Latin Collation is used for directory name search and result sorting. The letters to be searched in the specified order for dial pad button presses are defined in the row for each key.

Disconnect on No Answer by Call Type

Drops outgoing trunk calls (except DCS and AAR) that users leave unanswered too long.

Valid entries	Usage
y/n	Enables the system to disconnect calls that are not answered.

Display Character Set

The value in this field determines the character set used for all name values that do not have an ASCII-only restriction.

Note that the setting for this field can affect the [Display Language](#) field on the Station screen for Unicode-capable stations. Specifically, if **Display Character Set** is set to **katakana**, the **Display Language** field for 4624 sets will allow an entry for **Unicode**, which is required for Toshiba SIP Telephone (TSP) sets. If **Display Character Set** is not set to **katakana**, then Toshiba SIP telephones will not operate correctly in Japan.

WARNING:
Changing the value in this field might cause some telephones to perform improperly, and will cause non-ASCII data in non-native names to display incorrectly on telephones. To correct this, you must remove the non-native names of previously administered station(s) and re-administer them, together with any display messages that have been administered, to use non-ASCII characters.

Valid entries	Usage
Cyrillic **Katakana** **Roman** **Ukrainian**	Indicate the enhanced character set to display. See Telephone Display in *Avaya Aura™ Communication Manager Feature Description and Implementation,* 555-245-205, for more information. **Note: Cyrillic**, **Roman**, and **Ukrainian** map to the Eurofont character set. For **Katakana**, the Optrex font is used. If a Communication Manager server uses non-English in any name field, characters on a BRI station are not displayed correctly.

Enable Busy Tone Disconnect for Analog Loop-start Trunks

This field allows Communication Manager to recognize a busy tone from the central office as a disconnect signal.

Valid entries	Usage
y/n	Enter **y** to allow Communication Manager to disconnect the trunk when a busy tone is received from the central office.

Howler After Busy

Plays howler tone when users leave their analog telephone off-hook too long.

Valid entries	Usage
y/n	Enables howler tone.

Set Layer 1 timer T1 to 30 seconds

Valid entries	Usage
y/n	Specifies whether the Layer 1 timer is set to 30 seconds.

TONE DETECTION PARAMETERS

Interdigit Pause

Specifies the maximum length of the inter-digit pause. Breaks lasting less than this range will be bridged or ignored. (Valid with TN420C or later Tone Detector circuit pack.)

Valid entries	Usage
short	5 to 30ms
long	20 to 40ms

Tone Detection Mode

Use this field to specify the type of tone-detection algorithm that Communication Manager uses for a TN420B or later tone-detection circuit pack.

Valid entries	Usage
1	The precise Italian tone-detection algorithm.
2	The precise Australian tone-detection algorithm.
3	The precise UK tone-detection algorithm.
4	The imprecise normal broadband-filter algorithm. This option is valid only for a TN420C or later tone-detection circuit pack.
5	The imprecise wide broadband-filter algorithm. This option is valid only for a TN420C or later tone-detection circuit pack.
6	The precise USA tone-detection algorithm.

System Parameters Customer-Options (Optional Features)

This screen shows you which optional features are enabled for your system, as determined by the installed license file. The fields on this screen are populated by the license file, and are display only. If you have any questions about disabling or enabling one of these features, contact your Avaya representative.

Field descriptions for page 1

Figure 303: System Parameters Customer-Options **(Optional Features) screen**

```
display system-parameters customer-options                      Page   1 of  x
                                OPTIONAL FEATURES

G3 Version: V12  123456789012                   Software Package: Standard
  Location: 2                                RFA System ID (SID):
  Platform: 2                                RFA Module ID (MID): 123456

                                                              USED
                                Platform Maximum Ports: 300   174
                                      Maximum Stations: 300   174
                              Maximum XMOBILE Stations: 30    28
                    Maximum Off-PBX Telephones - EC500: 1200  0
                    Maximum Off-PBX Telephones -   OPS: 1200  0
                    Maximum Off-PBX Telephones - SCCAN: 0     0
                    Maximum Off-PBX Telephones - PBFMC: 0     0
                    Maximum Off-PBX Telephones - PVFMC: 0     0

        (NOTE: You must logoff & login to effect the permission changes.)
```

G3 Version

Identifies the version of Communication Manager being used.

Location

Indicates the location of this Avaya S8XXX Server. 1 indicates Canada or the United States. 2 indicates any other location, and allows the use of International Consolidation circuit packs and telephones.

Maximum Off-PBX Telephones - EC500

Valid entries	Usage
0 to license max	Stations that are administered for any Extension to Cellular (EC500/CSP) application count against this limit. Default is **0**. This field is administrable when H.323, IDSN-PRI, ISDN-BRI Trunks or Multifrequency Signaling is enabled on the System Parameters Customer-Options **(Optional Features)** screen (**`display system-parameters customer-options`**). The “license max” value is defined as follows: ● On legacy systems, the upper limit is 1/2 of the maximum number of administrable stations. Legacy platforms do not support SIP Enablement Services (SES) trunks. ● On Linux systems, the upper limit is the maximum number of administrable stations.

Maximum Off-PBX Telephones - OPS

Valid entries	Usage
0 to license max	Stations that are administered for any SIP Extension to Cellular/OPS application count against this limit. Default is **0**. This field is administrable when SIP is enabled on the System Parameters Customer-Options **(Optional Features)** screen (**`display system-parameters customer-options`**). The “license max” upper limit is: ● On legacy systems, 1/2 of the maximum number of administrable stations. Legacy platforms do not support SIP Enablement Services (SES) trunks. ● On Linux systems, the maximum number of administrable stations.

Maximum Off-PBX Telephones - PBFMC

Valid entries	Usage
0 to license max	Number of stations administered for Public Fixed-Mobile Convergence. Each station is allowed only one PBFMC application. Default is **0**. This field is administrable when H.323, IDSN-PRI, ISDN-BRI Trunks or Multifrequency Signaling is enabled on the System Parameters Customer-Options **(Optional Features)** screen (`display system-parameters customer-options`). The “license max” value is defined as follows: ● On legacy systems, the upper limit is 1/2 of the maximum number of administrable stations. Legacy platforms do not support SIP Enablement Services (SES) trunks. ● On Linux systems, the upper limit is the maximum number of administrable stations.

Maximum Off-PBX Telephones - PVFMC

Valid entries	Usage
0 to license max	Number of stations administered for Private Fixed-Mobile Convergence. Each station is allowed only one PVFMC application. Default is **0**. This field is administrable when SIP Trunks are enabled on the System Parameters Customer-Options **(Optional Features)** screen (`display system-parameters customer-options`). The “license max” upper limit is: ● On legacy systems, 1/2 of the maximum number of administrable stations. Legacy platforms do not support SIP Enablement Services (SES) trunks. ● On Linux systems, the maximum number of administrable stations.

Maximum Off-PBX Telephones - SCCAN

Valid entries	Usage
0 to license max	The “license max” value is defined as follows: ● SCCAN is only available on Linux systems. The upper limit is the maximum number of administrable stations. ● Stations that are administered for any Extension to Cellular/OPS application count against this limit. Default is **0**.

Maximum Stations

Displays the maximum number of stations allowed in the system. This feature is set based on the Communication Manager license file. Default is **0**.

Maximum XMOBILE Stations

Specifies the maximum number of allowable XMOBILE stations. In general, each XMOBILE station is assigned to a wireless handset. Each XMOBILE station counts as a station and a port in terms of system configuration.

Platform

A display-only field that identifies, via a number mapping, the platform being used for a specific customer. Valid values and server types are:

Platform number	Server type
6	S87XX Server
8	S87XX Server
12	S8500 Server
14	S87XX ESS Server
15	S8500 ESS server

Platform Maximum Ports

Number of ports active, per contract.

Software Package

Indicates whether the software package license is **Standard** or **Enterprise**.

Used

Shows the actual current usage as compared to the system maximum for each field.

Field descriptions for page 2

Figure 304: System Parameters Customer-Options **(Optional Features) screen**

```
display system-parameters customer-options                       page 2 of x

                              OPTIONAL FEATURES

IP PORT CAPACITIES                                              USED
                       Maximum Administered H.323 Trunks:
                          Maximum Administered IP Trunks: 100   96
             Maximum Concurrently Registered IP Stations: 10    10
               Maximum Administered Remote Office Trunks: 0     0
  Maximum Concurrently Registered Remote Office Stations: 0     0
                Maximum Concurrently Registered IP eCons: 0     0
                          Maximum Video Capable Stations: 0     0
                     Maximum Video Capable IP Softphones: 0     0
                         Maximum Administered SIP Trunks: 500   25
    Maximum Administered Ad-hoc Video Conferencing Ports: 0     0
     Maximum Number of DS1 Boards with Echo Cancellation: 0     0
                                Maximum TN2501 VAL Boards: 1     0
                      Maximum G250/G350/G700 VAL Sources: 0     0
             Maximum TN2602 Boards with 80 VoIP Channels: 20    12
            Maximum TN2602 Boards with 320 VoIP Channels: 4     3

     Maximum Number of Expanded Meet-me Conference Ports: 0      0

           (NOTE: You must logoff & login to effect the permission changes.)
```

IP PORT CAPACITIES

Maximum Administered Ad-hoc Video Conferencing Ports

Defines the number of ad-hoc ports allowed for the system; one for each simultaneous active conference port. The maximum number of ad-hoc video conferencing ports allowed is the sum of the maximum allowed IP trunks and the maximum allowed SIP trunks on your system. Default is **O**. The **IP Trunks** field on the System Parameters Customer Options screen must also be set to **y**.

Maximum Administered IP Trunks

Defines limits of the number of IP trunks administered.

Maximum Administered Remote Office Trunks

Defines limits of the number of IP endpoints based on the endpoint. Use the smaller of this number or the number based on the MAXIMUM IP REGISTRATIONS BY PRODUCT ID page of this screen.

Maximum Administered SIP Trunks

Defines limits on the number of SIP Enablement Services (SES) trunks administered.

Maximum Concurrently Registered IP eCons

Specifies the maximum number of IP SoftConsoles that can be registered at one time. The maximum number depends on the type of system.

Maximum Concurrently Registered IP Stations

Specifies the maximum number of IP stations that can be registered at one time. This field accepts 6,000 concurrently registered IP stations for the S87XX series servers, and 3,000 for S8500 servers.

Maximum G250/G350/G700 VAL Sources

Specifies the maximum number of VAL announcement sources.

Maximum Number of DS1 Boards with Echo Cancellation

Shows the number of DS1 circuit packs that can have echo cancellation.

Maximum Number of Expanded Meet-me Conference Ports

Displays the license-file based value of the system maximum for the number of Expanded Meet-me Conference ports.

Maximum TN2501 VAL Boards

This display-only field indicates the maximum number of TN2501AP (Voice Announcement over LAN) boards allowed in this system.

Valid entries	Usage
0 to **10** (S87XX Servers) **0** to **5** (DEFINITY CSI, and S8300 Servers)	• For values greater than 1, the **Val Full 1-Hour Capacity** field on page 4 of the System Parameters Customer-Options (Optional Features) screen must be set to **y**. • This field updates the **System Limit** field on the System Capacity report.

Maximum TN2602 Boards with 80 VoIP Channels

Valid entries	Usage
0 to license truncation limit.	This field defines the total number of TN2602AP boards that can be administered with 80 VoIP channels. The value is based on the value in the Communication Manager license file.The **USED** value is the total number of TN2602AP boards in the system administered with 80 VoIP channels. Default is **0**.

Maximum TN2602 Boards with 320 VoIP Channels

Valid entries	Usage
0 to license truncation limit.	This field defines the total number of TN2602AP boards that can be administered with 320 VoIP channels. The value is based on the value in the Communication Manager license file. The **USED** value is the total number of TN2602AP boards in the system administered with 320 VoIP channels. Default is **0**.

Maximum Video Capable Stations

Specifies the maximum number of stations that are video-capable. The maximum number depends on the type of system.

Maximum Video Capable IP Softphones

Specifies the maximum number of IP Softphones that are video-capable. The maximum number depends on the type of system.

Used

For each item with a capacity listed, the **USED** value is the actual number of units currently in use.

Field descriptions for page 3

Figure 305: System Parameters Customer-Options **(Optional Features) screen**

```
display system-parameters customer-options                       page 3 of x

                         OPTIONAL FEATURES

     Abbreviated Dialing Enhanced List?             Audible Message Waiting?
         Access Security Gateway (ASG)?                 Authorization Codes?
         Analog Trunk Incoming Call ID?                          CAS Branch?
   A/D Grp/Sys List Dialing Start at 01?                           CAS Main?
  Answer Supervision by Call Classifier?                  Change COR by FAC?
                                    ARS?    Computer Telephony Adjunct Links?
                  ARS/AAR Partitioning?      Cvg Of Calls Redirected Off-net?
           ARS/AAR Dialing without FAC?                         DCS (Basic)?
           ASAI Link Core Capabilities?                   DCS Call Coverage?
           ASAI Link Plus Capabilities?                  DCS with Rerouting?
        Async. Transfer Mode (ATM) PNC?
Async. Transfer Mode (ATM) Trunking?        Digital Loss Plan Modification?
               ATM WAN Spare Processor?                            DS1 MSP?
                                  ATMS?              DS1 Echo Cancellation?
                  Attendant Vectoring?
```

Abbreviated Dialing Enhanced List

Provides the capability to store and retrieve dialing lists that simplify or eliminate dialing. You dial an abbreviated code or depress an assigned button. The stored entries are organized in number lists. There are three types of number lists: personal, group, and enhanced.

Access Security Gateway (ASG)

Provides an additional level of security for remote administration.

A/D Grp/Sys List Dialing Start at 01

Allows you to number Abbreviated Dialing group or system lists starting with 01, rather than simply 1. This allows Abbreviated Dialing under Communication Manager to operate like it did with the DEFINITY G2 system.

Analog Trunk Incoming Call ID

This field allows collection and display the name and number of an incoming call information on analog trunks.

Answer Supervision by Call Classifier

This circuit pack detects tones and voice-frequency signals on the line and determines whether a call has been answered. This field is set to **y** if the system contains a call-classifier circuit pack.

ARS

Provides access to public and private communications networks. Long-distance calls can be routed over the best available and most economical routes. Provides partitioning of ARS routing patterns.

ARS/AAR Partitioning

Provides the ability to partition AAR and ARS into 8 user groups within a single server running Communication Manager. Can establish individual routing treatment for each group.

ARS/AAR Dialing without FAC

Provides for Automatic Alternate Routing (AAR) and Automatic Route Selection (ARS) calls without dialing a feature access code (FAC).

CAUTION:
Contact your Avaya technical support representative to discuss details of this feature before enabling it.

ASAI Link Core Capabilities

Provides linkage between Communication Manager and adjuncts. CallVisor ASAI improves the call handling efficiency of ACD agents and other system users by allowing an adjunct to monitor, initiate, control, and terminate calls on the server running Communication Manager.

Note:
ASAI Link Core Capabilities only applies to links administered as type **asai**. This field was previously named **ASAI Interface**.

If the **ASAI Link Core Capabilities** field is administered to **y** then it will be associated with the following ASAI capability groups:

- Adjunct Control
- Domain Control
- Event Notification
- Single Step Conference
- Request Feature
- II Digits
- Set Value
- Value Query

For more information, see the *Avaya Communication Manager ASAI Technical Reference*, 555-230-220.

ASAI Link Plus Capabilities

Provides linkage between Communication Manager and adjuncts. If the **ASAI Link Plus Capabilities** field is administered to **y**, then the following ASAI capability groups are enabled:

- Adjunct Routing
- Answering Machine Detection
- Selective Listening
- Switch Classified Outbound Calls
- ISDN Redirecting Number Information - the original dialed number information is provided within the ASAI messages if it arrives in ISDN SETUP messages from the public networks as either Original Dialed Number or Redirecting Party Number.

Note:
ASAI Link Plus Capabilities only applies to links administered as type **asai**.

For more information, see the *Avaya Communication Manager ASAI Technical Reference*, 555-230-220.

Asynch. Transfer Mode (ATM) PNC

ATM PNC can be enabled only if:

- all prior fiber-link administration has been removed
- all "switch-node" and "dup-switch-node" carrier types have been removed

Asynch. Transfer Mode (ATM) Trunking

If ATM trunking is enabled, multiple ISDN-PRI T1 or E1 trunks can be emulated on one ATM pipe. Can only be enabled if the **ISDN-PRI** field is set to **y**. Enables circuit emulation service (CES).

ATM WAN Spare Processor

An ATM WAN spare processor acts as a PPN in the event of network failure, and can function as an SPE if the main PPN is not functional. Cannot be set to **y** if the **Async. Transfer Mode (ATM) Trunking** field is **n**.

ATMS

Provides for voice and data trunk facilities to be measured for satisfactory transmission performance.

Attendant Vectoring

Allows you to use attendant vectoring. Cannot be set to **y** if the **CAS Main** and **CAS Branch** fields are **y**.

Audible Message Waiting

Provides audible message waiting.

Authorization Codes

Permits you to selectively specify levels of calling privileges that override in-place restrictions. In addition to facilities access, authorization codes are used for unique identification for billing security purposes.

CAS Branch

Provides Centralized Attendant Service - Branch. See **CAS Main** for more information. Cannot be set to **y** if the **Attendant Vectoring** is **y** and **Centralized Attendant** on the QSIG OPTIONAL FEATURES page of the System Parameters Customer Options screen is **y**.

CAS Main

Provides multi-location customers served by separate switching vehicles to concentrate attendant positions at a single, main Communication Manager location. The main Communication Manager is served by an attendant queue that collects calls from all locations (main and branch). Each branch location switches all of its incoming calls to the centralized attendant positions over release link trunks (RLTs). The calls are then extended back to the requested extension at the branch server/switch over the same RLT. When the call is answered, the trunks to the main server are dropped and can be used for another call. Cannot be set to **y** if the **Centralized Attendant** and **CAS Branch** fields are **y**.

Change COR by FAC

Provides certain users the ability to change the class of restriction of local extensions through a telephone by using a feature access code (FAC). Cannot be set to **y** if the **Tenant Partitioning** field is **y**.

Computer Telephony Adjunct Links

Provides linkage between Communication Manager and adjuncts. Includes both the ASAI Link Core and ASAI Link Plus capabilities, plus the Phantom Calls and CTI Stations.

Note:
Computer Telephony Adjunct Links only applies to links administered as type **adjlk**. This field was previously named **ASAI Proprietary Adjunct Links**.

For more information see the *Avaya Communication Manager ASAI Technical Reference*, 555-230-220.

Cvg Of Calls Redirected Off-net

Provides continued monitoring for calls redirected to off-network (remote) coverage points. Uses call classification via call classifier circuit pack or ISDN trunk signaling.

DCS (Basic)

Provides transparent operation of selected features across a Distributed Communications System (DCS). Users on one server running Communication Manager can use features located on another server. Includes 4- and 5-digit uniform dialing and 1 to 4 digit steering. Does not support a 6/7-digit dial plan.

DCS Call Coverage

Provides DCS-based transparency of the call coverage feature across a DCS network of servers.

DCS with Rerouting

Provides for rerouting calls transferred among DCS nodes, enabling rerouting of the call for more effective use of facilities. Cannot be set to **y** if the **ISDN PRI** field is **n**.

Digital Loss Plan Modification

Allows you to customize the digital loss and digital tone plans.

DS1 MSP

Provides the ability to change fields on DS1 Circuit Pack screen without removing the related translations of all trunks from the trunk group.

DS1 Echo Cancellation

Removes perceivable echo from the system.

Field descriptions for page 4

Figure 306: System Parameters Customer-Options **(Optional Features) screen**

```
display system-parameters customer-options                      Page   4 of  x
                                OPTIONAL FEATURES

  Emergency Access to Attendant? y                              IP Stations? y
          Enable 'dadmin' Login? y
          Enhanced Conferencing? y                        ISDN Feature Plus? y
                 Enhanced EC500? y      ISDN/SIP Network Call Redirection? y
   Enterprise Survivable Server? y
      Enterprise Wide Licensing? y                          ISDN-BRI Trunks? y
             ESS Administration? y                                 ISDN-PRI? y
         Extended Cvg/Fwd Admin? y               Local Survivable Processor? y
    External Device Alarm Admin? y                     Malicious Call Trace? y
         Extended Cvg/Fwd Admin? y    Mode Code for Centralized Voice Mail? y
    External Device Alarm Admin? y
 Five Port Networks Max per MCC? y                 Multifrequency Signaling? y
               Flexible Billing? y Multimedia Appl. Server Interface(MASI)? y
  Forced Entry of Account Codes? y       Multimedia Call Handling (Basic)? y
     Global Call Classification? y    Multimedia Call Handling (Enhanced)? y
            Hospitality (Basic)? y               Multimedia IP SIP Trunking? y
Hospitality (G3V3 Enhancements)? y
                      IP Trunks? y

          IP Attendant Consoles? y

           (NOTE: You must logoff & login to effect the permission changes.)
```

Emergency Access to Attendant

Provides for emergency calls to be placed to an attendant. These calls can be placed automatically by Communication Manager or dialed by users.

Enable 'dadmin' Login

Provides business partners the ability to install, administer, and maintain S87XX Servers and DEFINITY switches. The *dadmin* login has access to all the same commands as other logins with the exception of **`Go`** and **`WP`**. **`Go`** is used for **`go tcm`** and **`go debug`** as well as **`go server`**. WP is for writing memory.

The *dadmin* login has access to all commands available to the craft service level logins and to the following commands that are available to inads service level logins:

Action	Object/Qualifier
set	alarm options
enable/disable	all
status/clear	cumulative audits
enable/disable	disabled test
copy	disk/tape
add/change/remove	eda-external-device-alm
restart	hardware-group
cancel	hardware-group
display	internal data
change	isdn-network facilities
add/change/remove/ display/list/enable/disable	login
enable/disable/clear/ status/list	message trace
enable/disable/test	MO
change/display	mst
status/clear	peak audits
change/display	permissions
test	range-test hardware-group
display	software errors
monitor	system conn
monitor	system scr
change	system-parameters wireless
change	system-parameters maintenance (Customer Access to Inads Port field)

The *dadmin* login does *not* have access to the following commands that are available to the inads login:

Action	Object/Qualifier
add/remove	prec
upload/download	translations
go	debugger
go	tcm
wp/rp	byte/angel
wva/rva	process

Enhanced Conferencing

Enhanced Conferencing allows the customer to use the Meet-me Conference, Expanded Meet-me Conference, Selective Conference Party Display, Drop, and Mute, and the No Hold Conference features. Must be **y** to enable the Enhanced Conferencing features.

Enhanced EC500

Enables Extension to Cellular for administration. “EC500” refers to the Extension to Cellular feature. When this field is set to **y**, all screens under the `off-pbx-telephone` commands are available.

Enterprise Survivable Server

This display-only field is activated through the license file. When this field is set to **y**, this server is an Enterprise Survivable Server (ESS). For more information about ESS, see *Using Avaya Enterprise Survivable Servers (ESS)*, 03-300428.

ESS Administration

This display-only field enables administration of Enterprise Survivable Servers (ESS) on the System Parameters – Port Networks screen. For more information about ESS, see *Using Avaya Enterprise Survivable Servers (ESS)*, 03-300428.

External Device Alarm Admin

Provides for analog line ports to be used for external alarm interfaces. Allows identification of port location, adjunct associated with port location, and the alarm level to report.

Enterprise Wide Licensing

Enterprise Wide Licensing. See your Avaya representative for more information.

Five Port Networks Max Per MCC

Available only for S87XX Series Fiber-PNC. Allows system administrator to create five port networks in a multi-carrier cabinet. If there are any cabinets with more than two PNs assigned, this field cannot be set to **n**.

Flexible Billing

Provides an internationally accepted standard interface for end-to-end digital connectivity. Used with a T1 interface and supports twenty-three 64-KBPS voice or data B-Channels and one 64-Kbps signaling D Channel for total bandwidth of 1.544 Mbps.

Forced Entry of Account Codes

Allows system administration to force account users to enter account codes based on user or trunk class of restriction, or by an option on the Toll Analysis table. FEAC provides an easy method of allocating the costs of specific calls to the correct project, department, etc.

Global Call Classification

Provides call classification outside of North America. Listens for tones and classifies tones detected. Required for Call Coverage Off Net and Outgoing Call Management.

Hospitality (Basic)

Provides access to basic features including: Attendant Crisis Alert, Attendant Room Status, Automatic Wakeup, Custom Selection of VIP DID Numbers, Do Not Disturb, Names Registration, Single-Digit Dialing, and Mixed Station Numbering.

Hospitality (G3V3 Enhancements)

Software required for Property Management System and Automatic Wakeup. Property Management System Interface activates Forward PMS Messages to INTUITY Lodging and PMS Protocol Mode (transmit in ASCII mode). Cannot be set to **y** if the **Hospitality (Basic)** field is **n**.

Note:
Standard hospitality features are included in basic system software.

IP Attendant Consoles

Controls permission to administer the IP Attendant Console.

IP Stations

Controls permission to administer H.323 and/or SoftPhone stations. Must be **y** for IP telephones.

IP Trunks

Controls permission to administer H.323 trunks. Must be **y** for IP trunks.

ISDN Feature Plus

Provides ISDN Feature Plus signaling. This option is enabled when either the **ISDN-BRI Trunks** field or the **ISDN-PRI** field is **y**.

ISDN/SIP Network Call Redirection

Administrable if the **ISDN-PRI** or **ISDN-BRI Trunk** field is **y**. Network Call Redirection (NCR) redirects an incoming ISDN call from a server running Communication Manager to another PSTN endpoint. It is used in Call Centers with Best Service Routing and Lookahead Interflow.

ISDN-BRI Trunks

Provides the capability to add ISDN-BRI trunks to Communication Manager. If enabled, can add isdn trunk groups and the following screens are accessible:

- network-facilities
- private-numbering
- public-unknown- numbering

ISDN-PRI

Provides Integrated Services Digital Network (ISDN-PRI) software for either a switching-hardware platform migration only or a switching-hardware platform migration in combination with a software release upgrade. Also provides signaling support for H.323 signaling. Must be **y** for IP and SIP trunks.

Local Survivable Processor

This display-only field indicates that the server is a Local Survivable Processor (LSP). When this field is set to **y**, the LSP server is configured to provide standby call processing in case the primary server is unavailable.

Malicious Call Trace

Provides ability to retrieve certain information related to a malicious call.

Mode Code for Centralized Voice Mail

This feature provides the ability to share a Voice Mail System (VMS) among several servers/ switches using the Mode Code - Voice Mail System Interface.

Multifrequency Signaling

Provides for a screen of number signaling used between Communication Manager and the central office.

Multimedia Appl. Server Interface (MASI)

Allows users of the Multimedia Communications Exchange (MMCX) to take advantage of certain Communication Manager telephony features.

Multimedia Call Handling (Basic)

Allows administration of desktop video-conferencing systems as data modules associated with Communication Manager voice stations in a multimedia complex. Users can dial one number to reach either endpoint (voice or data) in the complex. Also provides support for IP SoftPhones.

Multimedia Call Handling (Enhanced)

Allows a multifunction telephone to control a multimedia call like a standard voice call.

Multimedia IP SIP Trunking

If enabled, extends applicability of the H.323 video station licensing/control to all non-ip-softphones.

Field descriptions for page 5

Figure 307: System Parameters Customer-Options **(Optional Features) screen**

```
display system-parameters customer-options                          page 5 of x

                                OPTIONAL FEATURES

                 Multinational Locations?           Station and Trunk MSP? n
Multiple Level Precedence and Preemption?      Station as Virtual Extension? n
                      Multiple Locations?
                                             System Management Data Transfer? n

         Personal Station Access (PSA)? y
                       Posted Messages? n               Tenant Partitioning? n
                       PNC Duplication? n       Terminal Trans. Init. (TTI)? y
                  Port Network Support? y               Time of Day Routing? y
              Processor and System MSP? n              Uniform Dialing Plan? y
                    Private Networking? y    Usage Allocation Enhancements? y
                    Processor Ethernet? y      TN2501 VAL Maximum Capacity? y

                         Remote Office? n               Wideband Switching? y
         Restrict Call Forward Off Net? y                         Wireless? n
                 Secondary Data Module? y
```

Multinational Locations

The Multinational Locations feature provides you with the ability to use a single Enterprise Communication Server (ECS) with stations, port networks, remote offices, or gateways in multiple countries. With this feature enabled, you can administer location parameters such as companding, loss plans, and tone generation per location, instead of system-wide.

Multiple Level Precedence and Preemption

Multiple Level Precedence and Preemption (MLPP) provides users the ability to assign levels of importance to callers, and when activated, to give higher-priority routing to individual calls based on the level assigned to the caller.

Multiple Locations

Allows you to establish numbering plans and time zone and daylight savings plans that are specific for each cabinet in a port network.

Personal Station Access (PSA)

Provides basic telecommuting package capability for Personal Station Access.

Posted Messages

Supports users being able to post messages, which they select from among a set of as many as 30 (15 fixed, 15 administrable), to be shown on display telephones.

PNC Duplication

If set to **y**, the **Enable Operation of PNC (Port Network Connectivity) Duplication** field appears on the System Parameters Duplication screen. The **Enable Operation of PNC Duplication** field is set with the **Enable Operation of SPE (Switch Processing Element) Duplication** field to provide non-standard reliability levels (high, critical, or ATM PNC Network Duplication).

Port Network Support

Indicates that the server is operating as a stand-alone Internal Communications Controller (ICC) when set to **n** and is used to disable traditional port networking. Set to **y** to indicate that traditional Avaya DEFINITY port networks are in use.

Private Networking

Upgrades PNA or ETN software RTU purchased with earlier systems.

Processor Ethernet

Appears only on S8300, S8400, and S8500 Servers. Used to enable use of the Ethernet card resident in the processor cabinet for use by the DEFINITY Call Processing software in place of a Control Lan (C-LAN) card (located in a port network). The Processor Ethernet interface is always enabled for S87XX Servers.

Processor and System MSP

Allows the customer administrator or technician to maintain processor and system circuit packs.

Remote Office

Allows administration of a remote office.

Restrict Call Forward Off Net

The system can monitor the disposition of an off-call and, if it detects busy, bring the call back for further processing, including call coverage.

Secondary Data Module

Provides ability to use any data module as a secondary data module.

Station and Trunk MSP

Provides the customer administrator or technician to maintain station and trunk circuit packs.

Station as Virtual Extension

Allows **virtual** to be entered in the **Type** field of the Station screen, which allows multiple virtual extensions to be mapped to a single physical analog telephone. The user can also administer a specific ringing pattern for each virtual extension. Useful in environments such as college dormitories, where three occupants can have three different extensions for one physical telephone.

System Management Data Transfer

Indicates Communication Manager is accessible by Network Administration.

Tenant Partitioning

Provides for partitioning of attendant groups and/or stations and trunk groups. Typically this is used for multiple tenants in a building or multiple departments within a company or organization.

Terminal Trans. Init. (TTI)

Allows administrators of Terminal Translation Initialization (TTI) to merge an station administered with **X** in the **Port** field, to a valid port by dialing a system-wide TTI security code and the extension from a terminal connected to that port. Must be set to **y** for Automatic Customer Telephone Rearrangement.

Time of Day Routing

Provides AAR and ARS routing of calls based on the time of day and day of the week. You can take advantage of lower calling rates during specific times.

TN2501 VAL Maximum Capacity

If this is enabled, you have the Enhanced offer, which allows up to 60 minutes storage capacity per pack and multiple integrated announcement circuit packs.

Uniform Dialing Plan

Provides 3- to 7-digit Uniform Dial Plan (UDP) and 1 to 7 digit steering. Also allows you to use Extended Trunk Access and Extension Number Portability features.

Usage Allocation Enhancements

Provides for assigning ISDN-PRI or ISDN-BRI Services/Features for Usage Allocation Plans. To use this enhancement, first set either the **ISDN-PRI** or **ISDN-BRI Trunks** fields to **y**.

Wideband Switching

Provides wideband data software for switching video or high-speed data. You can aggregate DSO channels up to the capacity of the span. Wideband supports H0, H11, and H12 standards, where applicable, as well as customer-defined data rates.

Wireless

Provides right to use for wireless applications in certain Network Systems sales. You can purchase it from Avaya Network Wireless Systems.

Field descriptions for Call Center Optional Features

Figure 308: Call Center Optional Features screen

```
  display system-parameters customer-options                        page 6 of x

                         CALL CENTER OPTIONAL FEATURES

                           Call Center Release:

                                 ACD? y    PASTE (Display PBX Data on Phone)? n
                        BCMS (Basic)? y                         Reason Codes? n
                                                   Service Level Maximizer? y
          BCMS/VuStats Service Level? n             Service Observing (Basic)? y
                   Business Advocate? n    Service Observing (Remote/By FAC)? n
                     Call Work Codes? y              Service Observing (VDNs)?
       DTMF Feedback Signals For VRU? y                               Timed ACW?
                    Dynamic Advocate? n                     Vectoring (Basic)? y
        Expert Agent Selection (EAS)? y                 Vectoring (Prompting)? y
                             EAS-PHD? n             Vectoring (G3V4 Enhanced)?
                   Forced ACD Calls? n    Vectoring (ANI/II-Digits Routing)? n
               Least Occupied Agent?      Vectoring (G3V4 Advanced Routing)? n
           Lookahead Interflow (LAI)?                       Vectoring (CINFO)? n
Multiple Call Handling (On Request)? n     Vectoring (Best Service Routing)? n
    Multiple Call Handling (Forced)? n                   Vectoring (Holidays)?
```

ACD

Automatic Call Distribution (ACD) automatically distributes incoming calls to specified splits or skills. Provides the software required for the Call Center Basic, Plus, Deluxe, and Elite features for the number of agents specified. Cannot be set to **n** if the **Call Work Codes** field is **y**.

BCMS (Basic)

Provides real-time and historical reports about agent, ACD split, Vector Directory Number (VDN) and trunk group activity.

BCMS/VuStats Service Level

Allows you to set up hunt groups or Vector Directory Numbers (VDNs) with an acceptable service level. An acceptable service level defines the number of seconds within which a call must be answered to be considered acceptable.

Business Advocate

Software that provides an integrated set of advanced features to optimize call center performance. If set to **n**, the **Least Occupied Agent** field displays. For information on Business Advocate, contact your Account Executive.

Call Center Release

Displays the call center release installed on the system.

Call Work Codes

Allows agents to enter digits for an ACD call to record customer-defined events such as account codes or social security numbers. Cannot be set to **y** if the **ACD** field is **n**.

DTMF Feedback Signals For VRU

Provides support for the use of C and D Tones to VRUs.

Dynamic Advocate

Software that provides an integrated set of advanced features to optimize call center performance.

EAS-PHD

Increases the number of skills an agent can log in to from four to 20. Increases the number of agent skill preference levels from two to 16.

Expert Agent Selection (EAS)

Provides skills-based routing of calls to the best-qualified agent.

Forced ACD Calls

See **Multiple Call Handling**.

Least Occupied Agent

Appears only if the **Business Advocate** field is **n**. Allows call center calls to be routed to the agent who has been the least busy, regardless of when the agent last answered a call. Cannot be set to **y** if the **Expert Agent Selection (EAS)** field is **n**.

Lookahead Interflow (LAI)

Provides Look-Ahead Interflow to balance the load of ACD calls across multiple locations. Cannot be set to **y** if the **Vectoring (Basic)** field is **n**.

Multiple Call Handling (On Request)

Allows agents to request additional calls when active on a call.

Multiple Call Handling (Forced)

Forces an agent to be interrupted with an additional ACD call while active on an ACD call. Splits or skills can be one forced, one per skill, or many forced. Cannot be set to **y** if the **ACD** field is **n** and the **Forced ACD Calls** field is **y**.

PASTE (Display PBX Data on Phone)

Provides an interface between the display of a DCP telephone set and PC-based applications.

Reason Codes

Allows agents to enter a numeric code that describes their reason for entering the AUX work state or for logging out of the system. Cannot be set to **y** if the **Expert Agent Selection (EAS)** field is **n**.

Service Level Maximizer

Allows an administrator to define a service level whereby X% of calls are answered in Y seconds. When Service Level Maximizer (SLM) is active, the software verifies that inbound calls are matched with agents in a way that ensures that the administered service level is met. SLM is used with Expert Agent Selection (EAS), and without Business Advocate. Call Center Release must be 12 or later.

Service Observing (Basic)

Allows a specified user to observe an in-progress call on a listen-only or listen-and-talk basis.

Service Observing (Remote/By FAC)

Allows users to service observe calls from a remote location or a local station using this feature’s access codes.

Service Observing (VDNs)

Provides the option of observing and/or monitoring another user's calls.

Timed ACW

Places an auto-in agent in ACW for an administered length of time after completion of the currently active ACD call.

Vectoring (Basic)

Provides basic call vectoring capability.

Vectoring (Prompting)

Allows flexible handling of incoming calls based on information collected from the calling party or from an ISDN-PRI message.

Vectoring (G3V4 Enhanced)

Allows the use of enhanced comparators, wildcards in digit strings for matching on collected digits and ANI or II-digits, use of Vector Routing Tables, multiple audio/music sources for use with wait-time command and priority level with the oldest-call-wait conditional.

Vectoring (ANI/II-Digits Routing)

Provides for ANI and II-Digits vector routing.

Vectoring (G3V4 Advanced Routing)

Provides for Rolling Average Speed of Answer Routing, Expected Wait Time Routing, and VDN Calls Routing.

Vectoring (CINFO)

Provides the ability to collect ced and cdpd from the network for vector routing. To use this enhancement, first set either the **ISDN-PRI** or **ISDN-BRI Trunks** fields to **y**.

Vectoring (Best Service Routing)

Enables the Best Service Routing feature. Through special vector commands, Best Service Routing allows you to compare splits or skills at local and remote locations and queue a call to the resource that will give the caller the best service.

Vectoring (Holidays)

Enables the Holiday Vectoring feature. It simplifies vector writing for holidays.

Field descriptions for Call Center Optional Features

Figure 309: Call Center Optional Features screen

```
display system-parameters customer-options                    Page 7 of x
                       CALL CENTER OPTIONAL FEATURES

     VDN of Origin Announcement? n                              VuStats? n
         VDN Return Destination? n              VuStats (G3V4 Enhanced)? n








                                  Used
               Logged-In ACD Agents: 500
          Logged-In Advocate Agents: 500
      Logged-In IP Softphone Agents: 500
```

Logged-In ACD Agents

Number of ACD Agents contracted for. This field limits the number of logged-in ACD agents to a number no more than the maximum purchased. The value of this field indicates the total of ACD agents that can be logged-in simultaneously.

The limit applies to ACD agents on ACD and EAS calls. Auto-Available Split (AAS) agent ports are counted when they are assigned. AAS split or skill members are also counted. If the port for an AAS split/skill member is logged out, (for example, when a ringing call is redirected) the logged-in agent count is not updated. These counts are updated only during administration.

Logged-In Advocate Agents

Appears when the **Business Advocate** field is **y**. Number of Business Advocate Agents contracted for.

The total number of logged-in Business Advocate agents must be equal to or less than the number allowed in the **Logged-In ACD Agents** field. The number of logged-in Business Advocate agents counts towards the total number of logged-in ACD agents.

Logged-In IP Softphone Agents

Number of IP Softphone Agents contracted for. This field limits the number of logged-in IP Softphone agents to a number no more than the maximum purchased. The value of this field indicates the total of IP Softphone agents that can be logged-in simultaneously.

VDN of Origin Announcement

Provides a short voice message to an agent indicating the city of origin of the caller or the service requested by the caller based on the VDN used to process the call.

VDN Return Destination

Allows an incoming trunk call to be placed back in vector processing after all parties, except the originator, drop.

VuStats

Allows you to present BCMS statistics on telephone displays.

VuStats (G3V4 Enhanced)

Allows you to use the G3V4 VuStats enhancements including historical data and thresholds.

Field descriptions for QSIG Optional Features

Figure 310: QSIG Optional Features screen

```
display system-parameters customer-options                       Page 8 of x
                            QSIG OPTIONAL FEATURES

                                         Basic Call Setup? n
                             Basic Supplementary Services? n
                                    Centralized Attendant? n
                                    Interworking with DCS? n
                    Supplementary Services with Rerouting? n
                            Transfer into QSIG Voice Mail? n
                                       Value-Added (VALU)? n
```

Basic Call Setup

Provides basic QSIG services: basic connectivity and calling line ID number. To use this enhancement, either the **ISDN-PRI** or **ISDN-BRI Trunks** fields must be **y**.

Basic Supplementary Services

To use this enhancement, either the **ISDN-PRI** or **ISDN-BRI Trunks** fields must be **y**. Provides the following QSIG Supplementary Services:

- Name ID
- Transit Capabilities; that is, the ability to tandem QSIG information elements
- Support of Notification Information Elements for interworking between QSIG and non-QSIG tandemed connections
- Call Forwarding (Diversion) by forward switching. No reroute capabilities are provided
- Call Transfer by join. No path replacement capabilities are provided.
- Call Completion (also known as Automatic Callback)

Centralized Attendant

Can be enabled only if the **Supplementary Services with Rerouting** field is **y**. Cannot be set to **y** if the **CAS Main** and **CAS Branch** fields are **y**. Allows all attendants in one location to serve users in multi locations. All signaling is done over QSIG ISDN lines. If this field is **y**, the **IAS** fields on the Console Parameters screen do not display.

Interworking with DCS

Allows the following features to work between a user on a DCS-enabled server in a network and a QSIG-enabled server:

- Calling/Called/Busy/Connected Name
- Voice Mail/Message Waiting
- Leave Word Calling

This field cannot be set to **y** if the **DCS (Basic)** field is **n**.

Supplementary Services with Rerouting

Provides the following QSIG Supplementary Services:

- Transit Capabilities; that is, the ability to tandem QSIG information elements.
- Support of Notification Information Elements for interworking between QSIG and non-QSIG tandemed connections.

- Call Forwarding (Diversion) by forward switching. In addition, reroute capabilities are provided.
- Call Transfer by join. In addition, path replacement capabilities are provided.

Transfer Into QSIG Voice Mail

Can be enabled only if the **Basic Supplementary Services** field is **y** and either the **ISDN-PRI Trunk** or **ISDN-BRI Trunk** field is **y**. Allows transfer directly into the voice-mail box on the voice-mail system when a QSIG link connects Communication Manager and the voice-mail system.

Value Added (VALU)

Provides additional QSIG functionality, including the ability to send and display calling party information during call alerting.

Field descriptions for ASAI

Figure 311: ASAI Features screen when the ASAI Link Plus Capabilities field is y

```
change system-parameters customer options                      Page 8 of X

                                  ASAI FEATURES

                                 CTI Stations? n
                                Phantom Calls? n



                            ASAI PROPRIETARY FEATURES

                                 Agent States? n
```

Agent States

Appears when the **Computer Telephony Adjunct Links** field is **y**. The **Agent States** field provides proprietary information used by Avaya applications. For more information, contact your Avaya technical support representative.

Note:
The **Agent States** field only applies to links administered as type **adjlk**. This field was previously named **Proprietary Applications**.

CTI Stations

Appears when the **ASAI Link Plus Capabilities** field is **y**. This field needs to be enabled for any application (using a link of Type ASAI) that uses a CTI station to receive calls.

For more information see the *Avaya Communication Manager ASAI Technical Reference*, 555-230-220.

Phantom Calls

Appears when the **ASAI Link Plus Capabilities** field is **y**. Enables phantom calls. The **Phantom Calls** field only applies to links administered as type ASAI.

For more information see the *Avaya Communication Manager ASAI Technical Reference*, 555-230-220.

Field descriptions for Maximum IP Registrations by Product ID

Figure 312: Maximum IP Registrations by Product ID screen

```
                                                                   Page 9 of x
                      MAXIMUM IP REGISTRATIONS BY PRODUCT ID

Product ID_Rel. Limit      Product ID_Rel. Limit      Product ID_Rel. Limit
__________ ___. _____      __________ ___. _____      __________ ___. _____
__________ ___. _____      __________ ___. _____      __________ ___. _____
__________ ___. _____      __________ ___. _____      __________ ___. _____
__________ ___. _____      __________ ___. _____      __________ ___. _____
__________ ___. _____      __________ ___. _____      __________ ___. _____
__________ ___. _____      __________ ___. _____      __________ ___. _____
__________ ___. _____      __________ ___. _____      __________ ___. _____
```

Limit

Maximum number of IP registrations allowed.

Valid entries	Usage
1000 or **5000**, depending on your server configuration	Maximum number of IP registrations allowed. For Avaya R300 Remote Office Communicator, defaults to the maximum allowed value for the **Concurrently Registered Remote Office Stations** on page 1 of this screen.

Product ID

Identifies the product using the IP (internet protocol) registration.

Valid entries*	Usage
Avaya_IR	Interactive Response product
IP_Agent	IP Agents
IP_eCons	SoftConsole IP attendant
IP_Phone	IP Telephones
IP_ROMax	R300 Remote Office telephones
IP_Soft	IP Softphones
*These are just a few examples of valid Product IDs. The valid Product IDs for your system are controlled by the license file.	

Rel

Release number of the IP endpoint.

Valid entries	Usage
0 to **99** or blank	Release number of the IP endpoint

System Parameters - Duplication

Use the System Parameters Duplication screen to enable PNC or IPSI duplication.

Field descriptions for page 1

Figure 313: System Parameters - Duplication screen

```
change system-parameters duplication                          Page 1 of x

                DUPLICATION RELATED SYSTEM PARAMETERS

 Enable Operation of PNC Duplication? y
Enable Operation of IPSI Duplication? y
```

Enable Operation of IPSI Duplication

Use this field to enable IPSI duplication.

Note:
This field is set to **n(o)** when either the TN8412AP or TN2312BP circuit pack is used in an S8400 configuration. This is because TN8412/TN2312 duplication is not supported in Phase 1 of the S8400. Duplication may be offered in the future.

Valid entries	Usage
y/n	Enter **y** to enable IPSI duplication.

Enable Operation of PNC Duplication

Valid entries	Usage
y/n	Enter **y** to enable PNC duplication. Appears when **PNC Duplication** is **y** on the System Parameters Customer-Options **(Optional Features)** screen.

System Parameters - Features

See Feature-Related System Parameters.

System Parameters - IP Options

See **IP-Options System Parameters**.

System Parameters - Maintenance

This screen is described in *Maintenance Commands for Avaya Aura™ Communication Manager, Media Gateways and Servers*, 03-300431.

System Parameters Media Gateway Automatic Recovery Rule

This screen is used to define rules for returning a fragmented network, where a number of H.248 Media Gateways are being serviced by one or more Local Survivable Processors (LSPs), to the primary Avaya S8XXX Server in an automated fashion. The system displays a different warning message and/or time window grid depending on the option selected for the **Migrate H.248 MG to primary** field. The following figures show the screens that appear for each option.

For more information on Auto Fallback for H.248 Gateways, see *Administering Network Connectivity on Avaya Aura™ Communication Manager*, 555-233-504.

Field descriptions for page 1

Figure 314: System Parameters Media Gateway Automatic Recovery Rule screen (immediately)

```
change system-parameters mg-recovery-rule 1

         SYSTEM PARAMETERS MEDIA GATEWAY AUTOMATIC RECOVERY RULE

Recovery Rule Number: 1
Rule Name: Rule 1
Migrate H.248 MG to primary:   immediately
Minimum time of network stability: 3

WARNING: The MG shall be migrated at the first possible opportunity. The MG may be
migrated with a number of active calls. These calls shall have their talk paths
preserved, but no additional call processing of features shall be honored. The
user must hang up to regain access to all features.






Note: set 'Migrate H.248 MG to primary' to Blank to disable rule.
```

Figure 315: Media Gateway Automatic Recovery Rule Time Entry screen (0-active-calls)

```
change system-parameters mg-recovery-rule 1

         SYSTEM PARAMETERS MEDIA GATEWAY AUTOMATIC RECOVERY RULE

Recovery Rule Number: 1
Rule Name: Rule 1
Migrate H.248 MG to primary:   0-active-calls
Minimum time of network stability: 3

WARNING: The MG shall only be migrated when there are no active calls.





Note: set 'Migrate H.248 MG to primary' to Blank to disable rule.
```

Figure 316: Media Gateway Automatic Recovery Rule Time Entry screen (time-day-window)

```
change system-parameters mg-recovery-rule 1

         SYSTEM PARAMETERS MEDIA GATEWAY AUTOMATIC RECOVERY RULE

Recovery Rule Number: 1
Rule Name: Rule 1
Migrate H.248 MG to primary:   time-day-window
Minimum time of network stability: 3

WARNING: The MG may be migrated with a number of active calls. These calls shall
have their talk paths preserved, but no additional call processing of features
shall be honored. The user must hang up in order to regain access to all features.
Valid registrations shall only be accepted during these intervals.


                                        Time of Day
Day of Week  00                               12                              23
Sunday       _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _
Monday       _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _
Tuesday      _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _
Wednesday    _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _
Thursday     _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _
Friday       _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _
Saturday     _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _

Note: set 'Migrate H.248 MG to primary' to Blank to disable rule.
```

Figure 317: Media Gateway Automatic Recovery Rule Time Entry screen (time-window OR 0-active-calls)

```
change system-parameters mg-recovery-rule 1

         SYSTEM PARAMETERS MEDIA GATEWAY AUTOMATIC RECOVERY RULE

Recovery Rule Number: 1
Rule Name: Rule 1
Migrate H.248 MG to primary:   time-window-OR-0-active-calls
Minimum time of network stability: 3

WARNING: The MG shall be migrated ANY time when there are no active calls OR the
MG may be migrated with a number of active calls when a registration is received
during the specified intervals. These calls shall have their talk paths
preserved, but no additional call processing of features shall be honored.


                                        Time of Day
Day of Week  00                              12                              23
Sunday       __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __
Monday       __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __
Tuesday      __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __
Wednesday    __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __
Thursday     __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __
Friday       __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __
Saturday     __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __

Note: set 'Migrate H.248 MG to primary' to Blank to disable rule.
```

Migrate H.248 MG to primary

Use this field to indicate auto-fallback preferences. For each option the system displays a unique warning message and/or time window grid.

You must specify an **x** or **X** for each hour during which you want to permit the return migration. If you do not want to permit a given hour, then they leave it blank. A blank for any given hour indicates that migration is not permitted during that hour. This method helps with overlapping time issues between days of the week. You can specify as many intervals as you wish. The Time of Day value indicates the exact time of day when a migration shall be permitted for each day of the week.

Valid entries	Usage
immediately	The first media gateway registration that comes from the media gateway is honored, regardless of call count or time of day. this is the default.
0-active calls	The first media gateway registration reporting “0 active calls” is honored.
time-day-window	A valid registration message received during any part of this interval is honored. When this option is selected the system displays a grid for defining desired hours/days for the time window.
time-window-OR-0-active-calls	A valid registration is accepted anytime, when a 0 active call count is reported OR if a valid registration with any call count is received during the specified time/day intervals. When this option is selected the system displays a grid for defining desired hours/days for the time window.

Minimum time of network stability

Use this field to administer the time interval for stability in the H.248 link before auto-fallback is attempted.

Valid entries	Usage
3 to **15**	Enter the number of minutes before auto-fallback is attempted. Default is **3**.

Recovery Rule Number

Valid entries	Usage
1 to server maximum	Enter the number of the recovery rule.

Rule Name

Use this field for an optional text name for the rule, as an aid in associating rules with media gateways.

Valid entries	Usage
Alpha-numeric characters	Enter a name for this recovery rule.

System Parameters - Mode Code

See **Mode Code Related System Parameters**.

System Parameters - Multifrequency Signaling

See **Multifrequency-Signaling-Related Parameters**.

System Parameters OCM Call Classification

This screen enters the tone characteristics for your country for Outbound Call Management (OCM) applications. It is not required for United States OCM applications. If you cannot access this screen, contact your Avaya technical support representative.

This screen appears when **Global Call Classification** field on the **System Parameters Customer-Options (Optional Features)** screen is set to **y**, or when the **Enable Busy Tone Disconnect for Analog loop-start Trunks** field on the System Parameters Country Options screen is set to **y**. This screen defines the busy tone and cadence and can be administered with up to 4 on and off steps, which is four valid cycles to determine busy tone.

Avaya recommends that you use a minimum of two on and off steps to determine a valid busy tone. If the cadence is administered with one on and off step, any time the classifier hears the cadence it is considered BTD signal.

Field descriptions for page 1

Figure 318: System Parameters OCM Call Classification screen

```
                                                               Page 1 of x
                    SYSTEM PARAMETERS OCM CALL CLASSIFICATION


TONE DETECTION PARAMETERS
  Global Classifier Adjustment (dB): ___
               USA Default Algorithm? _n_
                   USA SIT Algorithm? ___
```

Global Classifier Adjustment (dB)

Enter a number to specify the dB loss adjustment.

Valid entries	Usage
0 to **15**	**0** is the least and **15** the most adjustment.

USA Default Algorithm

Valid entries	Usage
y/n	To use the default United States tone detection, set this field to **y**. If you enter **n**, the **US Special Information Tones (SIT) Algorithm** field appears.

USA SIT Algorithm

Valid entries	Usage
y	To use the United States (SIT) tone characteristics for SIT tone detection.
n	The system treats tones with the administered tone name "intercept" as if they were SIT VACANT, and treats tones with the administered tone name "information" as if they were SIT UNKNOWN.

Field descriptions for page 2

Figure 319: System Parameters OCM Call Classification screen

```
                                                               Page 2 of x
                  SYSTEM PARAMETERS OCM CALL CLASSIFICATION

Tone Name  Instance      Tone       Cadence   Duration     Duration
                      Continuous    Step      Minimum      Maximum
_________   _____          ___      1.  on     _____        _____
                                    2.  off    _____        _____
                                    3.  on     _____        _____
                                    4.  off    _____        _____
                                    5.  on     _____        _____
                                    6.  off    _____        _____
                                    7.  on     _____        _____
                                    8.  off    _____        _____
```

Cadence Step

A display-only field identifying the number of each tone cadence step and indicating whether the tone is on or off during this cadence step.

Duration Maximum

Specifies the upper limit in milliseconds of the tone duration.

Note:
On the Feature-Related System Parameters screen, set the **Off-Premises Tone Detect Timeout Interval** field to its maximum value.

Valid entries	Usage
75 to **6375**	Enter in increments of 25 msec.

Duration Minimum

Specifies the lower limit in milliseconds (msec) of the tone duration.

Valid entries	Usage
75 to **6375**	Enter in increments of 25 msec.

Instance

Enter the instance number of the tone. If the system identifies a tone that matches the characteristics defined on more than one page of this screen the system applies the tone definition from the earlier page.

Valid entries	Usage
1 to **8**	The number distinguishes tones that have the same name but more than one definition of silence and tone-on characteristics.

Tone Continuous

Valid entries	Usage
y	Indicates a continuous tone. If you enter **y**, you cannot enter data in the **Duration Minimum** or **Duration Maximum** fields.
n	Indicates a non-continuous tone.

Tone Name

This field is required for tone definition outside of the U.S. and Canada.

If the **Global Call Classification** field on the System Parameters Customer-Options (Optional Features) screen is **n**, only busy can be entered into this **Tone Name** field. If **Busy Tone Disconnect** is enabled, only busy can be entered into this field.

Valid entries	Usage
busy **information** **intercept** **reorder** **ringback**	Enter the name of the tone that you are adding or modifying. Enter **busy** for **Busy Tone Disconnect**.

System Parameters – Port Networks

Use these screens to assign port networks to communities and to specify recovery rules for port networks to return to the main server.

Field descriptions for page 1

Figure 320:

```
change system-parameters port-networks                          Page   1 of   2

               COMMUNITY ASSIGNMENTS FOR PORT NETWORKS


 PN Community      PN Community      PN Community      PN Community     PN Community
 ------------      ------------      ------------      ------------     ------------
  1: 1             14: 1             27: 1             40: 1            53: 1
  2: 1             15: 1             28: 1             41: 1            54: 1
  3: 1             16: 1             29: 1             42: 1            55: 1
  4: 1             17: 1             30: 1             43: 1            56: 1
  5: 1             18: 1             31: 1             44: 1            57: 1
  6: 1             19: 1             32: 1             45: 1            58: 1
  7: 1             20: 1             33: 1             46: 1            59: 1
  8: 1             21: 1             34: 1             47: 1            60: 1
  9: 1             22: 1             35: 1             48: 1            61: 1
 10: 1             23: 1             36: 1             49: 1            62: 1
 11: 1             24: 1             37: 1             50: 1            63: 1
 12: 1             25: 1             38: 1             51: 1            64: 1
 13: 1             26: 1             39: 1             52: 1
```

PN

Valid entries	Usage
1 to **64**	Displays the port network.

Community

Valid entries	Usage
1 to **64**	Enter the Network Community number you want to associate with this port network. **Note:** If the port network is administered in the system, the default community is **1** and administrable with a value between **1** and **64**. If the port network is not administered in the system, the community value is **1** and not administrable.

Field descriptions for page 2

Figure 321: Port Network Recovery Rules screen

```
change system-parameters port-networks                          Page   2 of   x

                    PORT NETWORK RECOVERY RULES

 FAILOVER PARAMETERS                         FALLBACK PARAMETERS

 No Service Time Out Interval (min): 5                  Auto Return: no

    PN Cold Reset Delay Timer (sec): 60
```

Auto Return

The Auto Return functionality is used to schedule a day and time for all Port Networks to return to the control of the Main server after a failover occurs.The schedule can be set up to seven days prior to its activation.

Valid entries	Usage
y(es)	When the value is set to **y(es)**, the **IPSI Connection up time** field appears. When **Auto Return** is set to **y(es)**, the port networks can automatically return to the main server after the value set in the **IPSI Connection up time** expires.
n(o)	**Auto Return** is disabled. When the value is set to **n(o)**, the port networks cannot automatically return to the control of the main server. No additional fields appear when the value is set to **n(o)**.
s(cheduled)	**Auto Return** is enabled. When set to **s**, the **Day** and **Time** fields appear. Schedule a day and time to return the port networks to the control of the main server. The schedule can be set up to seven days prior to its activation. ● **Day**: Enter the day of the week ● **Time**: Enter the time of day in a 24 hour (military) format

No Service Time Out Interval (min)

The reduction of the minimum ESS No Service Time Out Interval from 3 to 2 minutes improves customer overall availability.

Valid entries	Usage
2 - **15**	No Service Time Out Interval in minutes.

PN Cold Reset Delay Timer (sec)

The **PN Cold Reset Delay Timer** field can be set in the range of **60** to **120** seconds. After you set a value, it is retained after an upgrade event.

Valid entries	Usage
60 to **120** secs	Time in seconds after which the PN cold reset occurs. Default is **60** seconds.

For more information on Improved Port Network Recovery from Control Network Outages, see *Avaya Aura™ Communication Manager Feature Description and Implementation,* 555-245-205.

System Parameters - SCCAN

Field descriptions for page 1

Figure 322: SCCAN-Related System Parameters screen

```
change system-parameters sccan                                 Page 1 of x
                      SCCAN - RELATED SYSTEM PARAMETERS

                MM(WSM) Route Pattern: ________
                          H1 Handover: ________
                          H2 Handover: ________
                         Announcement: ________
             Special Digit Conversion? ________
```

MM (WSM) Route Pattern

Enter a route pattern number that is SCCAN-enabled. Partition route pattern indexes, RHNPA indexes, deny, or nodes are not allowed.

Valid entries	Usage
blank	Default value. If this field is left blank, the feature is turned off. To enable this feature, you must enter an acceptable value. This is the default.
digits	Right-click on the field on the SAT screen to see valid entries for your system.

H1 Handover

Valid entries	Usage
unassigned extension	The primary handover number called to facilitate handover of a cellular call to the WAN or WLAN. Depending on whether the user is entering or exiting the Enterprise space, Communication Manager replaces the active call with the new call made using the hand-off H1 or H2 number.

H2 Handover

Valid entries	Usage
unassigned extension	A secondary handover number used when no acknowledgement is received from the H1 Handover number.

Announcement

Valid entries	Usage
assigned announcement extension	Enter the extension of the announcement you want to play during call handin or handout.

Special Digit Conversion

This field allows a user to call a cellular telephone number and get the same treatment as calling an extension that is running Communication Manager.

Valid entries	Usage
y	ARS checks the dialed string to determine if the dialed string is a SCCAN telephone number. If the number is a SCCAN telephone number, the cellular telephone number is replaced with the extension number that the cellular telephone is mapped to.
n	The feature is turned off. This is the default.

System Parameters - Security

See **Security-Related System Parameters**.

Telecommuting Access

This screen allows the System Administrator to administer the extension which allows remote users to use the feature.

Field descriptions for page 1

Figure 323: Telecommuting Access screen

```
add telecommuting-access
                                    TELECOMMUTING ACCESS

                       Telecommuting Access Extension: ________
```

Telecommuting Access Extension

This only allows remote access to the Telecommuting Access feature.

Valid entries	Usage
Unassigned extension of 1 to 13 digits, or blank	Enter an extension that conforms to your system's dial plan and is not assigned to any other system object.

Tenant

This screen defines tenants to the system. If your server running Communication Manager uses tenant partitioning, see Tenant Partitioning in *Avaya Aura™ Communication Manager Feature Description and Implementation,* 555-245-205, for more information.

Field descriptions for page 1

Figure 324: Tenant screen

```
change tenant n                                                   Page 1 of x
                           Tenant n

               Tenant Description: ________________________________________
                  Attendant Group: 1
            Ext Alert Port (TAAS): _______ Ext Alert (TAAS) Extension: ____
                Night Destination: _____
                     Music Source: 1
          Attendant Vectoring VDN:

DISTINCTIVE AUDIBLE ALERTING
             Internal: 1   External: 2   Priority: 3
                       Attendant Originated Calls: external

                        COS Group: 1
```

Attendant Group

This required information relates a tenant to an attendant group.

Note:
The default for the system is that all attendant groups exist. However, the attendant group will be empty if no consoles are assigned to it.

Valid entries	Usage
1 to **128**	See *Avaya Aura™ Communication Manager Hardware Description and Reference*, 555-245-207, for your system's range of allowable attendant group numbers.

Attendant Vectoring VDN

This field appears only if, on the System Parameters Customer-Options (Optional Features) screen, the **Attendant Vectoring** field is **y** and the **Tenant Partitioning** field is **n**. Enter the assigned Attendant VDN extension or blank. When set to **y**, the VDN and Call Vector screens display.

COS Group

This field appears when, on the System Parameters Customer-Options (Optional Features) screen, the **Tenant Partitioning** field is **y**. Use this field to assign this tenant to a Class of Service group.

Valid entries	Usage
1 to **100**	Enter the Class of Service group to which this tenant is assigned.

Ext Alert Port (TAAS)

Enter Trunk Answer Any Station (**TAAS**) alert port information, if any. The port type and the object type must be consistent, and the port can be assigned to only one tenant.

Valid entries	Usage
A valid port address or X **01** to **03** (DEFINITY CSI) or **1** to **64** (S87XX/S8300 Servers)	First and second characters are the cabinet number
A to **E**	Third character is the carrier
0 to **20**	Fourth and fifth character are the slot number
01 to **04** (Analog TIE trunks) **01** to **31**	Six and seventh characters are the circuit number

Valid entries	Usage
1 to **80** (DEFINITYCSI) or **1** to **250** (S87XX/S8300 Servers)	Gateway
V1 to **V9**	Module
01 to **31**	Circuit

Ext Alert (TAAS) Extension

This field appears only if you entered an **x** in the **Ext Alert Port (TAAS)** field. A system installer can then use the Terminal Translation Initialization (TTI) feature from a telephone plugged into any port to assign this extension number to that port. Doing so makes that port the external alert TAAS port.

Valid entries	Usage
A valid extension	Assign an extension as the external alert TAAS extension.

Music Source

Valid entries	Usage
1 to **20**	Enter the music/tone source for this partition. These sources are defined on the **Music Sources** screen.

Night Destination

Valid entries	Usage
A valid extension	Enter the night service station extension, if you want night service for this tenant.

Tenant

This is a display only field. It contains the tenant number that you entered on the command line.

Tenant Description

Valid entries	Usage
40 alpha-numeric characters or blank	You can leave the description field blank, but future administration will be easier if you provide descriptive information.

DISTINCTIVE AUDIBLE ALERTING

The following Distinctive Audible Alerting fields appear when **Tenant Partitioning** on the System Parameters Customer Options screen is **y**. Use these fields to administer distinctive ring patterns per tenant.

Attendant Originated Calls

This field appears when **Tenant Partitioning** on the System Parameters Customer Options screen is **y**.

Valid entries	Usage
internal **external** **priority**	Indicates which type of ringing (defined above) applies to attendant-originated calls. Default is **external**.

Distinctive Audible Alerting (Internal, External, Priority)

This field appears when **Tenant Partitioning** on the System Parameters Customer Options screen is **y**.

This is also known as Distinctive Ringing. Enter the number of rings for **Internal**, **External**, and **Priority** calls. For virtual stations, this applies to the mapped-to physical telephone. Defaults are as follows:

- **1**: Internal calls
- **2**:External and attendant calls
- **3**:Priority calls

Note:
SIP Enablement Services (SES) messaging includes the ring types internal, external, intercom, auto-callback, hold recall, transfer recall, or priority. In Communication Manager, types intercom, auto-callback, hold recall, and transfer recall are treated as priority.

Valid entries	Usage
1	1 burst, meaning one burst of ringing signal per period
2	2 bursts, meaning two bursts of ringing signal per period
3	3 bursts, meaning two bursts of ringing signal per period

Field descriptions for page 2

Figure 325: Tenant screen

```
change tenant n                                                Page 1 of x
                                   Tenant n


CALLING PERMISSION (Enter y to grant permission to call specified Tenant)


 1? y  11? n  21? n  31? n  41? n  51? n  61? n  71? n  81? n  91? n
 2? n  12? n  22? n  32? n  42? n  52? n  62? n  72? n  82? n  92? n
 3? n  13? n  23? n  33? n  43? n  53? n  63? n  73? n  83? n  93? n
 4? n  14? n  24? n  34? n  44? n  54? n  64? n  74? n  84? n  94? n
 5? n  15? n  25? n  35? n  45? n  55? n  65? n  75? n  85? n  95? n
 6? n  16? n  26? n  36? n  46? n  56? n  66? n  76? n  86? n  96? n
 7? n  17? n  27? n  37? n  47? n  57? n  67? n  77? n  87? n  97? n
 8? n  18? y  28? n  38? n  48? n  58? n  68? n  78? n  88? n  98? n
 9? n  19? n  29? n  39? n  49? n  59? n  69? n  79? n  89? n  99? n
10? n  20? n  30? n  40? n  50? n  60? n  70? n  80? n  90? n  100? n
```

Calling permissions

The system default allows each tenant to call only itself and Tenant 1. If you want to change that, you can do that on this screen.

Valid entries	Usage
y/n	Enter **y** to establish calling permission between the tenant number that you entered on the command line and any other tenant.

Tenant

This is a display only field. It contains the tenant number that you entered on the command line.

Terminal Parameters

This screen administers system-level parameters and audio levels for the 603 CALLMASTER telephones and the 4600-series, 6400-series, 8403, 8405B, 8405B+, 8405D, 8405D+, 8410B, 8410D, 8411B, 8411D, 8434D, and 2420/2410 telephones. Only authorized Avaya personnel can administer this screen.

Note:
With the Multinational Locations feature enabled, you can administer terminal parameters per location, rather than system-wide.

Field descriptions for page 1

Figure 326: 603/302 Terminal Parameters screen

```
change terminal-parameters                                     Page 1 of x
                      302/603/606-TYPE TERMINAL PARAMETERS

          Base Parameter Set: 1             Customize Parameters? _
     Note: Location-parameters forms assign terminal parameter sets.

   OPTIONS*
                   Display Mode: _*                DLI Voltage Level: _________*


   PRIMARY LEVELS*
            Voice Transmit (dB): _____*          Voice Sidetone (dB): _____*
             Voice Receive (dB): _____*     Touch Tone Sidetone (dB): _____*
       Touch Tone Transmit (dB): _____*
```

Figure 327: 6400/607A1/4600/2420 Type Terminal Parameters screen

```
change terminal-parameters                                     Page 2 of x

                  6400/607A1/4600/2420-TYPE TERMINAL PARAMETERS

          Base Parameter Set: 1         Customize Parameters? y
     Note: Location-parameters forms assign terminal parameter sets.
     Note: LEVELS do not apply to the 4600 terminals.*

    OPTIONS*
                    Display Mode:_*       Handset Expander Enabled?
           Volume for DCP Types:_*
            Volume for IP Types:_*
    PRIMARY LEVELS*
             Voice Transmit (dB):__*          Voice Sidetone (dB):__*
              Voice Receive (dB):__*     Touch Tone Sidetone (dB):__*
        Touch Tone Transmit (dB):__*


    BUILT-IN SPEAKER LEVELS*
             Voice Transmit (dB):__*           Voice Receive (dB):__*
                                         Touch Tone Sidetone (dB):__*
    6402 BUILT-IN SPEAKER LEVELS*
              Voice Receive (dB):__*     Touch Tone Sidetone (dB):__*
```

Figure 328: 8400-Series Terminal Parameters screen

```
change terminal-parameters                                   Page 3 of x
                           8400-TYPE TERMINAL PARAMETERS

          Base Parameter Set: __          Customize Parameters? _
     Note: Location-parameters forms assign terminal parameter sets.

OPTIONS*
                  Display Mode: _*               DLI Voltage Level: _________*

   PRIMARY LEVELS*
            Voice Transmit (dB): _____*          Voice Sidetone (dB): _____*
             Voice Receive (dB): _____*     Touch Tone Sidetone (dB): _____*
       Touch Tone Transmit (dB): _____*
   ADJUNCT LEVELS*+
            Voice Transmit (dB): _____*          Voice Receive (dB): _____*
            Voice Sidetone (dB): _____*    Touch Tone Sidetone (dB): _____*
   BUILT-IN SPEAKER LEVELS
            Voice Transmit (dB): _____*          Voice Receive (dB): _____*
                                           Touch Tone Sidetone (dB): _____*

   8403 BUILT-IN SPEAKER LEVELS
             Voice Receive (dB): _____*    Touch Tone Sidetone (dB): _____*
```

Base Parameter Set

Determines which default set of telephone options and levels are used. This field corresponds to the country codes. For the country code listing, see the Country code table on page 886.

Customize Parameters

Indicates whether the administrator wishes to change one or more of the default parameters.

Note:
Beginning with the May 2004 2.1 Release of Communication Manager, when the **Customize Parameters** field on the Terminal Parameters n screen is set to **y**, all Base Parameter Set default values display in the parameter fields. You must change the values in fields for which the default is not desired. To change just a few parameters back to default values, temporarily set the **Customize Parameters** field on the Terminal Parameters n screen to **n**, but do not submit the screen (do not press **Enter**). Make note of the default values for the specific fields you want to change, then set the Customize Parameters field back to **y**, and enter the default values in the fields.

Valid entries	Usage
y	If this field is **y** (yes), the **Option** and **Level** fields appear and all fields can be edited.
n	If this field is **n** (no), the system uses all default parameters associated with the **Base Parameter Set** field and all fields are display-only.

OPTIONS

Display Mode

Determines how the #) and ~ characters appear on the telephone's display.

Valid entries	Usage
1	If this field is set to **1**, the # and ~ do not change.
2	If this field is set to **2**, the telephone displays a # as a British pound sterling symbol and a ~ as a straight overbar.

DLI Voltage Level

Determines whether DCP Line Voltage used by the telephones is forced high, forced low, or allowed to automatically adjust.

Handset Expander Enabled

Determines whether the telephone reduces noise on the handset.

Valid entries	Usage
y	If the field is **y**, the telephone reduces background noise.

Primary levels

The system displays the default setting from the Base Parameter Set for all fields. Also, these fields all require the same input; valid entries are from **-44.0** db through **+14.0** db in 0.5 increments (for example, -44.0, -43.5, -43.0 and son on).

Volume for DCP Types

This field allows the DCP telephone volume to be adjusted while the call is in progress.

Valid entries	Usage
default speaker, handset unchangeable	The speaker resets to the default settings while the adjusted handset setting is retained.
default settings used to begin each call	No adjusted handset and speaker settings are retained.
retain handset and speaker between calls	The adjusted handset and speaker settings are retained.
retain speaker, handset unchangeable	Only the adjusted speaker setting is retained.

Volume for IP Types

This field allows the IP telephone volume to be adjusted while the call is in progress.

Note:
If you use this field, Avaya recommends that you not change any values in the **PRIMARY LEVELS** or **BUILT-IN SPEAKER LEVELS** areas.

Valid entries	Usage
default speaker, handset unchangeable	The speaker resets to the default settings while the adjusted handset setting is retained.
default settings used to begin each call	No adjusted handset and speaker settings are retained.
retain handset and speaker between calls	The adjusted handset and speaker settings are retained.
retain speaker, handset unchangeable	Only the adjusted speaker setting is retained.

PRIMARY LEVELS

Touch Tone Sidetone (dB)

Determines the touchtone volume fed back from the telephone when a users presses a button.

Touch Tone Transmit (dB) —

Determines the touchtone volume fed outbound from the telephone.

Voice Receive (dB)

Determines the volume of voice inbound to the telephone.

Voice Sidetone (dB)

Determines the volume of voice fed back from the handset voice microphone to the user's ear.

Voice Transmit (dB)

Determines the volume of voice outbound from the telephone.

Note:
You cannot administer all five of the **Primary Level** fields to +14.0 dB. If you attempt to submit the Terminal Parameters screen with all **Primary Levels** set to +14.0 dB, you receive an error message.

ADJUNCT LEVELS

Touch Tone Sidetone (dB)

Determines the touchtone volume fed back from the telephone when a users presses a button.

Voice Receive (dB)

Determines the volume of voice inbound to the adjunct.

Voice Sidetone (dB)

Determines the volume of voice fed back from the handset voice microphone to the user's ear.

Voice Transmit (dB)

Determines the volume of voice outbound from the adjunct.

Terminating Extension Group

This screen defines a Terminating Extension Group (TEG). Any telephone can be assigned as a TEG member; however, only a multi-appearance telephone can be assigned a **TEG** button with associated status lamp. The **TEG** button allows the telephone user to select a TEG call appearance for answering or for bridging onto an existing call.

The TEG members are assigned on an extension number basis. Call reception restrictions applicable to the group are specified by the group class of restriction (COR). The group COR takes precedence over an individual member's COR. When a TEG receives an incoming call, the TEG's primary Class of Restrictions (COR) should be considered for the calling restrictions, called restrictions, or both. The members could all be termination restricted but still receive calls if the group is not restricted.

The system allows for as many as 32 TEGs with up to 4 members each. An extension number can be assigned to more than one TEG but can have only one appearance of each group.

Field descriptions for page 1

Figure 329: Terminating Extension Group screen

```
change term-ext-group 1                                           Page 1 of x
                       TERMINATING EXTENSION GROUP

                      123456789012345678901234567                1234567890123
        Group Number: 1                          Group Extension: 40999
          Group Name: TERMINATING EXT. GROUP 1     Coverage Path: t77
       Security Code:                                        COR: 1
                                                              TN: 1
ISDN/SIP Caller Disp:                              LWC Reception: spe
          AUDIX Name:

GROUP MEMBER ASSIGNMENTS
    Ext             Name
    1234567890123   123456789012345678901234567
 1: 41153
 2: 41910           Station 41910 on ST2
 3: 41504           Gry Mrkt x41504 4a1803
 4: 41750           st2 4a1802
```

AUDIX Name

Name of the AUDIX machine as it appears in the IP Node Names screen.

Valid entries	Usage
Audix machine description	Unique identifiers for adjunct equipment.

COR

Valid entries	Usage
0 to **995**	Enter the class of restriction (COR) number that reflects the desired restrictions.

Coverage Path

Enter a number for the call coverage path for this group. A TEG cannot serve as a coverage point; however, calls to a TEG can redirect to coverage.

Valid entries	Usage
1 to **9999**	path number
t1 to **t999**	time of day table
blank	

Group Extension

Enter the extension of the terminating extension group.

Valid entries	Usage
0 to **9**	Unused extension number (cannot be a VDN extension). Do not leave blank.

Group Name

Enter the name used to identify the terminating extension group.

Group Number

A display-only field when the screen is accessed using an administration command such as `add` or `change`.

ISDN Caller Disp

This field is required if, on the System Parameters Customer-Options (Optional Features) screen, the **ISDN-PRI** or **ISDN-BRI Trunks** field is **y**.

Valid entries	Usage
grp-name	Specify whether the TEG group name or member name (member of TEG where call terminated) will be sent to the originating user.
mbr-name	Specify whether the TEG group name or member name (member of TEG where call terminated) will be sent to the originating user.
blank	If the **ISDN-PRI** or **ISDN-BRI Trunks** field is **n**, leave blank.

LWC Reception

Defines the source for Leave Word Calling (LWC) messages.

Valid entries	Usage
audix	If LWC is attempted, the messages are stored in AUDIX.
spe	If LWC is attempted, the messages are stored in the system processing element (spe).
none	If LWC is attempted, the messages are not stored.

Security Code

Valid entries	Usage
3 to 8 digit security code	This code is used for the Demand Print feature.

TN

Valid entries	Usage
1 to **100**	Enter the Tenant Partition number.

GROUP MEMBER ASSIGNMENTS

Ext

Enter the extension number (cannot be a VDN extension) assigned to a station.

Valid entries	Usage
0 to **9**	A valid extension number of 1 to 13 digits.

Name

This display-only field shows the name assigned to the preceding extension number when the TEG member's telephone is administered.

TFTP Server

The Trivial File Transfer Protocol screen allows specification of the TFTP server that Communication Manager uses to get download files.

Field descriptions for page 1

Figure 330: TFTP Server Configuration screen

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

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

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

Filename in Memory

A display-only field showing the name of the file currently in Communication Manager memory.

File Size

A display-only field showing the number of bytes transferred.

File Status

A display-only field showing Download In Progress, Download Failed, File Not Found, or Download Completed.

File to Retrieve

Valid entries	Usage
up to 32 alpha-numeric, case sensitive, characters	Enter the name of the file you are going to retrieve using up to 32 alpha-numeric, case sensitive, characters for identification.

Local Node Name

The local node name must be a valid entry from the IP Node Names screen. The node must be assigned to a CLAN IP interface or **procr** (processor CLAN).

Valid entries	Usage
1 to **15** characters	Valid entry from the IP Node Names screen.
procr	Processor CLAN for S8300/S87XX Servers

TFTP Server Node Name

Valid entries	Usage
1 to 15 characters	The TFTP server node name must be a valid entry from the IP Node Names screen.

TFTP Server Port

Valid entries	Usage
1 to **64500**	Enter a number for the remote TCP port.

Time of Day Coverage Table

This screen allows up to five different coverage paths, associated with five different time ranges, for each day of the week. Only one coverage path can be in effect at any one time.

Field descriptions for page 1

Figure 331: Time of Day Coverage Table screen

```
change coverage time-of-day n
                   TIME OF DAY COVERAGE TABLE n___

           Act   CVG      Act   CVG      Act   CVG      Act   CVG      Act   CVG
           Time  PATH     Time  PATH     Time  PATH     Time  PATH     Time  PATH

  Sun      00:00 ____     __:__ ____     __:__ ____     __:__ ____     __:__ ____
  Mon      00:00 ____     __:__ ____     __:__ ____     __:__ ____     __:__ ____
  Tue      00:00 ____     __:__ ____     __:__ ____     __:__ ____     __:__ ____
  Wed      00:00 ____     __:__ ____     __:__ ____     __:__ ____     __:__ ____
  Thu      00:00 ____     __:__ ____     __:__ ____     __:__ ____     __:__ ____
  Fri      00:00 ____     __:__ ____     __:__ ____     __:__ ____     __:__ ____
  Sat      00:00 ____     __:__ ____     __:__ ____     __:__ ____     __:__ ____
```

Act Time

Specify the activation time of the coverage path administered in the next **CVG PATH** field. Enter the information in 24-hour time format.

Valid entries	Usage
00:01 to **23:59**	If there are time gaps in the table, there will be no coverage path in effect during those periods. The first activation time for a day is set to 00:00 and cannot be changed. Activation times for a day must be in ascending order from left to right.

CVG Path

Enter the coverage path number.

Valid entries	Usage
1 to **9999** or blank	For the S87XX Series IP-PNC

Time of Day Coverage Table

A display-only field when the screen is accessed using an administration command. Specifies the Time of Day Coverage Table number. Up to 999 can be administered.

Time of Day Routing Plan

Use this screen to set up Time of Day Routing Plans. You can route AAR and ARS calls based on the time of day each call is made. You can design up to 8 Time of Day Routing Plans, each scheduled to change up to 6 times a day for each day in the week.

Match the Time of Day Routing Plan PGN# with the **PGN#** field on the Partition Routing Table for the route pattern you want to use.

Note:
Automatic Route Selection (ARS) or **Private Networking**, **AAR/ARS Partitioning**, and **Time of Day Routing** must be enabled on the System Parameters Customer-Options (Optional Features) screen before you can use Time of Day Routing.

Field descriptions for page 1

Figure 332: Time Of Day Routing Plan screen

```
change time-of-day
                               TIME OF DAY ROUTING PLAN ___          Page 1 of x
      Act      PGN   Act     PGN   Act     PGN   Act     PGN   Act     PGN   Act     PGN
      Time      #    Time     #    Time     #    Time     #    Time     #    Time     #
Sun   00:00     1    __:__    _    __:__    _    __:__    _    __:__    _    __:__    _
Mon   00:00     1    __:__    _    __:__    _    __:__    _    __:__    _    __:__    _
Tue   00:00     1    __:__    _    __:__    _    __:__    _    __:__    _    __:__    _
Wed   00:00     1    __:__    _    __:__    _    __:__    _    __:__    _    __:__    _
Thu   00:00     1    __:__    _    __:__    _    __:__    _    __:__    _    __:__    _
Fri   00:00     1    __:__    _    __:__    _    __:__    _    __:__    _    __:__    _
Sat   00:00     1    __:__    _    __:__    _    __:__    _    __:__    _    __:__    _
```

Act Time

Specifies the time of day the route pattern (identified by PGN) begins.

Valid entries	Usage
00:00 to **23:59**	Time is represented using a 24 hour clock. List times for the same day in increasing order. There must be at least one entry for each day.

PGN

Identifies the route pattern for activation time listed.

Valid entries	Usage
1 to **8**	Enter a PGN that matches the PGN and route pattern on the Partition Routing Table. There must be at least one entry for each day. When SA9050 is active, all PGN fields are increased from **1** - **8** to **1** - **32**.

Time of Day Routing Plan

Displays the Time of Day Routing Plan number (1 through 8).

Time of Day Station Lock Table

Use this screen to lock stations automatically by a time of day schedule.

Field descriptions for page 1

Figure 333: Time of Day Coverage Table screen

```
change time-of-day station-lock table 1                     page 1 of x
                   TIME OF DAY Station Lock Table 1

     Table Active? y                   Manual Unlock allowed? y


       INTERVAL 1          INTERVAL 2                INTERVAL 3
       Begin End           Begin End                 Begin End
       Time  Time          Time  Time                Time  Time
  Sun  00:00 00:00         00:00 00:00               00:00 00:00
  Mon  00:00 00:00         00:00 00:00               00:00 00:00
  Tue  00:00 00:00         00:00 00:00               00:00 00:00
  Wed  00:00 00:00         00:00 00:00               00:00 00:00
  Thu  00:00 00:00         00:00 00:00               00:00 00:00
  Fri  00:00 00:00         00:00 00:00               00:00 00:00
  Sat  00:00 00:00         00:00 00:00               00:00 00:00
```

Interval (1, 2, 3)

Use these fields to indicate the TOD Station Lock Interval. There are seven rows of entries for 7 days of the week, each row starting with a fixed day entry. The first row starts with Sunday (Sun). The administration will impose validation of overlapping intervals or invalid blank entries.

Valid entries	Usage
0 to **23** or blank for hours and **0** to **59** or blank for minutes	Enter the desired TOD Station Lock Intervals.

Manual unlock allowed

Use this field to indicate if the TOD Station Lock Interval can be deactivated by the manual Station Lock sequence.

Valid entries	Usage
y/n	When set to **y**, the user can manually unlock the TOD-locked station using either a **sta-lock** button or a Feature Access Code followed by an SSC. When set to **n**, the user cannot unlock the station. Default is **n**.

Table Active

Use this field to indicate if this Time-Of-Day-Lock Table is activated or deactivated. Enter n to turn off TOD Station lock for all stations associated to this table. Valid entries are **y**(es), **n**(o). Default is **n**.

Valid entries	Usage
y	Enter **y** to turn on TOD Station lock for all stations associated to this table.
n	Enter **n** to turn off TOD Station lock for all stations associated to this table. Default is **n**.

Toll Analysis

Note:
The **Toll List** field on this screen does not interact with or relate to the ARS Toll Table.

This screen associates dialed strings to the system's Restricted Call List (RCL), Unrestricted Call List (UCL), and Toll List. You can force users to dial an account code if you associate dialed strings with CDR Forced Entry of Account Codes.

To maximize system security, Avaya recommends that toll calling areas be restricted as much as possible through the use of the **RCL (Restricted Call List)** and **Toll List** fields on this screen.

Field descriptions for page 1

Figure 334: Toll Analysis screen

```
change toll n                                                  Page 1 of x
                                TOLL ANALYSIS                 Percent Full:_
                               Location:
                         Total        Toll  CDR  <--Unrestricted Call List-->
    Dialed String      Min  Max  RCL  List  FEAC  1  2  3  4  5  6  7  8  9 10

_____________________  ___  ___   _     _     _   _  _  _  _  _  _  _  _  _  _
_____________________  ___  ___   _     _     _   _  _  _  _  _  _  _  _  _  _
_____________________  ___  ___   _     _     _   _  _  _  _  _  _  _  _  _  _
_____________________  ___  ___   _     _     _   _  _  _  _  _  _  _  _  _  _
_____________________  ___  ___   _     _     _   _  _  _  _  _  _  _  _  _  _
_____________________  ___  ___   _     _     _   _  _  _  _  _  _  _  _  _  _
_____________________  ___  ___   _     _     _   _  _  _  _  _  _  _  _  _  _
_____________________  ___  ___   _     _     _   _  _  _  _  _  _  _  _  _  _
_____________________  ___  ___   _     _     _   _  _  _  _  _  _  _  _  _  _
_____________________  ___  ___   _     _     _   _  _  _  _  _  _  _  _  _  _
_____________________  ___  ___   _     _     _   _  _  _  _  _  _  _  _  _  _
_____________________  ___  ___   _     _     _   _  _  _  _  _  _  _  _  _  _
```

CDR FEAC

Valid entries	Usage
x	Enter **x** to require an account code from a call whose facility COR requires a Forced Entry of Account Code.

Dialed String

Valid entries	Usage
digits **0** to **9** (up to 18 characters)	Enter the dialed string you want Communication Manager to analyze.
***, x, X**	wildcard characters

Location

Display-only field.

.

Valid entries	Usage
1 to **64**	Defines the server's location for this Toll Analysis Table. On the System Parameters Customer-Options (Optional Features) screen, the **ARS** field and the **Multiple Locations** field must be set to **y** for values other than **all** to appear.
all	Indicates that this Toll Analysis Table is the default for all port network (cabinet) locations.

Max

Valid entries	Usage
Min to **28**	Enter the maximum number of user-dialed digits the system collects to match to this dialed string.

Min

Valid entries	Usage
1 to **Max**	Enter the minimum number of user-dialed digits the system collects to match to this dialed string.

Percent Full

Display-only field showing the percentage (**0** to **100**) of the system's memory resources that have been used by AAR/ARS. If the figure is close to 100%, you can free-up memory resources.

RCL

Enter **x** to assign the Dialed String to the Restricted Call List (RCL).

Valid entries	Usage
x	All entries of **x** and their associated dialed strings are referred to as the System's Restricted Call List. The RCL can be assigned to any COR. A call attempt from a facility whose COR is marked as being associated with the RCL and whose dialed string matches a RCL dialed string field will be denied. The caller receives intercept treatment.

Toll List

Valid entries	Usage
x	Enter **x** to assign the Dialed String to the Toll List.

Dialed String	Min	Max	Toll List
0	1	23	x
1	4	23	x
20	10	10	x
21	10	10	x
30	10	10	x
31	10	10	x
40	10	10	x
41	10	10	x
50	10	10	x
51	10	10	x
60	10	10	x
61	10	10	x
70	10	10	x
71	10	10	x
			1 of 2

Dialed String	Min	Max	Toll List
80	10	10	x
81	10	10	x
90	10	10	x
91	10	10	x
			2 of 2

Unrestricted Call List

Valid entries	Usage
x	Enter **x** to assign the dialed string to one of the system's Unrestricted Call Lists (UCL).

Tone Generation

The Tone Generation screen allows you to administer the tone characteristics that parties on a call hear under various circumstances.

Note:
With the Multinational Locations feature enabled, tone generation can be administered per location, rather than system-wide.

Field descriptions for page 1

Figure 335: Tone Generation screen

```
change tone-generation 2                                          Page 1 of  X
                             TONE GENERATION 2

                                                 Base Tone Generator Set: 1
          440Hz PBX-dial Tone? n               440Hz Secondary-dial Tone? n
```

440Hz PBX-dial Tone

Specifies whether the switch (primary) dial tone will be changed to a continuous 440Hz/-17 tone.

Valid entries	Usage
y/n	A value of **n** implies the tone will either be administered on a later page of this screen or, if no individual definition is administered, as defined in **Base Tone Generation Set**.

440Hz Secondary-dial Tone

Specifies whether the Secondary (CO) dial tone will be changed to a continuous 440Hz/-17 tone.

Valid entries	Usage
y/n	A value of **n** implies the tone will either be administered on a later page of this screen or, if no individual definition is administered, as defined in **Base Tone Generation Set**.

Base Tone Generator Set

The country code identifies the base tone generation set to be used. For information on the appropriate tone-generation hardware to use in a specific country, see *Avaya Aura™ Communication Manager Hardware Description and Reference,* 555-245-207.

Valid entries	Usage
1 to **25**	See the Country code table at the beginning of the System-Parameters Country-Code screen description.

Field descriptions for page 2

Figure 336: Tone Generation screen

```
change tone-generation                                          Page   2 of  X
                       TONE GENERATION CUSTOMIZED TONES

        Tone Name        Cadence        Tone
                          Step    (Frequency/Level)
         Hold               1:     480/-17.25           Duration(msec): 1000
                            2:     goto                           Step:
                            3:
                            4:
                            5:
                            6:
                            7:
                            8:
                            9:
                           10:
                           11:
                           12:
                           13:
                           14:
                           15:
```

Cadence Step

Display-only fields that identify the number of each tone cadence step.

Valid entries	Usage
1 to **15**	Identifies the number of each tone cadence step.

Duration (msec)

Valid entries	Usage
50 to **12750**, in increments of 50	Enter the duration of this step in the tone sequence.

Step

This field appears when you enter **goto** in the **Tone/Frequency Level** field.

Valid entries	Usage
Cadence step	.Enter the number of the cadence step for this **goto** command.

Tone (Frequency/Level)

Valid entries	Usage
silence	An entry of **silence** means no tone. A final step of **silence** with an infinite duration is added internally to any tone sequence that does not end in a **goto**.
	1 of 2

Valid entries	Usage
goto	An entry of **goto** means to repeat all or part of the sequence, beginning at the specified cadence step.
350/-17.25 **350+425/-4.0** **350+440/-13.75** **375+425/-15.0** **404/-11.0** **404/-16.0** **404+425/-11.0** **404+450/-11.0** **425/-4.0** **425/-11.0** **425/-17.25** **440/-17.25** **440+480/-19.0** **450/-10** **480/-17.25** **480+620/-24.0** **525/-11.0** **620/-17.25** **697/-8.5** **770/-8.5** **852/-8.5** **941/-8.5** **1000/0.0** **1000/+3.0** **1004/0.0** **1004/-16.0** **1209/-7.5** **1336/-7.5** **1400/-11.0** **1477/-7.5** **1633/-7.5** **2025/-12.1** **2100/-12.1** **2225/-12.1** **2804/-16.0**	Specifies the frequency and level of the tone.
	2 of 2

Tone Name

Each entry uses one of the keywords below to indicate which of the individually administrable tones this screen modifies.

Valid entries	Usage
blank	If this field is blank, all entries are ignored in the corresponding **Tone (Frequency/Level)** field.
1-call-wait **2-call-wait** **3-call-wait** **busy** **busy-verify** **call-wait-ringback** **conference** **confirmation** **disable-dial** **hold** **hold-recall** **immed-ringback** **intercept** **intrusion** **mntr/rec-warning** **PBX-dial** **recall-dial** **recall-dont-ans** **redirect** **reorder** **rep-confirmation** **reset-shift** **ringback** **secondary-dial** **special-dial** **whisper-page** **zip**	

Note:
For information on setting the Caller Response Interval before a call goes to coverage (when the value for this field is **redirect**), see Caller Response Interval in the Call Coverage section of *Avaya Aura™ Communication Manager Feature Description and Implementation*, 555-245-205.

Trunk Group

Use the Trunk Group screen to set basic characteristics for every type of trunk group and to assign ports to the group. This section lists and describes all the fields you might see on the screen. Many fields are dependent on the settings of other fields and only appear when certain values are entered in other fields on the screen. For example, the entry in the **Group Type** field might significantly change the content and appearance of the Trunk Group screen.

For more information on administering trunk groups, see *Administering Avaya Aura™ Communication Manager*, 03-300509.

Note:
This section does not cover ISDN trunks. For descriptions of the screens and fields that are unique to ISDN trunks, see ISDN Trunk Group on page 539.

Field descriptions for page 1

The figure below is only an example, and is intended to show most of the fields that might appear on page 1 of the Trunk Group screen. This example might not show all fields, or might show fields that normally do not appear together; it is not intended to reflect a specific trunk group type. Your own screen might vary from this example according to specific field and system settings. The list of field descriptions that follows the figure is in alphabetical order for quick reference. This list is intended to be comprehensive, and might include information on fields that are not shown in the example. The field descriptions identify fields that are specific to particular trunk group types.

Figure 337: Trunk Group screen - page 1

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

Group Number: 8                    Group Type: co CDR Reports: y
  Group Name: OUTSIDE CALL                COR: 1 TN: 1 TAC:
   Direction: two-way       Outgoing Display? n
 Dial Access? n            Busy Threshold: 255   Night Service: 1234567890123
Queue Length: 0          Country: 1       Incoming Destination: 1234567890123
   Comm Type: voice                 Auth Code? n    Digit Absorption List:
    Prefix-1? y                   Trunk Flash? n          Toll Restricted? y

         Trunk Type:
```

Analog Gain

Use this field to reduce the strength of incoming signals on TN2199 ports if users regularly experience echo, distortion, or unpleasantly loud volume. Experiment to find the best setting for your situation. This field appears if the **Country** field is **15** and the **Trunk Type (in/out)** field is **2-wire-ac**, **2-wire-dc**, or **3-wire**.

Valid entries	Usage
a	Reduces the incoming signal by -3dB.
b	Reduces the incoming signal by -6dB.
c	Reduces the incoming signal by -8dB.
none	No reduction. Don’t change this setting unless the trunk group’s sound quality is unacceptable.

Auth Code

The **Auth Code** field is available only for incoming or two-way trunk groups and if you enable the **Authorization Codes** feature. If you enable the **Auth Code** field, Communication Manager performs an auth code check for the incoming trunk call that is routed over another trunk. For more infomation on the **Authorization Codes** feature, see Authorization Code - PIN Checking for Private Calls on page 76.

Note: The **Auth Code** field is unavailable if:

- the **Group Type** field is **tandem**
- the **Group Type** field is **ISDN** and the **Service Type** field is **tandem**.

In these situations, permissions of the caller are transmitted using Traveling Class Mark.

The following table explains the situations when the caller must enter an auth code.

Calling party	Called party	Facility Restriction Level (FRL) check?	Auth Code, required if
Station	Station	No	No
	Trunk	Yes, if the trunk was accessed by Route Pattern and not by Trunk Access Code	the FRL of the calling station is less than the FRL of the outgoing route pattern

Calling party	Called party	Facility Restriction Level (FRL) check?	Auth Code, required if
Incoming Trunk	Station	No	the Auth Code field is enabled on the incoming trunk group
	Trunk	Yes, if the trunk was accessed by Route Pattern and not by Trunk Access Code	the FRL of the incoming trunk is less than the FRL of the outgoing route pattern or the Auth Code field is enabled on the incoming trunk group

BCC

Generalized Route Selection uses the BCC to select the appropriate facilities for routing voice and data calls. Far-end tandem servers/switches also use the BCC to select outgoing routing facilities with equivalent BCC classes.The entry in the **Bearer Capability Class** field is used to select the appropriate facilities for incoming ISDN calls. Communication Manager compares the entry in the **BCC** field to the value of the Bearer Capability information element for the incoming call and routes the call over appropriate facilities. For example, a call with BCC 4 will only be connected through facilities that support 64 kbps data transmission.

The **Bearer Capability Class** field appears when all of the following are true:

- Either the **ISDN-BRI Trunks** field or the **ISDN-PRI** field on the System Parameters Customer-Options **(Optional Features)** screen is **y**.
- The **Group Type** field is **access**, **co**, **fx**, **tandem**, **tie**, or **wats**.
- The **Comm Type** field is **data**, **avd**, or **rbavd**.

Valid entries	Usage
0	For voice and voice-grade data
1	For 56 kbps synchronous data transmitted with robbed-bit signaling
2	Less than 19.2 kbps synchronous or asynchronous data
4	For 64 kbps data on unrestricted channels

Busy Threshold

Use this field if you want attendants to control access to outgoing and two-way trunk groups during periods of high use. When the threshold is reached and the warning lamp for that trunk group lights, the attendant can activate trunk group control: internal callers who dial out using a trunk access code will be connected to the attendant, and the attendant can prioritize outgoing calls for the last remaining trunks. Calls handled by AAR and ARS route patterns go out normally.

Valid entries	Usage
1 to **255** (S87XX Series IP-PNC)	Enter the number of trunks that must be busy in order to light the warning lamp on the Attendant Console. For example, if there are 30 trunks in the group and you want to alert the attendant whenever 25 or more are in use, enter **25**.

CDR Reports

Valid entries	Usage
y	All outgoing calls on this trunk group will generate call detail records. If the **Record Outgoing Calls Only** field on the CDR System Parameters screen is **n**, then incoming calls on this trunk group will also generate call detail records.
n	Calls over this trunk group will not generate call detail records.
r (ring-intvl)	CDR records will be generated for both incoming and outgoing calls. In addition, the following ringing interval CDR records are generated: ● Abandoned calls: The system creates a record with a condition code of "H," indicating the time until the call was abandoned. ● Answered calls: The system creates a record with a condition code of "G," indicating the interval from start of ring to answer. ● Calls to busy stations: The system creates a record with a condition code of "I," indicating a recorded interval of 0.

Note:
For ISDN trunk groups, the **Charge Advice** field affects CDR information. For CO, DIOD, FX, and WATS trunk groups, the **PPM** field affects CDR information.

CESID I Digits Sent

This field appears when **Group Type** is **cama**. For emergency 911 service, Communication Manager might send Caller's Emergency Service Identification (CESID) information to the central office or E911 tandem server/switch. This digit string is part of the E911 signaling protocol.

Valid entries	Usage
1 to 3 digits	Determine the correct entry for this field by talking to your E911 provider.

Comm Type

Use this field to define whether the trunk group carries voice, data, or both.

Note:
Comm Types of **avd**, **rbavd** and **data** require trunk member ports on a DS1 circuit pack.

Valid entries	Usage
avd	Enter **avd** for applications that mix voice and Digital Communication Protocol data, such as video conferencing applications. The receiving end server discriminates voice calls from data calls and directs each to an appropriate endpoint. Neither originating nor terminating ends insert a modem pool for any calls when **Comm Type** is **avd**. The **Signaling Mode** field on the DS1 Circuit Pack screen must be set for either **common-chan** or **CAS** signaling.
data	Enter **data** only when all calls across the trunk group originate and terminate at Communication Manager digital data endpoints. Public networks don't support **data**: supported by Avaya's DCP protocol, this entry is used almost exclusively for the data trunk group supporting DCS signaling channels. The **Signaling Mode** field on the DS1 Circuit Pack screen might be set to **robbed-bit** or **common-chan**.
rbavd	For digital trunk groups that carry voice and data with robbed-bit signaling. The **Signaling Mode** field on the DS1 Circuit Pack screen must be set to **robbed-bit** unless mixed mode signaling is allowed on the DS1 circuit pack. In that case, the **Signaling Mode** field might be **isdn-ext** or **isdn-pri**.
voice	For trunk groups that carry only voice traffic and voice-grade data (that is, data transmitted by modem). Analog trunk groups must use **voice**.

COR

Decisions regarding the use of Class of Restriction (COR) and Facility Restriction Levels (FRLs) should be made with an understanding of their implications for allowing or denying calls when AAR/ARS/WCR route patterns are accessed. For details on using COR and FRLs, see *Avaya Toll Fraud and Security Handbook*, 555-025-600.

Valid entries	Usage
0 to **995**	Enter a class of restriction (COR). Classes of restriction control access to trunk groups, including trunk-to-trunk transfers.

Tip:
Remember that facility restriction levels (FRL) are assigned to classes of restriction. Even if 2 trunk groups have classes of restriction that allow a connection, different facility restriction levels might prevent operations such as off-net call forwarding or outgoing calls by remote access users.

CO Type

This field appears when the **Country** field is **14** and is used only by trunk group members administered on a TN464D vintage 2 or later DS1 circuit pack.

Valid entries	Usage
analog	This field specifies whether the trunk group is connected to analog or digital facilities at the central office.
digital	

Country

This field is administered at installation and sets numerous parameters to appropriate values for the public network in which the server running Communication Manager operates. For example, the value of this field, with the values of the **Trunk Termination** and the **Trunk Gain** fields, determines the input and trans-hybrid balance impedance requirements for ports on TN465B, TN2146, and TN2147 circuit packs.

This field appears for the trunk groups that connect Communication Manager to a central office in the public network — CO, DID, DIOD, FX, and WATS trunk groups.

CAUTION:
Do not change this field. If you have questions, contact your Avaya technical support representative.

Note:
For DID trunk types, country code **19** is not accepted in the Trunk Group screen in Communication Manager. This will be supported at a later date.

Valid entries	Usage
1 to **25**	Set at installation. For a list of country codes, see the Country code table on page 886.
11	If the **Country** field is **11**, Communication Manager is administered for Public Network Call Priority (Call Retention and Re-ring).
14	If the **Country** field is **14** and the **Group Type** is **DID** or **DIOD**, the **CO Type** field appears.
15	If the **Country** field is **15**, Communication Manager is administered for Public Network Call Priority (Intrusion and Re-ring). Also, the **Protocol Type** field appears for **Group Type DID** or **DIOD**.
18	If the **Country** field is **18**, Communication Manager can be administered for Public Network Call Priority (Mode of Release Control, Forced Disconnect, and Re-ring).
23	If the **Country** field is **23** and **Group Type** field is either **CO** or **DID**, Communication Manager is administered for Block Collect Calls.

Dial Access

This field controls whether users can route outgoing calls through an outgoing or two-way trunk group by dialing its trunk access code. Allowing dial access does not interfere with the operation of AAR/ARS.

SECURITY ALERT:
Calls dialed with a trunk access code over WATS trunks bypass AAR/ARS and aren't restricted by facility restriction levels. For security, you might want to leave the field set to n unless you need dial access to test the trunk group.

Valid entries	Usage
y	Allows users to access the trunk group by dialing its access code.
n	Does not allow users to access the trunk group by dialing its access code. Attendants can still select this trunk group with a **Trunk Group Select** button.

Digit Absorption List

This field assigns a digit absorption list, when used, to a trunk group that terminates at a step-by-step central office.

Valid entries	Usage
0 to **4** or blank	Enter the number of the digit absorption list this trunk group should use.

Note:
In a DCS network, DCS features that use the **remote-tgs** button (on telephones at a remote end) do not work when the incoming trunk group at your end deletes or inserts digits on incoming calls. The **remote-tgs** button on a remote server/switch, for example, tries to dial a TAC on your switch. If your end adds or deletes digits, it defeats this operation. If you need to manipulate digits in a DCS network (for example, to insert an AAR feature access code), do it on the outgoing side based on the routing pattern.

Direction

Enter the direction of the traffic on this trunk group. The entry in this field affects which timers appear on the Administrable Timers page. This field appears for all trunk groups except DID and CPE.

Valid entries	Usage
incoming	Traffic on this trunk group is incoming.
outgoing	Traffic on this trunk group is outgoing.
two-way	Enter two-way for Network Call Redirection.

Group Name

Valid entries	Usage
1 to 27 characters	Enter a unique name that provides information about the trunk group. Do not use the default entry or the group type (DID, WATS) here. For example, you might use names that identify the vendor and function of the trunk group: Qwest Local; Sprint Toll, etc. **Note:** For 4610SW, 4620SW, 4621SW, and 4622SW, Sage, Spark, and 9600-series Spice telephones, the **Group Name** field has an associated optional native name field that is supported by the Unicode language display. The native name field is accessible through the Integrated Management Edit Tools such as Avaya Site Administration (ASA). Unicode is also an option for the 2420J telephone when **Display Character Set** on the **System Parameters Country-Options** screen is **katakana**. For more information on the 2420J, see *2420 Digital Telephone User's Guide*, 555-250-701. Avaya BRI stations support ASCII characters only. They do not support non-ASCII characters, such as Eurofont or Kanafont. Therefore, if you use non-ASCII characters in any Communication Manager **Name** field, such characters do not display correctly on a BRI station.

Group Number

This field displays the group number assigned when the trunk group was added.

Note:
For trunk groups connecting two Avaya S8XXX Servers in Distributed Communication System networks, Avaya recommends that you assign the same group number on both servers.

Group Type

Enter the type of trunk group. The fields that are displayed and available might change according to the trunk group type selected. Busy-out the trunk group before you change the group type. Release the trunk group after you make the change. For more information about busying out and releasing trunk groups, see your system's maintenance documentation.

For more information about ISDN trunk group screens, see ISDN Trunk Group.

Valid entries	Usage
Access	Use access trunks to connect satellite servers to the main switch in Electronic Tandem Networks (ETN). Access trunks do not carry traveling class marks (TCM) and thus allow satellite callers unrestricted access to out-dial trunks on the main server. Allows Inband ANI.
APLT	Advanced Private Line Termination (APLT) trunks are used in private networks. APLT trunks allow inband ANI.
CAMA	CAMA trunks route emergency calls to the local community's Enhanced 911 systems.
CO	CO trunks typically connect Communication Manager to the local central office, but they can also connect adjuncts such as external paging systems and data modules.
CPE	Use CPE trunks to connect adjuncts, such as paging systems and announcement or music sources, to the server running Communication Manager.
DID	Use DID trunks when you want people calling your organization to dial individual users directly without going through an attendant or some other central point. Allows Inband ANI.
DIOD	DIOD trunks are two-way trunks that transmit dialed digits in both directions. In North America, use tie trunks for applications that require two-way transmission of dialed digits. Allows Inband ANI.
DMI-BOS	Digital Multiplexed Interface - Bit-Oriented Signaling (DMI-BOS) trunks allow communication with systems using DMI-BOS protocol. DMI-BOS trunks allow inband ANI.
FX	An FX trunk is essentially a CO trunk that connects your server running Communication Manager directly to a central office outside your local exchange area. Use FX trunks to reduce long distance charges if your organization averages a high volume of long-distance calls to a specific area code.
ISDN	Use ISDN trunks when you need digital trunks that can integrate voice, data, and video signals and provide the bandwidth needed for applications such as high-speed data transfer and video conferencing. ISDN trunks can also efficiently combine multiple services on one trunk group. Use **ISDN** for Network Call Transfer. **Note:** You cannot enter **ISDN** unless the **ISDN-PRI** field, the **ISDN-BRI Trunks** field, or both have been enabled on the System Parameters Customer-Options (Optional Features) screen.
	1 of 2

Valid entries	Usage
RLT	Release Link trunks work with Centralized Attendant Service in a private network.
SIP	Use **SIP** trunks to connect a server running Communication Manager to a SIP Enablement Services (SES) home server, or to connect two Communication Manager servers. **Note:** The Automatic CallBack, Priority Calling, and Whisper Page features currently do not work correctly if each of the call's parties is using a SIP endpoint administered on and managed by a different instance of Communication Manager.
Tandem	Tandem trunks connect tandem nodes in a private network. Tandem trunks allow inband ANI.
Tie	Use tie trunks to connect a server running Communication Manager to a central office or to another server or switch in a private network. Tie trunks transmit dialed digits with both outgoing and incoming calls, and allow inband ANI.
WATS	Use WATS trunks to reduce long-distance bills when your organization regularly places many calls to a specific geographical area in North America. Outgoing WATS service allows calls to certain areas ("WATS bands") for a flat monthly charge. Incoming WATS trunks allow you to offer toll-free calling to customers and employees.
	2 of 2

Incoming Destination

Use this field to set the destination for all incoming calls on trunk groups such as CO, FX, and WATS that must terminate at a single destination. The destination you enter here is also the default night service destination unless you enter a different destination in the **Night Service** field. Appears when the **Direction** field is **incoming** or **two-way**.

Valid entries	Usage
blank	Leave this field blank if the **Trunk Type** (in/out) field is not **auto/….**
valid extension number	Calls go to the extension you enter. You can enter any type of extension, though typically the extension entered here identifies a VDN, a voice response unit, or a voice messaging system. Night service overrides this setting when it is active. **Note:** If entering a Multi-Location Dial Plan shortened extension, note the following: When entering a Multi-Location Dial Plan shortened extension in a field designed for announcement extensions, certain administration end validations that are normally performed on announcement extensions are not done, and resultant warnings or submittal denials do not occur. The shortened extensions also do not appear in any display or list that shows announcement extensions. Extra care should be taken to administer the correct type of announcement for the application when assigning shortened extensions.
attd	Calls go to the attendant and are recorded as Listed Directory Number (LDN) calls on call detail records.

ITC

The Generalized Route Selection feature, part of the automatic routing technology used in Communication Manager, compares the line coding of available digital facilities and selects appropriate routes for voice and data calls. The **Information Transfer Capability** field appears when the **Comm Type** field is **data**, **avd**, or **rbavd** and the **BCC** field is not **0**.

Valid entries	Usage
rest(ricted)	Restricted trunks use **ami-basic** or **ami-zcs** line coding and can carry only restricted calls.
unre(stricted)	Unrestricted trunks use **b8zs**, **hdb3,** or **cmi** line coding and can carry restricted or unrestricted calls. A trunk group with an unrestricted ITC can have only unrestricted trunks as members.

Night Service

This field sets the destination to which incoming calls go when **Night Service** is in operation. If a **Night** field on the Group Member Assignments page is administered with a different destination, that entry will override the group destination for that trunk. CPE, DID, and DIOD trunk groups do not support night service.

Tip:
Whenever possible, use a night service destination on your switch: otherwise some features won't work correctly, even over a DCS network.

Valid entries	Usage
blank	Leave this field blank if the **Trunk Type (in/out)** field is not auto/….
An extension number (can be a VDN)	Enter the extension of your night service destination. **Note:** If entering a Multi-Location Dial Plan shortened extension, note the following: When entering a Multi-Location Dial Plan shortened extension in a field designed for announcement extensions, certain administration end validations that are normally performed on announcement extensions are not done, and resultant warnings or submittal denials do not occur. The shortened extensions also do not appear in any display or list that shows announcement extensions. Extra care should be taken to administer the correct type of announcement for the application when assigning shortened extensions.
attd	Calls go to the attendant and are recorded as Listed Directory Number (LDN) calls on call detail records.

Number of Members

This field appears only when **Group Type** is **sip**.

Valid entries	Usage
1 to **255**	Type the number of SIP Enablement Services (SES) trunks that are members of the trunk group. All members of an SES trunk group will have the same characteristics. **Note:** Member pages for SES trunk groups are completed automatically based on this entry and are not individually administrable.

Outgoing Display

This field allows display telephones to show the name and number of the trunk group used for an outgoing call before the call is connected. This information might be useful to you when you're trying to diagnose trunking problems.

Valid entries	Usage
y	Displays the trunk group name and number.
n	Displays the digits the caller dials.

Prefix-1

Use this field for outgoing and two-way trunk groups handling long distance service. This field appears for CO, FX, and DIOD trunk groups.

Valid entries	Usage
y/n	Enter **y** to add the prefix "1" to the beginning of the digit string for outgoing calls. Do not enter **y** for trunk groups in AAR or ARS route patterns.

Protocol Type

This field specifies the type of line signaling protocol used for DID and DIOD trunk groups. This field appears when the **Country** field is **15** and is used only by trunk group members administered on a TN2199 or TN464D vintage 3 or later circuit pack. For a list of country codes, see the Country code table on page 886.

Valid entries	Usage
inloc (Incoming local)	Enter the protocol the central office is using for this trunk group. Only the inloc protocol provides ANI.
intol (Incoming toll)	

Queue Length

Outgoing calls can wait in a queue, in the order in which they were made, when all trunks in a trunk group are busy. If you enter 0, callers receive a busy signal when no trunks are available. If you enter a higher number, a caller hears confirmation tone when no trunk is available for the outgoing call. The caller can then hang up and wait: when a trunk becomes available, Communication Manager calls the extension that placed the original call. The caller hears 3 short, quick rings. The caller doesn't need to do anything but pick up the handset and wait: Communication Manager remembers the number the caller dialed and automatically completes the call.

This field appears when the **Direction** field is **outgoing** or **two-way**.

Valid entries	Usage
0 to **100**	Enter the number of outgoing calls that you want to be held waiting when all trunks are busy.
0	Enter **0** for DCS trunks.

Service Type

Indicates the service for which this trunk group is dedicated. The following table provides a listing of predefined entries. In addition to the Services/Features listed in this table, any user-defined Facility Type of **0** (feature) or **1** (service) on the **Network Facilities** screen is allowed.

As many as 10 ISDN trunk groups can have this field administered as **cbc** (for Avaya DEFINITY Server CSI).

Valid entries	Usage
access	A tie trunk giving access to an Electronic Tandem Network.
accunet	ACCUNET Switched Digital Service — part of ACI (AT&T Communications ISDN) phase 2.
cbc	Call-by-Call service — provides different dial plans for different services on an ISDN trunk group. Indicates this trunk group is used by the Call-By-Call Service Selection feature.
dmi-mos	Digital multiplexed interface — message oriented signaling.
i800	International 800 Service — allows a subscriber to receive international calls without a charge to the call originating party.
inwats	INWATS — provides OUTWATS-like pricing and service for incoming calls.
	1 of 2

Valid entries	Usage
lds	Long-Distance Service — part of ACI (AT&T Communications ISDN) phase 2.
megacom	MEGACOM Service — an AT&T communications service that provides unbanded long-distance services using special access (switch to 4ESS switch) from an AT&T communications node.
mega800	MEGACOM 800 Service — an AT&T communications service that provides unbanded 800 service using special access (4ESS switch to switch) from an AT&T communications node.
multiquest	AT&T MULTIQUEST Telecommunications Service — dial 700 service. A terminating-user's service that supports interactive voice service between callers at switched-access locations and service provides directly connected to the AT&T Switched Network (ASN).
operator	Network Operator — provides access to the network operator.
outwats-bnd	OUTWATS Band — WATS is a voice-grade service providing both voice and low speed data transmission capabilities from the user location to defined service areas referred to as bands; the widest band is 5.
public-ntwrk	Public network calls — It is the equivalent of CO (outgoing), DID, or DIOD trunk groups. If Service Type is public-ntwrk, Dial Access can be set to **y**.
sddn	Software Defined Data Network — provides a virtual private line connectivity via the AT&T switched network (4ESS switches). Services include voice, data, and video applications. These services complement the SDN service. Do not use for DCS with Rerouting.
sdn	Software Defined Network (SDN) — an AT&T communications offering that provides a virtual private network using the public switched network. SDN can carry voice and data between customer locations as well as off-net locations.
sub-operator	Presubscribed Common Carrier Operator — provides access to the presubscribed common carrier operator.
tandem	Tandem tie trunks integral to an ETN
tie	Tie trunks — general purpose
wats-max-bnd	Maximum Banded Wats — a WATS-like offering for which a user's calls are billed at the highest WATS band subscribed to by users.
	2 of 2

Signaling Group

The screen displays this field only when the value of the entry in the **Group Type** field is **sip**.

Valid entries	Usage
1 to **650**	Type the number of the SIP Enablement Services (SES) signaling group associated with this trunk group in the **Group Number** field on the Signaling Group screen.

TAC

Enter the trunk access code (TAC) that must be dialed to access the trunk group. A different TAC must be assigned to each trunk group. CDR reports use the TAC to identify each trunk group.

Valid entries	Usage
1 to 4 digit number	Enter any number that fits the format for trunk access codes or dial access codes defined in your dial plan.
*****, #	* and # can be used as the first character in a TAC.

TN

Valid entries	Usage
1 to **100** (S87XX Series IP-PNC)	Enter a Tenant Partition number to assign this trunk group to the partition.

Tip:
Double-check your entry. If you accidentally enter an unassigned tenant partition number, the system accepts the entry but no calls go to the trunk group.

Toll Restricted

Valid entries	Usage
y	Enter **y** to prevent toll-restricted users from using a trunk access code to make restricted outgoing calls over this trunk group.
n	Enter **n** if the field is automatic or if you don't want to restrict access.

Tip:
To find out what kind of line coding a trunk group member uses, check the **Line Coding** field on the DS1 Circuit Pack screen for the DS1 port to which the member is assigned.

Trunk Flash

Trunk Flash enables multifunction telephones on Communication Manager to access central office customized services that are provided by servers at the far-end or Central Office (CO). These central office customized services are electronic features, such as conference and transfer, that are accessed by a sequence of flash signal and dial signals from the Communication Manager telephone on an active trunk call.

Valid entries	Usage
y/n	Enter **y** to allow trunk flash.

Trunk Signaling Type

This field controls the signaling used by members in private network trunk groups, mainly in Italy, Brazil, and Hungary. This field also controls the signaling used by members in public network digital trunk groups. This field displays if the **Group Type** field is **access**, **aplt**, **rlt**, **tandem**, or **tie**. Entries in this field affect which timers appear on the Administrable Timers page.

Valid entries	Usage
cont (continuous)	E&M trunks in Italy, Brazil, and Hungary can use either continuous or discontinuous signaling. Each entry specifies a set of signals and available timers used in the process of setting up and releasing connections. The type of signaling you select on Communication Manager must match the signaling type administered on the far-end server. Use these values only when all trunk group members are assigned to ports on a TN464F, TN2464, or TN2140 circuit pack. Entering one of these values causes the **Send Release Ack**, **Receive Release Ack**, and **Send Answer Supervision** fields to appear. See Trunk Type (in/out) on page 990 for more information.
dis (discontinuous)	
Use the following entries for tie trunks in Main-Satellite/Tributary networks. Each entry defines a function of the trunk group in the network. Use these values only when all trunk group members are assigned to a TN497 circuit pack.	
tgu (for outgoing trunks)	Enter **tgu** at the main server running Communication Manager to administer a tie trunk group connected to a satellite server. (This same group should be administered as **tge** at the satellite.)
	1 of 2

Valid entries	Usage
tge (for incoming trunks)	Enter **tge** at a satellite server to administer a tie trunk group connected to the main server running Communication Manager. (This same group should be administered as **tgu** at the main server.)
tgi (for internal trunks)	Enter **tgi** at to administer a two-way tie trunk group between 2 satellites or between the main server and a satellite. (This trunk group should be administered as **tgi** on both servers.)
DIOD trunks support pulsed and continuous E&M signaling in Brazil and discontinuous E&M signaling in Hungary. Use the following entries for DIOD trunks. Use these values only when all trunk group members are assigned to a TN464F (or later version) or TN2464 circuit pack.	
cont	Enter **cont** for continuous E&M signaling.
pulsed	Enter **pulsed** for pulsed E&M signaling.
discont	Leave blank forR2 signaling. Hungary uses discontinuous E&M signaling when this field is **dis**. Brazil E&M trunks use continuous and pulsed E&M.
	2 of 2

Trunk Type (in/out)

Use this field to control the seizure and start-dial signaling used on this trunk group. The setting of the **Trunk Signaling Type** field can affect the entries allowed in this field. In addition, settings might differ for incoming and outgoing trunks.

Valid entries	Usage
auto **cont** **delay** **disc** **immed** **wink**	There are numerous valid entries for this field: use the online help in Communication Manager to view all the possible combinations. Here are what the elements used in those combinations mean: ● auto — Used for immediate connection to a single preset destination (incoming CO trunks, for example). No digits are sent, because all calls terminate at the same place. ● cont — Continuous signaling is used with Italian E&M tie trunks. The server/switch seizes a trunk by sending a continuous seizure signal for at least the duration specified by the Incoming Seizure Timer. See Trunk Hunt on page 1011 for more information. ● delay — The sending switch does not send digits until it receives a delay dial signal (an off-hook signal followed by an on-hook signal) from the far-end switch, indicating that it's ready to receive the digits. ● disc —Discontinuous signaling is used with Italian tie trunks that use E&M signaling. The Avaya S8XXX Server can seize a trunk by sending a single, short signal for the duration specified by the **Normal Outgoing Seize Send** field. However, with the Three-Way Seizure option, the calling end can also send routing information to the called end by sending one or a series of brief seizure signals. ● wink — The sending server or switch does not send digits until it receives a a wink start (momentary off-hook) signal from the far-end server or switch, indicating that it's ready to receive the digits. ● immed — The sending server or switch sends digits without waiting for a signal from the far-end server or switch.
2-wire-ac **2-wire-dc** **3-wire**	These entries are used with CO trunks in Russia: enter the type of connection to your central office. Check with you network service provider if you don't know what type of connection they're using. To use these entries, the **Country** field must be **15** and the CO trunks must use ports on a TN2199 circuit board.

Tip:

When incoming trunks use the setting immed/immed, the far-end server seizes the trunk and sends digits without waiting for acknowledgment from the receiving end. When traffic is heavy, the receiving server or switch might not immediately attach a Touch Tone Receiver to a call and therefore lose digits. Use wink-start trunks or increase the dial-guard timer value on the far-end server or switch to avoid this problem.

Note:
The value in this field affects the appearance of the **Incoming Partial Dial (sec)** field on the Administrable Timer page.

Version

Use this field to adjust the signaling on multi-country CO trunk circuit packs. Entries in this field adjust signaling characteristics on these circuit packs to match the signaling characteristics of the public network in a specific country. The field appears only for CO, FX, and WATS trunk groups when the **Country** field is **5**, **16**, or **23**.

Valid entries	Usage
If the **Country** field is **5**, the **Version** field only controls TN2147 ports.	
a	Enter **a** to use standard signaling for the Netherlands public network.
b	Enter **b** to use country 1 (U.S.) signaling. The value **b** is appropriate if Communication Manager is connected to a central office using an Ericcson AXE-10 switch.
If the **Country** field is **16** or **23**, the **Version** field sets the input impedance value and only controls TN465C (vintage 2 or later) ports.	
a	Enter **a** to set input impedance to 600 Ohms.
b	Enter **b** to set input impedance to 900 Ohms. The value **b** is appropriate in Brazil.

This field appears when the "out" side of the entry in the **Trunk Type (in/out)** field is **…/wink** or **…/delay** and the **Group Type** is **tie**, **access**, **aplt**, **dmi-bos**, **rlt**, or **tandem**. The setting in this field only affects trunks administered to ports on TN760C (vintage 4 or later), TN767, TN464C (or later), and TN2242 circuit packs. If the trunk group also contains trunks assigned to ports on other types of circuit packs, those trunks are unaffected.

Valid entries	Usage
300 to **5000** in increments of 50	In general, Avaya recommends that you not change this field. If you do, remember that timing on your server running Communication Manager must be compatible with the timing on the far-end server.

Field descriptions for page 2

The figure below is only an example, and is intended to show the fields that might appear on page 2 of the Trunk Group screen for one particular trunk type. This example might not show all fields, or might show fields that normally do not appear together; it is not intended to reflect a specific trunk group type. Your own screen might vary from this example according to specific field and system settings. The list of field descriptions that follows the figure is in alphabetical order for quick reference. This list is intended to be comprehensive, and might include information on fields that are not shown in the example. The field descriptions identify fields that are specific to particular trunk group types.

Note:
This section does not cover ISDN trunks. For descriptions of the screens and fields that are unique to ISDN trunks, see ISDN Trunk Group on page 539.

Figure 338: Trunk Group screen (Group Type - CO)

```
add trunk-group next                                               Page   2 of  x
      Group Type: co                 Trunk Type:

TRUNK PARAMETERS

    Outgoing Dial Type: tone                                  Cut-Through? n
     Trunk Termination: rc                             Incoming Dial Type: tone
                                                  Disconnect Timing(msec): 500


            Auto Guard? n    Call Still Held? n     Sig Bit Inversion: none
     Analog Loss Group: 6                              Digital Loss Group: 11
                                   Trunk Gain: high

 Disconnect Supervision - In? y  Out? n
 Answer Supervision Timeout: 10            Receive Answer Supervision? n
          Administer Timers? y
```

Figure 339: Trunk Group screen (Group Type - SIP)

```
add trunk-group next                                            Page   2 of  x
      Group Type: sip

TRUNK PARAMETERS

     Unicode Name: auto

                                            Redirect On OPTIM Failure: 5000

            SCCAN? n                               Digital Loss Group: 18
                      Preferred Minimum Session Refresh Interval(sec): 600
```

Administer Timers

This field is displayed for all trunk group types except **cpe**, **h.323**, and **sip**.

Valid entries	Usage
y/n	Enter **y** to allow administration of timers on this trunk group. For **Group Type isdn**, the default value is **n**. For all other trunk group types, the default is **y**.

Analog Loss Group

This field determines which administered 2-party row in the loss plan applies to this trunk group if the call is carried over an analog signaling port in the trunk group.

Valid entries	Usage
1 to **17**	Shows the index into the loss plan and tone plan.

Answer Supervision Timeout

If the **Receive Answer Supervision** field is **n**, use this field to set the answer supervision timer for outgoing and two-way trunks. During a cut-through operation, timing begins after each outgoing digit is sent by Communication Manager and timing ceases after the far-end sends answer supervision. If the timer expires, Communication Manager acts as if it had received answer supervision. On senderized operation, the timer begins after the last digit collected is sent.

Valid entries	Usage
0 to **250**	Enter the number of seconds you want Communication Manager to wait before it acts as though answer supervision has been received from the far-end. Set this field to **0** if **Receive Answer Supervision** is **y**.

Note:
This field's setting does not override answer supervision sent from the network or from DS1 port circuit timers. To control answer supervision sent by DS1 firmware, set the **Outgoing End of Dial (sec)** field on the Administrable Timers page of the Trunk Group screen.

Auto Guard

This field appears if the **Group Type** field is **co** or **fx**. This field controls ports only on TN438B, TN465B, and TN2147 circuit packs. TN438B ports have hardware support for detecting a defective trunk. TN465B and TN2147 ports consider a trunk defective if no dial tone is detected on an outgoing call, and the **Outpulse Without Tone** field is **n** on the **Feature-Related System Parameters** screen.

Valid entries	Usage
y/n	Enter **y** to prevent repeated seizures of a defective trunk. Communication Manager will do a maintenance busy-out on these trunks.

Bit Rate

This field specifies the baud rate to be used by pooled modems. This field appears when the **Comm Type** field is **avd** or **rbavd**. It also appears if the **Comm Type** field is **data**, but only if the **ISDN-PRI** field is **y** on the System Parameters Customer-Options (Optional Features) screen.

Valid entries	Usage
300	Enter the speed of the fastest modem that will use this trunk group.
1200	
2400	
4800	
9600	
19200	

Call Still Held

This field appears if the **Group Type** field is **co** or **fx**. This field is used when the receiving end server initiates the disconnection of incoming calls. It effectively extends the Incoming Glare Guard timer by 140 seconds. This field affects only TN438B, TN465B, and TN2147 ports and is used primarily when the Country Code field is **2**.

Valid entries	Usage
y/n	Enter **y** to prevent glare by delaying an outgoing seizure of a trunk for at least 140 seconds after it is released from an incoming call.

Cut-Through

This field appears when the **Outgoing Dial Type** field is either **rotary** or **tone**.

SECURITY ALERT:
Entering **y** in this field reduces your ability to prevent toll fraud.

Valid entries	Usage
y	Enter **y** to allow users to get dial tone directly from the central office. Outgoing calls over this trunk group will bypass AAR/ARS call type checking, and will bypass any of your administered restrictions (such as COR or FRL).
n	Enter **n** and the user will receive switch dial tone. Instead of digits being sent to the central office, they will be collected and checked against administered restrictions. If no restrictions apply, the digits are sent to the central office.

Cyclical Hunt

When a call is offered to a trunk group, Communication Manager searches for an available trunk. This field, which appears when the **Direction** field is **two-way** and the **Trunk Type** field is **loop-start**, controls the starting point of this search.

You can change this field from **n** to **y** at any time. To change from **y** to **n**, however, all the trunks in the group must be idle or busied out.

Valid entries	Usage
y	Enter **y** to have Communication Manager start its search from the last trunk seized. This method is faster, and thus better suited for high-traffic trunk groups.
n	Enter **n** to have Communication Manager start each search at member 1 (the first trunk administered on the Group Member Assignments page).

Dial Detection

Applies only to TN2199 ports. The **Country** field must be **15**.

Valid entries	Usage
A-wire	Indicate whether digit pulses are detected by observing the A-wire (default) or the B-wire only.
B-wire	

Digital Loss Group

This field determines which administered 2-party row in the loss plan applies to this trunk group if the call is carried over a digital signaling port in the trunk group.

Valid entries	Usage
1 to **19**	Shows the index into the loss plan and tone plan.

Digits

This field is used with the **Digit Treatment** field, and its meaning depends on the entry in that field. If the **Digit Treatment** field is **absorption**, this field specifies *how many* digits are deleted. If the **Digit Treatment** field is **insertion**, this field identifies the *specific digits* that are added.

Valid entries	Usage
1 to **5**	Enter the number of digits to be deleted (absorbed).
Up to 4 digits, including * and #	Enter the actual digits to be added (inserted).
blank	This field can be blank only if the **Digit Treatment** field is blank.

Digit Treatment

Use this field to modify an incoming digit string (as on DID and tie trunks, for example) by adding or deleting digits. You'll need to do this if the number of digits you receive doesn't match your dial plan.

Valid entries	Usage
blank	The incoming digit string is not changed.
absorption	Deletes digits, starting at the beginning of the string.
insertion	Adds digits, starting at the beginning of the string.

If you enter absorption or insertion, then you must enter a value in the **Digits** field.

Disconnect Supervision-In

This field indicates whether Communication Manager receives disconnect supervision for incoming calls over this trunk group. It appears when the **Direction** field is **incoming** or **two-way**. (If the **Direction** field is **outgoing**, Communication Manager internally sets this field to **n**.)

The entry in this field is crucial if you allow trunk-to-trunk transfers. (To allow trunk-to-trunk transfers involving trunks in this group, this field must be **y** and the **Trunk-to-Trunk Transfer** field on the Feature-Related System Parameters screen must be **y**).

If a user connects 2 trunks through conference or transfer, and neither far-end server on the resulting connection provides disconnect supervision, the trunks involved will not be released because Communication Manager cannot detect the end of the call. Communication Manager will not allow trunk-to-trunk transfers unless it believes that at least one party on the call can provide disconnect supervision. Therefore, setting this field incorrectly might cause trunks to become unusable until the problem is detected and the trunks are reset.

Valid entries	Usage
y	Enter **y** to allow trunk-to-trunk transfers involving trunks in this group. Enter **y** if the far-end server/switch sends a release signal when the calling party releases an incoming call, and you want to make the far-end server/switch responsible for releasing the trunk. Enter **y** to enhance Network Call Redirection.
n	Enter **n** if the far-end server/switch doesn’t provide a release signal, if your hardware can’t recognize a release signal, or if you prefer to use timers for disconnect supervision on incoming calls. Entering **n** prevents trunk-to-trunk transfers involving trunks in this group.

CAUTION:
In general, U.S. central offices provide disconnect supervision for incoming calls but not for outgoing calls. Public networks in most other countries do not provide disconnect supervision for incoming or outgoing calls. Check with your network services provider.

Disconnect Supervision-Out

This field indicates whether Communication Manager receives disconnect supervision for outgoing calls over this trunk group. It appears when the **Direction** field is either **outgoing** or **two-way**. (If the **Direction** field is **incoming**, Communication Manager internally sets this field to **n**.)

The entry in this field is crucial if you allow trunk-to-trunk transfers. (To allow trunk-to-trunk transfers involving trunks in this group, this field must be **y** and the **Trunk-to-Trunk Transfer** field on the Feature-Related System Parameters screen must be **y**). If a user connects 2 trunks through conference or transfer, and neither far-end server/switch on the resulting connection provides disconnect supervision, the trunks involved won't be released because Communication Manager can't detect the end of the call. Communication Manager does not allow trunk-to-trunk transfers unless it believes that at least one party on the call can provide disconnect supervision. Therefore, setting this field incorrectly might cause trunks to become unusable until the problem is detected and the trunks are reset.

Also, remember that Communication Manager must receive *answer* supervision on outgoing analog CO, FX, WATS, Tie, Tandem, and Access trunks before it recognizes a disconnect signal. If this trunk group does not receive *answer* supervision from the far-end server/switch, and you enter **y** in this field, Communication Manager internally sets the field to **n**.

Valid entries	Usage
y	Enter **y** to allow trunk-to-trunk transfers involving trunks in this group. Enter **y** if the far-end sends a release signal when the called party releases a call an outgoing call, and you want to make the far-end responsible for releasing the trunk. The **Answer Supervision Timeout** field must be **0** and the **Receive Answer Supervision** field must be **y** for the switch to recognize a **y** entry. Enter **y** to enhance Network Call Redirection.
n	Enter **n** if the far-end server/switch doesn't provide a release signal, if your hardware can't recognize a release signal, or if you prefer to use timers for disconnect supervision on outgoing calls. Entering **n** prevents trunk-to-trunk transfers involving trunks in this group.

CAUTION:
Do not set this field to **y** unless you are certain that the far-end server/switch will provide answer supervision and disconnect supervision. Most public networks do not provide disconnect supervision over analog trunks. Check with your network services provider.

Disconnect Timing (msec)

This field specifies the minimum time in milliseconds that the central office or far-end server requires to recognize that your end has disconnected from a call. This timer does not affect ports on a circuit pack that uses the administrable Incoming Disconnect and Outgoing Disconnect timers; in fact, settings on those two timers override this field.

Valid entries	Usage
140 to **2550** ms in increments of 10	The default of **500** is an industry standard and you shouldn't change it. If you set this field too high, your server/switch won't disconnect sometimes when it should; too low, and it will disconnect when it shouldn't.

Disconnect Type

This field indicates which side or user controls the disconnect, where **A** refers to the calling party and **B** refers to the called party. Appears only if the **Country** field is **15** and the **Trunk Type** field is **2-wire-ac**, **2-wire-dc**, or **3-wire**.

This applies *only* to the TN2199 port.

Valid entries	Usage
AandB	Both parties control the disconnect.
AorB	Either party controls the disconnect.

Drop Treatment

This field only applies to DID trunks. It determines what the calling party hears when the called party terminates an incoming call.

Valid entries	Usage
intercept	Select one. For security reasons, it's better to apply a tone: silence could provide an opening for hackers.
busy	
silence	

Note:
In Italy, the **Drop Treatment** field must be administered as **intercept** for all DID trunk groups.

Duplex

This field specifies whether a two-way trunk group allows simultaneous transmission in both directions. This field appears when the **Comm Type** field is **avd** or **rbavd**. It also appears if the **Comm Type** field is **data**, but only if the **ISDN-PRI** field is enabled on the **System Parameters Customer-Options (Optional Features)** screen.

Note:
Even if the trunk group supports full-duplex transmission, other equipment in a circuit might not.

Valid entries	Usage
full	Enter **full** in most cases: this allows simultaneous two-way transmission, which is most efficient.
half	Enter **half** to support only one transmission direction at a time.

End-to-End Signaling

This field appears when the **Group Type** field is **cpe** (customer-provided equipment trunk groups). Auxiliary equipment such as paging equipment and music sources might be connected to Communication Manager by auxiliary trunks. Communication Manager might send DTMF signals (touch tones) to these devices. This field sets the duration of these tones.

Valid entries	Usage
60 to **360** ms in increments of 10	Use this field to set the duration in milliseconds of the touch-tone signal that is sent to the connected equipment.

Note:
For trunks that do not receive real answer supervision, a "connect" Event report is sent when the Answer Supervision Timeout occurs.

Expected Digits

Note:
Set this field to **blank** if the **Digit Treatment** field is set to **insert** and the **Digits** field contains a feature access code (for example, AAR or ARS) followed by digits. In this case, the number of digits expected are set on the AAR and ARS Digit Analysis Table and AAR and ARS Digit Conversion Table.

Valid entries	Usage
1 to **18**	Enter the number of digits that the far-end server sends for an incoming connection. If your end is absorbing digits on this trunk group, the entry in this field must be larger than the entry in the **Digits** field.
	If you leave this field blank, you cannot administer digit absorption.

Extended Loop Range

This field appears only for a **DID** trunk group and is used only with the TN459A circuit pack.

Valid entries	Usage
y/n	Enter **y** or **n** depending on the distance between the central office and your server. If greater than the required distance, then the field should be **y**.

Format

This field appears if the **Send Calling Number** field is **y** or **r** or the **Send Connected Number** field is **y** or **r**. This specifies the encoding of Numbering Plan Indicator for identification purposes in the Calling Number and/or Connected Number IEs, and in the QSIG Party Number. Valid entries are **public**, **unknown**, **private**, and **unk-pvt**. **Public** indicates that the number plan according to CCITT Recommendation E.164 is used and that the **Type of Number** is national. **Unknown** indicates the **Numbering Plan Indicator** is unknown and that the **Type of Number** is unknown. **Private** indicates the **Numbering Plan Indicator** is PNP and the **Type of Number** is determined from the Numbering - Private Format screen. An entry of **unk-pvt** also determines the **Type of Number** from the Numbering - Private Format screen, but the **Numbering Plan Indicator** is unknown.

Group Type

Displays the type of trunk group selected for this field on page 1 of the Trunk Group screen. For details, see the field description for the page 1 **Group Type** field.

Incoming Calling Number - Delete

The Incoming **Calling Number - Delete**, **Insert**, and **Format** fields are the administrable fields for the Calling Line Identification Prefix feature. They appear when the **Direction** field is **incoming** or **two-way**.

Valid entries	Usage
1 to **15**, **all**, or blank	Enter the number of digits, if any, to delete from the calling party number for all incoming calls on this trunk group.

Incoming Calling Number - Format

This field indicates the TON/NPI encoding applied to CPN information modified by the CLI Prefix feature. This encoding does not apply to calls originating locally. The **Numbering Format** field on page 2 of this screen applies to calls originated from this server running Communication Manager.

If this field is blank, Communication Manager passes on the encoding received in the incoming setup message. If the incoming setup message did not contain CPN information and digits are added, the outgoing message will contain these digits. If the **Format** field is blank in this case, the value defaults to **pub-unk**.

If the **Format** field on page 2 of this screen is also administered as **unknown**, the trunk group is modified to "unk-unk" encoding of the TON/NPI. Therefore, this field also must contain a value other than **unknown**.

Note:
The values for this field map to the Type of Numbering (TON) and Numbering Plan Identifier (NPI) values shown below.

Valid entries	Type of numbering (TON)	Numbering plan identifier (NPI)
blank	incoming TON unmodified	incoming NPI unmodified
natl-pub	national(2)	E.164(1)
intl-pub	international(1)	E.164(1)
locl-pub	local/subscriber(4)	E.164(1)
pub-unk	unknown(0)	E.164(1)

Valid entries	Type of numbering (TON)	Numbering plan identifier (NPI)
lev0-pvt	local(4)	Private Numbering Plan - PNP(9)
lev1-pvt	Regional Level 1(2)	Private Numbering Plan - PNP(9)
lev2-pvt	Regional Level 2(1)	Private Numbering Plan - PNP(9)
unk-unk	unknown(0)	unknown(0)

Incoming Calling Number - Insert

Valid entries	Usage
Enter up to 15 characters (0 to 9), **all**, or blank	Enter up to specific digits, if any, to add to the beginning of the digit string of incoming calls when the calling party is a member of this trunk group.

Incoming Dial Type

Indicates the type of pulses required on an incoming trunk group. Usually, you should match what your central office provides. This field appears when **Group Type** is **Access**, **APLT**, **DID**, **DIOD**, **DMI-BOS**, **FX**, **RLT**, **Tandem**, or **WATS**. It also appears for **Tie** trunk groups when the **Trunk Signaling Type** field is blank, **cont**, or **dis**.

Valid entries	Usage
tone	Enter **tone** to use Dual Tone Multifrequency (DTMF) addressing, also known as "touchtone" in the U.S. Entering **tone** actually allows the trunk group to support both DTMF and rotary signals. Also, if you're using the Inband ANI feature, enter **tone**. For pulsed and continuous E&M signaling in Brazil and for discontinuous E&M signaling in Hungary, use **tone**.
rotary	Enter **rotary** if you only want to allow the dial pulse addressing method used by non-touch tone telephones. Though the tone entry supports rotary dialing as well, it's inefficient to reserve touch tone registers for calls that don't use DTMF.
mf	Enter **mf** if the **Trunk Signaling Type** field is blank. The **Multifrequency Signaling** field must be enabled on the System Parameters Customer-Options (Optional Features) screen in order for you to enter **mf** here. You cannot enter **mf** if the **Used for DCS** field (field descriptions for page 2) is **y**. For pulsed and continuous E&M signaling in Brazil and for discontinuous E&M signaling in Hungary, use **mf**.

Incoming Dial Tone

Indicates whether or not your server running Communication Manager will give dial tone in response to far-end seizures of the trunk group.

Valid entries	Usage
y	Enter **y** if the incoming trunk group transmits digits. For example, you would enter **y** for two-way, dial-repeating tie trunks that users select by dialing a trunk access code.
n	Enter **n** for trunks that aren't sending digits, such as tandem or incoming CO trunks.

Incoming Rotary Timeout (sec)

Call setup at central offices that still use older switching equipment, such as step-by-step technology, is considerably longer then at central offices with more modern servers. If you're receiving digits with incoming calls from a central office that uses less efficient switching technology, your server needs to allow more time to ensure it receives all the incoming digits. When the **Incoming Dial Type** field is **rotary**, use this field to set the maximum time your end will wait to receive all incoming digits from the far-end switch.

Valid entries	Usage
5 to **99** or blank	If the system is connected to a step-by-step central office, or any CO using older switching technology, enter at least **18** seconds; if not, enter at least **5** seconds.

Line Length

This field appears only when the **Group Type** field is **tie** and the **Trunk Signaling Type** field is **tge**, **tgi**, or **tgu**.

Valid entries	Usage
short	Indicate the line length.
long	

Note:
Unless one or more trunk members are administered, the administered value is not saved when you submit the screen (press **Enter**).

Outgoing Dial Type

This field sets the method used to transmit digits for an outgoing call. Usually, you should match what your central office provides. This field appears for Access, APLT, CO, DIOD, DMI-BOS, FX, RLT, and WATS trunk groups. It also appears for Tie trunk groups when the **Trunk Signaling Type** field is blank, **cont**, or **dis**.

DIOD trunks support pulsed and continuous E&M signaling in Brazil and discontinuous E&M signaling in Hungary.

Valid entries	Usage
tone	Enter **tone** to use Dual Tone Multifrequency (DTMF) addressing, also known as "touchtone" in the U.S. Entering **tone** actually allows the trunk group to support both DTMF and rotary signals. For pulsed and continuous E&M signaling in Brazil and for discontinuous E&M signaling in Hungary, use **tone** or **mf**.
rotary	Enter **rotary** if you only want to allow the dial pulse addressing method used by non-touch tone telephones. If you have a full touch tone system internally and a connection to a central office that only supports rotary dialing, for example, it would be appropriate to enter **rotary**.
r1mf	Enter **r1mf** for CAMA trunk groups. It is the only outgoing dial type allowed on CAMA trunk groups. Enter **r1mf** to allow Russian MF Packet Signaling on outgoing trunks. Russian MF Packet Signaling carries calling party number and dialed number information. **Group type** field must be set to **co**.
mf	Enter **mf** if the **Trunk Signaling Type** field is blank. The **Multifrequency Signaling** field must be enabled on the System Parameters Customer-Options (Optional Features) screen in order for you to enter **mf** here. You cannot enter **mf** if the **Used for DCS** field (Field descriptions for page 2) is **y**. For pulsed and continuous E&M signaling in Brazil and for discontinuous E&M signaling in Hungary, use **tone** or **mf**.
automatic	Enter **automatic** for tie trunks if the **Trunk Signaling Type** field is blank. This provides "cut-through" operation to outgoing callers who dial a trunk access code, connecting them directly to central office dial tone and bypassing any toll restrictions administered on Communication Manager.

Preferred Minimum Session Refresh Interval (sec)

The system displays this field only for the SIP trunk groups that do not support Seamless Converged Communications Across Network (SCCAN) calls.

Use this field to set the value of the session refresh timer of non-SCCAN applications. The timer starts once Communication Manager establishes a session with the far-end SIP entities. Communication Manager then sends a session refresh request as a Re-INVITE or an UPDATE message after every timer interval to the SIP entities. If the SIP entities do not receive a session refresh request before the interval period expires, the session ends.

Valid entries	Usage
90 to **64800**	Defines the session refresh timer. The default value is 600.

Receive Answer Supervision

Use this field to specify whether the network provides answer supervision for a trunk group.

For Outbound Call Management applications, set this field to **y** for trunks supporting network answer supervision. For trunks that do not receive a real answer, this field determines when the CallVisor Adjunct-Switch Application Interface (ASAI) connect event is sent.

Valid entries	Usage
y	Enter **y** if the network provides answer supervision. Set the **Answer Supervision Timeout** field to **0**.
n	Enter **n** if the network does not provide answer supervision, and set the **Answer Supervision Timeout** field. Also enter **n** for incoming trunk groups.

Note:
When you set this field to **y**, the **Outgoing End of Dial (sec)** field is not displayed. The firmware timeout on circuit packs controlled by the **Outgoing End of Dial (sec)** field is automatically set to **0**.

Receive Release Ack

This field appears when the **Trunk Signaling Type** field is **cont** or **dis** and only applies to TN2140 ports (used for Italian and Hungarian tie trunks).

Valid entries	Usage
y/n	Enter **y** if Communication Manager receives a release acknowledgment in response to a forward or backward release signal.

Redirect on OPTIM failure

This field is a timer that determines how long to wait for OPTIM to intercede before the call is redirected. Redirect on OPTIM failure is sometimes known as ROOF.

Valid entries	Usage
250 to **32000 milliseconds**	See Off-PBX documentation for details on this field.

SCCAN

This field appears when the **Group Type** field is **sip** and **Enhanced EC500** on the System Parameters Customer-Options (Optional Features) screen is set to **y**. When this field is set to **y**, the non-SCCAN-associated fields are hidden.

Valid entries	Usage
y/n	Enter **y** to indicate that this trunk group provides support for incoming SCCAN calls. Default is **n**.

Send Answer Supervision

This field appears when the **Trunk Signaling Type** field is **cont** or **dis** and only applies to TN2140 ports.

Valid entries	Usage
y/n	Enter **y** to make Communication Manager signal the calling server when an incoming call is answered. You can only set this field to **y** if the **Direction** field is **incoming** or **two-way**.

Send Release Ack

This field appears when the **Trunk Signaling Type** field is **cont** or **dis** and only applies to TN2140 ports (used for Italian and Hungarian tie trunks).

Valid entries	Usage
y/n	Enter **y** to send a release acknowledgment in response to a forward or backward release signal.

Sig Bit Inversion

When transmission facilities use bit-oriented signaling (such as CAS), 2 bits are used to transmit seizure and release signals for calls. Called the A-bit and the B-bit, their meaning can vary. For example, in the A-bit a "1" might mean on-hook and a "0" might mean off-hook. The entry in the **Country Protocol** field on the DS1 Circuit Pack screen sets the default meaning of these bits.

For trunk ports on TN2242 and TN464B and later circuit packs, this field allows you to invert the A- and B-bits as necessary so that the far-end server/switch can understand seizure and release signals from Communication Manager. If the far-end server, such as a central office, on this trunk group interprets the A- and B-bits differently from the default, you might need to invert one or both bits — to change "1" to "0" and vice-versa in the A-bit, for example.

Valid entries	Usage
A	For the TN464B and later circuit packs, indicate which bits, if any, should be inverted.
B	
A&B	
none	
A and **none**	For the Japanese 2Mbit trunk circuit pack, indicate which bits, if any, should be inverted.

Supplementary Service Protocol

Appears only when **Group Type** is **ISDN**.

Valid entries	Usage
a	Allows ASAI Flexible Billing. AT&T, Bellcore, Nortel. When the **Country Code** field on the DS1 screen is **1A**, SSA selects AT&T custom supplementary services. When the **Country Code** field on the DS1 screen is **1B**, SSA selects Bellcore Supplementary Services. When the **Country Code** field on the DS1 screen is **1C**, SSA selects Nortel Proprietary Supplementary Services.
b	QSIG; also used for SBS signaling trunk groups when full QSIG functionality is needed. If the international call routing parameters are not administered on the **system-parameters features** screen and SBS is enabled on a trunk screen, a warning is displayed: `Must set INTERNATIONAL CALL ROUTING parameters on system-parameters features`.

Valid entries	Usage
c	ETSI Use c protocol for Network Call Deflection.
d	ECMA QSIG
e	Allows ASAI Flexible Billing. Allows DCS with rerouting. **DCS with Rerouting** must be **y**, and the **Used for DCS** field on the Trunk Group screen must be **y**.
f	Feature Plus
g	ANSI. Available only if, on the System Parameters Customer-Options (Optional Features) screen, the **ISDN-PRI** or **ISDN-BRI** field is **y** or the **Used for DCS** field is **y**. Use **g** protocol for Network Call Transfer.

Synchronization

This field determines whether the trunk group will use synchronous or asynchronous communications. This field appears if:

- the **Group Type** field is **dmi-bos** or **isdn**, or
- the **Group Type** field is **access**, **co**, **fx**, **tandem**, **tie**, or **wats** and the **Comm Type** field is **avd** or **rbavd**, or
- the **Group Type** field is **access**, **co**, **fx**, **tandem**, **tie**, or **wats**, the **Comm Type** field is **data**, and the **ISDN-PRI** field or the **ISDN-BRI Trunks** field is **y** on the System Parameters Customer-Options (Optional Features) screen.

Valid entries	Usage
async	Do not change this field without the assistance of Avaya or your network service provider.
sync	

Trunk Gain

This field specifies the amplification applied to the trunks in this group. With the values of the **Trunk Termination** and **Country** fields, the value in this field also determines the input and trans-hybrid balance impedance for TN465B, TN2146, TN2147, and TN2184 ports. All other CO and DID circuit packs are set automatically to high

Valid entries	Usage
high	Enter **high** if users complain of low volume.
low	Enter **low** if users complain of squeal or feedback.

Trunk Hunt

Communication Manager performs a trunk hunt when searching for available channels within a facility in an ISDN trunk group. With both **ascend** and **descend**, all trunks within an ISDN trunk group are selected based on this field and without regard to the order in which trunks are administered within the trunk group. When using ISDN-BRI interfaces, only **cyclical** is allowed

.

Valid entries	Usage
ascend	Enter to enable a linear trunk hunt search from the lowest to highest numbered channels.
cyclical	Enter to enable a circular trunk hunt based on the sequence the trunks were administered within the trunk group.
descend	Enter for a linear trunk hunt search from the highest to lowest numbered channels.

Note:
The cyclical option cannot be set if the trunk group using ISDN-PRI interfaces is to be used for Wideband operations (the **Wideband Support** field set to **y**).

The search can be administered per ISDN-PRI trunk group, but it infers the direction of search within all ISDN-PRI facilities (or portions of those facilities) administered within the trunk group.

Trunk Termination

This field adjusts the impedance of the trunk group for optimal transmission quality. Check with your service provider if you're not sure of the distance to your central office.

Valid entries	Usage
600ohm	Enter **600ohm** when the distance to the central office or the server at the other end of the trunk is less than 3,000 feet.
rc	Enter **rc** when the distance to the central office or the server at the other end of the trunk is more than 3,000 feet.

Trunk Type

Use this field to control the seizure and start-dial signaling used on this trunk group. Entries in this field vary according to the function of the trunk group and *must* match the corresponding setting on the far-end server or switch. This field appears for CO, DID, FX, and WATS trunk groups.

Valid entries	Usage
ground-start	Use ground-start signaling for two-way trunks whenever possible: ground-start signaling avoids glare and provides answer supervision from the far end.
loop-start	In general, try to use loop-start signaling only for one-way trunks. Loop-start signaling is susceptible to glare and does not provide answer supervision.
auto/auto **auto/delay** **auto/immed** **auto/wink**	The term before the slash tells Communication Manager how and when it will receive incoming digits. The term after the slash tells Communication Manager how and when it should send outgoing digits. ● auto — Used for immediate connection to a single preset destination (incoming CO trunks, for example). No digits are sent, because all calls terminate at the same place. ● delay — The sending server running Communication Manager does not send digits until it receives a delay dial signal (an off-hook signal followed by an on-hook signal) from the far-end server or switch, indicating that it's ready to receive the digits. ● wink — The sending server running Communication Manager does not send digits until it receives a a wink start (momentary off-hook) signal from the far-end server or switch, indicating that it's ready to receive the digits. ● immed — The sending server running Communication Manager sends digits without waiting for a signal from the far-end server or switch.
2-wire-ac **2-wire-dc** **3-wire**	These entries are used with CO trunks in Russia: enter the type of connection to your central office. Check with you network service provider if you don't know what type of connection they're using. To use these entries, the **Country** field must be **15** and the CO trunks must use ports on a TN2199 circuit board.

Unicode Name

Appears when the **Group Type** field is **sip**. The value for this field is only examined for calls to SIP Enablement Services (SES) stations over an SES trunk group and is used to determine whether to send the name (as administered on the Station screen) or the native name (Unicode name is administered through an Integrated Management application). Note that Avaya BRI stations support ASCII characters only. They do not support non-ASCII characters, such as Eurofont or Kanafont. Therefore, if you use non-ASCII characters in any Communication Manager **Name** field, such characters do not display correctly on a BRI station.

Valid entries	Usage
y	Enter **y** to indicate the use of Unicode Name.
n	Enter **n** to use the name as specified on the station form. **Note:** Any non-ASCII characters in the name may not appear correctly on a SIP Phone.
auto	The choice to use the station name or the unicode name is automatically determined based on the called phone's capability and the user's preference.

Wink Timer (msec)

This field appears when one of the "**wink**" options is entered in the **Trunk Type** field. This field allows you to reduce the risk of glare by controlling part of call setup. Requirements for the United States domestic network specify that the wink signal for wink-start trunks must begin within 5 seconds after a trunk is seized. For trunks with a delay-dial start, the wink must not last longer than 5 seconds. While some circuit packs are hard-coded to allow the full 5 seconds in both cases, other circuit packs allow you reduce the allowed start time and duration, thus reducing the window in which glare could occur.

Unlike other fields on this screen, the **Wink Timer** field therefore controls 2 different variables. What your entry does depends on the outgoing value in the **Trunk Type (in/out)** field.

Setting of the Trunk Type (in/out) field	What the Wink Timer field sets
…/wink	Maximum duration of the wink signal (wait-for-wink-to-end)
…/delay	Maximum interval after trunk seizure for the wink to begin (wait-for-wink-to-start)

Field descriptions for page 3

The figure below is only an example, and is intended to show most of the fields that might appear on page 3 of the Trunk Group screen. This example might not show all fields, or might show fields that normally do not appear together; it is not intended to reflect a specific trunk group type. Your own screen might vary from this example according to specific field and system settings. The list of field descriptions that follows the figure is in alphabetical order for quick reference. This list is intended to be comprehensive, and might include information on fields that are not shown in the example. The field descriptions identify fields that are specific to particular trunk group types.

Note:
This section does not cover ISDN trunks. For descriptions of the screens and fields that are unique to ISDN trunks, see ISDN Trunk Group on page 539.

Figure 340: Trunk Group screen (page 3)

```
add trunk-group next                                               Page 3 of x
                              TRUNK FEATURES
           ACA Assignment? _          Measured: ____      Wideband Support? _
 Long Holding Time(hours): _                             Maintenance Tests? _
  Short Holding Time(sec): _  Data Restriction? _     NCA-TSC Trunk Member: _
  Short Holding Threshold: __        Send Name: _      Send Calling Number: _
             Used for DCS? _                          Send EMU Visitor CPN?

Suppress # Outpulsing? _  Numbering Format: _______
Outgoing Channel ID Encoding: _________     UUI IE Treatment: ______
                                        Maximum Size of UUI IE Contents: ___
                                              Replace Restricted Numbers? _
                                             Replace Unavailable Numbers? _
                                                   Send Connected Number: _
                                                Hold/Unhold Notifications? _

        Send UUI IE? _
          Send UCID? _                 BRS Reply-best DISC Cause Value: __
                                                   Ds1 Echo Cancellation? _

                                        US NI Delayed Calling Name Update? _
Show ANSWERED BY on Display? y
                          Network (Japan) Needs Connect Before Disconnect? _

Time (sec) to Drop Call on No Answer:_
Outgoing ANI:_
R2 MFC Signaling:_

DSN Term? n      Precedence Incoming _______   Precedence Outgoing _______
Used Only for Paging?                         Voice Paging Timeout (sec)? 10
```

CAUTION:
Customers: Do not change fields on this page without assistance from Avaya or your network service provider.

Abandoned Call Search

Use this field when the **Trunk Type** field is **ground-start**. Abandoned Call Search is designed to work with analog ground-start CO trunks that do not provide disconnect supervision. Your central office must support Abandoned Call Search for the feature to work properly. If your central office provides disconnect supervision, you do not need to use the Abandoned Call Search feature.

Valid entries	Usage
y/n	Enter **y** if this trunk group conducts an Abandoned Call Search to identify ghost calls.

ACA Assignment

Valid entries	Usage
y/n	Enter **y** if you want Automatic Circuit Assurance (ACA) measurements to be taken for this trunk group. If **y** is entered, complete the **Long Holding Time**, **Short Holding Time**, and **Short Holding Threshold** fields.

Charge Conversion

Communication Manager multiplies the number of charge units by the value of this field and displays it as a currency amount. If there is no value in this field, Communication Manager displays the number of charge units without converting it to currency. This field appears for CO, DIOD, FX, and WATS trunk groups when the **Direction** field is **outgoing** or **two-way**. For ISDN trunk groups, it appears when the **Charge Advice** field is *not* **none**.

Valid entries	Usage
1 to **64**, **500**	Enter the value of a charge unit in terms of the currency you use.

Charge Type

Entries in this field are text strings you use to describe charges related to a telephone call. This field appears for CO, DIOD, FX, and WATS trunk groups when the **Direction** field is **outgoing** or **two-way**. For ISDN trunk groups, it appears when the **Charge Advice** field is *not* **none**.

Valid entries	Usage
1 to 7 characters (embedded spaces count as characters) blank	Enter the words or characters you want to appear on telephone displays after the charge amount. Most likely you can use either the currency symbol or the charge type, bur not both.

Connected to CO

This field appears when the **Group Type** field is **tie**.

Valid entries	Usage
y/n	Enter **y** to allow overlap sending to a Central Office.

Currency Symbol

This field appears for CO, DIOD, FX, and WATS trunk groups when the **Direction** field is **outgoing** or **two-way**. For ISDN trunk groups, it appears when the **Charge Advice** field is *not* **none**.

Valid entries	Usage
1 to 3 characters (leading and embedded spaces count as characters) or blank	Enter the symbol you want to appear on telephone displays before the charge amount.

Data Restriction

If **y**, whisper page is denied on this trunk.

Valid entries	Usage
y/n	Enter **y** to prevent features from generating tones on a data call that would cause erroneous data transmission.

Decimal Point

This field appears for CO, DIOD, FX, and WATS trunk groups when the **Direction** field is **outgoing** or **two-way**. For ISDN trunk groups, it appears when the **Charge Advice** field is *not* **none**. Chose the appropriate representation for a decimal point as it will appear on telephone displays. Entering **comma** or **period** in this field divides the charge value by 100.

Note:
If the received charge contains no decimals, no decimal point is displayed (that is, the administered decimal point is ignored for charge information received with no decimals). On an upgrade from a QSIG trunk group with the **Decimal Point** field administered as **none**, the field defaults to **period**.

Valid entries	Usage
comma	If the received charge contains decimals, the charge is displayed at the calling endpoint's display with a comma as the decimal point.
period	This is the default. If the received charge contains decimals, the charge is displayed at the calling endpoint's display with a period as the decimal point.
none	No decimal point is displayed.

DS1 Echo Cancellation

Reduces voice call echo.

Note:
Changes to the **DS1 Echo Cancellation** field do not take effect until one of the following occurs:

- Port is busied-out/released
- Trunk group is busied-out/released
- SAT command test trunk group is performed
- Periodic maintenance runs

Valid entries	Usage
y/n	Enter **y** to allow echo cancellation on a per port basis.

DSN Term

Valid entries	Usage
y/n	Use the **DSN Term** field to identify the trunk group as a DSN termination telephone. The default is **n**.

Format

The screen displays this field if the **Send Calling Number** field is either **y** or **r**, or the **Send Connected Number** field is either **y** or **r**. The **Numbering Format** field specifies the encoding of Numbering Plan Indicator for identification purposes in the Calling Number and/or Connected Number IEs, and in the QSIG Party Number. Valid entries are **public**, **unknown**, **private**, and **unk-pvt**.

Valid entries	Usage
Public	Indicates that the number plan according to CCITT Recommendation E.164 is used and that the **Type of Number** is national. This is the default entry for SIP Enablement Services (SES) trunks.
Unknown	Indicates that the **Numbering Plan Indicator** is unknown and that the **Type of Number** is unknown.
Private	Indicates the **Numbering Plan Indicator** is PNP and the **Type of Number** is determined from the Numbering - Private Format screen.
unk-pvt	Also determines the **Type of Number** from the Numbering - Private Format screen, but the **Numbering Plan Indicator** is unknown.

Glare Handling

This field determines what Communication Manager will do when glare occurs. This field appears when the **Direction** field is **two-way** and the outgoing side of the **Trunk Type** field is either **…/wink** or **…/delay**.

If you enter **control** or **backoff**, and ports for the trunk group are not capable of detecting glare, warnings are generated. The following circuit packs can detect glare: TN767 (all releases), TN760C (or later releases), and TN464C (or later releases).

Valid entries	Usage
control	Communication Manager will seize the trunk and proceed with call setup. The other switch will find another trunk.
backoff	The other server or switch will seize the trunk and proceed with call setup. Your server/switch will find another trunk.
none	

Hold/Unhold Notifications

Appears only when the **Group Type** field is **isdn**. Use this field to indicate whether or not hold/ unhold messages are sent over the isdn trunk when a user places a call on hold/unhold.

Valid entries	Usage
y/n	Enter **y** to enable sending of hold/unhold messages over this isdn trunk. Default is **y**.

Incoming Tone (DTMF) ANI

This field appears only when the **Incoming Dial Type** field is **tone**. Digits received through Automatic Number Identification (ANI) are printed on a CDR record, passed to the INTUITY AUDIX and ASAI interfaces, and displayed on the telephone (and on tandem calls if the outgoing trunk requires ANI). Then the digits are sent to the outgoing trunk.

Valid entries	Usage
ANI*DNIS	If 555-3800 calls extension 81120, the trunk group receives *55538000*81120*. The telephone displays Call from 555-3800.
ANI*DNIS*	If 555-3800 calls extension 81120, the trunk group receives 55538000*81120*. The telephone displays Call from 555-3800.
no	

Internal Alert

Valid entries	Usage
y/n	Enter **y** if internal ringing and coverage will be used for incoming calls.

Long Holding Time (hours)

This field appears only when the **ACA Assignment** field is **y**.

Valid entries	Usage
0 to **10**	Enter the length of time (in hours) that the system will consider as being a long holding time. If you enter **0**, the system will not consider long holding calls.

Maintenance Tests

Appears when the **Group Type** field is **aplt**, **isdn**, **sip**, or **tie**.

Valid entries	Usage
y/n	Enter **y** if hourly maintenance tests will be made on this trunk group. One or more trunk members must be administered for this entry to be saved.

Measured

Indicates if the system will transmit data for this trunk group to the Call Management System (CMS). You cannot use **internal** and **both** unless either the **BCMS (Basic)** or the **VuStats** field is **y** on the System Parameters Customer-Options (Optional Features) screen. If the **ATM** field is set to **y** on the System Parameters Customer-Options (Optional Features) screen, this field accepts only **internal** or **none**. If this field contains a value other than **internal** or **none** when **ATM** is **y**, **none** appears.

Valid entries	Usage
internal	Enter **internal** if the data can be sent to the Basic Call Management System (BCMS), the VuStats data display, or both.
external	Enter **external** to send the data to the CMS.

Valid entries	Usage
both	Enter **both** to collect data internally and to send it to the CMS.
none	Enter **none** if trunk group measurement reports are not required.

MF Tariff Free

This field appears for Access, APLT, DID, DIOD, DMI-BOS, and Tandem trunk groups when the **Incoming Dial Type** field is **mf** or the **Group Type** field is **tie**, the **Trunk Signaling Type** field is blank, **cont**, or **dis**, and the **Incoming Dial Type** field is **mf**.

Valid entries	Usage
y/n	Enter **y** to make Communication Manager generate an MFC Tariff-Free Backward Signal (administered on the Multifrequency-Signaling-Related-System-Parameters screen) during call setup instead of the "free" signal. This aids CO billing.

Network Call Redirection

Valid entries	Usage
deflect	Use to allow Network Call Deflection.
ANSI-transfer	Use to allow Network Call Transfer for MCI DEX 600 ISDN trunks.
Nortel-transfer	Use to allow Network Call Transfer for MCI DMS 250 switches.
telcordia-tbct	Use to allow Network Call Transfer for Lucent 5ESS or Nortel DMS100 switches.

Outgoing ANI

If this trunk group is used for an outgoing call with ANI, the entry in this field overrides the normal ANI. The ANI is sent exactly as administered, except for the normal truncation to 7 digits for Russian ANI. This ANI override works both for calls originated in Communication Manager and calls tandemed through it. This field appears for CO, DIOD, FX, and WATS trunk groups.

Valid entries	Usage
1 to 15 digits	Enter the digit string to be sent in place of normal ANI.
blank	Leave this field blank to allow ANI to work normally.

Path Replacement Method

Appears when the following fields are set on the Trunk Group screen: trunk **Group Type** is **ISDN**, **Supplementary Service Protocol** is **b** or **e**, the **Path Replacement with Retention** is **n**, and the **Supplementary Services with Rerouting** field or the **DCS with Rerouting** field on the System Parameters Customer-Options (Optional Features) screen is **y**.

Valid entries	Usage
always	Use any QSIG (SSB) trunk group as the replacement trunk group. A new call is always originated, even when the original trunk group is determined to be the replacement trunk group.
BR (better route)	Route pattern preferences help determine trunk group path replacement. The original trunk group is retained if the **Path Replacement with Retention** field is **y**. Path replacement fails (and the original trunk group is retained) if the **Path Replacement with Retention** field is **n**.

Path Replacement with Retention

Appears when the following fields are set on the Trunk Group screen: trunk **Group Type** is **ISDN**, **Supplementary Service Protocol** is **b** or **e**, and the **Supplementary Services with Rerouting** field or the **DCS with Rerouting** field on the System Parameters Customer-Options (Optional Features) screen is **y**.

Valid entries	Usage
y/n	Enter **y** to retain the original trunk group. Set to **n** to allow path replacement according to setting on the **Path Replacement Method** field.

PBX ID

Appears when the **Used for DCS** field is **y**. This field identifies the remote switch in the network with which the trunk will communicate on a DCS signaling link.

Valid entries	Usage
1 to **63** or blank	Enter the ID of the switch at the other end of this trunk.

Per Call CPN Blocking Code

For Access, APLT, CO, DIOD, FX, tandem, tie, and WATS trunk groups only.

Valid entries	Usage
1 to 4 digit number or blank	
*****, **#**	Can be used as the first digit

Per Call CPN Unblocking Code

For access, APLT, CO, DIOD, FX, tandem, tie, and WATS trunk groups only.

Valid entries	Usage
1 to 4 digit number or blank	
*****, **#**	Can be used as the first digit

Precedence Incoming

The **Precedence Incoming** field defines whether the precedence level for dual-tone multifrequency (DTMF) or tone trunks is received as digits (rotary pulses) or as DTMF signals (touchtones). Appears when the **DSN Term** field is **y** and **Group Type** is **tie**.

Valid entries	Usage
digit	Precedence level is received as digits (rotary pulses).
dtmf (a-d)	Precedence level is received as DTMF signals (touchtones).

Precedence Outgoing

The **Precedence Outgoing** field defines whether the precedence level for dual-tone multifrequency (DTMF) or tone trunks is sent as digits (rotary pulses) or as DTMF signals (touchtones). Appears when the **DSN Term** field is **y** and **Group Type** is **tie**.

Valid entries	Usage
digit	Precedence level is sent as digits (rotary pulses).
dtmf (a-d)	Precedence level is sent as DTMF signals (touchtones).

R2 MFC Signaling

Appears when, on the **System Parameters Customer-Options (Optional Features) screen**, **Multinational Locations** is **y**, and on the Trunk Group screen, **Outgoing Dial Type** is **mf**. Also appears if, on the Trunk Group screen, **Incoming Dial Type** or **Outgoing Dial Type** is **rotary,** and **Country** is **15** (Russia).

Valid entries	Usage
1 to **8**	Enter the MFC signaling parameters set used by this trunk group.

Receive Analog Incoming Call ID

15 characters of name and number information associated with an incoming call on analog trunks (ICLID, or incoming call line identification information) is stored and displays. This field appears for CO, DID, and DIOD trunk groups when the **Analog Trunk Incoming Call ID** field on the **System Parameters Customer-Options (Optional Features)** screen is **y** and the **Direction** field is **incoming** or **two-way**.

Valid entries	Usage
Bellcore	Used to collect ICLID information in the U.S.
NTT	Used to collect ICLID information in Japan.
disabled	Stops the collection of ICLID information on analog trunks.
V23-Bell	Enter V23-Bell for Bellcore protocol with V.23 modem tones. Used in Bahrain and similar countries.

Replace Unavailable Numbers

Appears when the **Group Type** field is **isdn** or **sip**. Indicates whether to replace unavailable numbers with administrable strings for incoming and outgoing calls assigned to the specified trunk group. This field applies to BRI/PRI, H.323, and SIP Enablement Services (SES) trunks. This field also applies to analog trunks if, on the System Parameters Customer Options screen, **Analog Trunk Incoming Call ID** is **y**, and on the Trunk Group screen, **Receive Analog Incoming Call ID** is set to any value except **disabled**.

Valid entries	Usage
y/n	Enter **y** for the display to be replaced regardless of the service type of the trunk.

Request Category

This field appears when the **Country** field is **15** and the **Shuttle** field is **y**.

Valid entries	Usage
y/n	Enter **y** if Communication Manager should request a call category from the central office.

Seize When Maintenance Busy

This field only affects ports on TN760C (or later release), TN767, and TN464C (or later release) circuit packs. It indicates whether this server generates an outgoing seizure when a trunk in this trunk group is maintenance busied and whether the far-end server or switch is administered to do likewise. This supports the Electronic Tandem Network Busyout feature, which is intended to prevent a far-end server or switch from reporting problems with a trunk that has been removed from service on your end. This field's setting does not affect the behavior of the far-end server or switch; it controls the behavior of your server and defines the expected far-end behavior.

For DIOD trunks using TN464F (or later release) or TN2464, displays only when the **Group Type** field is **diod** and the **Trunk Signaling Type** field is **pulsed**, **cont**, or **dis**.

Valid entries	Usage
near-end	Enter **near-end** if Communication Manager generates an outgoing seizure when a trunk is maintenance busied, but the far-end server or switch does not. The seizure is maintained until the maintenance busyout is released.
far-end	Enter **far-end** if the far-end server or switch generates an outgoing seizure when a trunk is maintenance busied, but this server running Communication Manager does not.
both-ends	Enter **both-ends** if both this server running Communication Manager and the far-end server or switch generate an outgoing seizure when a trunk is maintenance busied.

If a server generates an outgoing seizure when a trunk is busied out, the seizure will probably cause alarms at the far-end server or switch, perhaps leading to a far-end maintenance busy out, unless the far-end server or switch is administered to expect this behavior.

If the administered value of this field is either **far-end** or **both-ends**, any abnormally long incoming seizure (including failure to drop from a completed call) is assumed to be the result of a far-end maintenance busy condition. Note that this assumption might be incorrect, since the abnormally long seizure might actually be due to failure of the trunk circuit. Regardless of the cause of the abnormally long seizure, your server running Communication Manager does the following things:

1. Generates a warning alarm indicating that the trunk is assumed to be maintenance busy at the far-end
2. Removes the trunk from service
3. Keeps the trunk out of service until a far-end disconnect is received

Allowable values depend on the entry in the **Direction** field: check the online help in Communication Manager.

Send Calling Number

Specifies whether the calling party's number is sent on outgoing or tandemed ISDN calls. Valid entries are **y**, **n**, or **r** (restricted). If **y** is entered, the ISDN Numbering - Public/Unknown Format screen is accessed to construct the actual number to be sent, or the ISDN Numbering - Private screen (based on the **Numbering Format** field) is used. If the value is **r**, the calling number is sent "presentation restricted."

Note:
The ISDN Numbering - Public/Unknown Format screen can override the **Send Calling Number** field entry for any administrable block of extensions.

Send Called/Busy/Connected Number

Appears if the **QSIG Value-Added** field on the Trunk Group screen is **y**. Specifies if the dialed number, whether called (ringing), busy (busy tone), or connected (answered) is sent on incoming or tandemed ISDN calls. Valid entries are **y**, **n**, or **r** (restricted). If **y** is entered, the ISDN Numbering - Public/Unknown Format screen is accessed to construct the actual number sent, or the ISDN Numbering - Private screen (based on the **Numbering Format** field) is used. If the value is **r**, the connected number is sent "presentation restricted." The **Send Called/Busy/Connected Number** field must be set to **y** in order for the Calling Party Number of an incoming ISDN call to display at the transferred-to station after a QSIG transfer operation.

Send Connected Number

Appears if the **QSIG Value-Added** field on the Trunk Group screen is **n**. Specifies if the connected party's number is sent on incoming or tandemed ISDN calls. Valid entries are **y**, **n**, or **r** (restricted). If **y** is entered, the ISDN Numbering - Public/Unknown Format screen is accessed to construct the actual number sent, or the ISDN Numbering - Private screen (based on the **Numbering Format** field) is used. If the value is **r**, the connected number is sent "presentation restricted." The **Send Connected Number** field must be set to **y** in order for the Calling Party Number of an incoming ISDN call to display at the transferred-to station after a QSIG transfer operation.

Note:
The AT&T Switched Network Protocol does not support restricted displays of connected numbers. Therefore, if you administer the 1a country-protocol/ protocol-version combination on the DS1 Circuit Pack screen, you should not administer the **Send Connected Number** field to **r** (restricted) on the ISDN Trunk Group screen, as this causes display problems.

The ISDN Numbering - Public/Unknown Format screen overrides the **Send Connected Number** field entry for any administrable block of extensions.

Send EMU Visitor CPN

Use this field to control which calling party identification (extension of the primary telephone or extension of the visited telephone) is used when a call is made from a visited telephone. If you want to use the calling party information of the primary telephone, set this field to **n**.

There are areas where public network trunks disallow a call if the calling party information is invalid. In this case, there can be instances where the extension of the primary telephone is considered invalid and the extension of the visited telephone must be used. To use the extension of the visited telephone, set the **Send EMU Visitor CPN** field to **y**. For more information on Enterprise user Mobility, see *Administering Avaya Aura™ Communication Manager*, 03-300509.

Valid entries	Usage
y	Sends calling party identification information on the extension of the EMU user's telephone.
n	Sends calling party identification information on the primary telephone.

Send Name

Appears only when the **Group Type** field is **isdn** or **sip**. Specifies whether the calling/connected/called/busy party's administered name is sent to the network on outgoing/incoming calls. Valid entries are **y**, **n**, or **r** (restricted). The value **r** indicates that the calling/connected name will be sent by Communication Manager, but will be marked “presentation restricted.” This value is valid only if the **Supplementary Service Protocol** field is **a** (national supplementary service), **b** (for called/busy only) or **d** for the QSIG Global Networking Supplementary Service Protocol. When the **Supplementary Service Protocol** field is **e** (DCS with Rerouting), only values of **y** and **n** are permitted.

Note:
If name information is not administered for the calling station or the connected/called/busy station, the system sends the extension number in place of the name.

Short Holding Threshold

This field appears when the **ACA Assignment** field is **y**.

Valid entries	Usage
0 to **30**	Enter the number of times the system will record a short holding call before alerting an attendant to the possibility of a faulty trunk. Enter **0** for no short holding calls.

Short Holding Time (seconds)

This field appears when the **ACA Assignment** field is **y**.

Valid entries	Usage
0 to **160**	Enter the length of time (in seconds) that the system considers as being a short holding time. If **0** is entered, the system will not consider short holding calls.

Show ANSWERED BY on Display

This field appears when the **Group Type** field is **isdn pri/bri** or **sip**. Use this field to administer whether or not the words "ANSWERED BY" are displayed in addition to the connected telephone number on calls over this trunk.

Note:
Based on display language settings for stations, "ANSWERED BY" is translated into and displayed in the appropriate language.

Valid entries	Usage
y	When set to **y**, the words "ANSWERED BY" are displayed in addition to the connected telephone number. This is the default.
n	When set to **n**, only the connected telephone number is displayed. This might be preferred when outgoing calls are over a trunk that might be redirected.

Shuttle

This field appears when the **Group Type** field is **co**, **fx**, or **wats**, the **Country** field is **15**, and the **Outgoing Dial Type** field is **rotary**. It can be administered on TN464D (or later release) or TN2199 circuit packs.

Valid entries	Usage
y/n	Enter **y** to enable MF shuttle signaling.

Signaling Group

This field displays the group number assigned when the group was added.

Start B Signal

This field appears when the **Country** field is **15** and the **Shuttle** field is **y**. Enter **1** to **3** to indicate which B-signal should be used to start a call. The value administered in this field must be coordinated with your central office. See Start Position on page 1030.

Valid entries	Usage
1	Start calls with signal B1 (first digit)
2	Start calls with signal B2 (next digit)
3	Start calls with signal B3 (previous digit)

Start Position

The value administered in this field must be coordinated with your central office. This field appears when the **Country** field is **15** and the **Shuttle** field is **y**.

Valid entries	Usage
1 to **9**	Indicate which digit in the digit string is considered to be the “previously sent” digit (see Start B Signal on page 1029).

Suppress # Outpulsing

Valid entries	Usage
y/n	Enter **y** if end-to-end signaling begins with (and includes) "#". The final "#" is suppressed in cases where the system would normally outpulse it. This field should be **y** when the Central Office (for example, rotary) or any other facility treats "#" as an error.

Time (sec) to Drop Call on No Answer

This field appears if the **Group Type** field is **co** or **diod** and the **Outgoing Dial Type** field is **mf**, or if the **Group Type** field is **co**, **diod**, **fx**, or **wats** and the **Country** field is **15**.

Valid entries	Usage
0 to **1200**	Enter the duration (in seconds) Communication Manager should wait for outgoing calls to be answered. If the call is not answered in the specified number of seconds, the call drops. If this field is **0**, the timer is not set and no calls drop.

Used for DCS

Valid entries	Usage
y/n	Enter **y** if this trunk group will send and receive messages on a DCS signaling link.

Note:
This field cannot be activated if the trunk group number is greater than 255 or if the Trunk Access code is more than 3-digits long.

If this field is **y**, you can administer ISDN-BRI trunk groups unless the **DCS Signaling** field is **d-chan**.

Used Only for Paging

This field appears when the **Group Type** field on the Trunk Group screen is **wats**, and the **Port Network Support** field on the System Parameters Customer Options screen is **n**.

Valid entries	Usage
y/n	Enter **y** to designate this trunk for paging use. Default is **n**.

Voice Paging Timeout (sec)

This field appears when **Used Only for Paging** is **y**.

Valid entries	Usage
10 to **600**	Enter the number of seconds before a paged trunk call drops. Default is **10**.

Field descriptions for Administrable Timers page

This screen might not appear for all trunk group types. The figure below is only an example, and is intended to show most of the fields that might appear on this page of the Trunk Group screen. This example might not show all fields, or might show fields that normally do not appear together. Your own screen might vary from this example according to specific field and system settings. The list of field descriptions that follows the figure is in alphabetical order for quick reference. This list is intended to be comprehensive, and might include information on fields that are not shown in the example. The field descriptions identify fields that are specific to particular trunk group types.

Note:
This section does not cover ISDN trunks. For descriptions of the screens and fields that are unique to ISDN trunks, see ISDN Trunk Group on page 539.

Figure 341: Administrable Timers for Trunk Group screen

```
add trunk-group next                                               Page 3 of x
                                  ADMINISTRABLE TIMERS
                    Send Incoming/Outgoing Disconnect Timers to TN465 Ports? _
                                                 Outgoing Dial Guard(msec): _____
  Incoming Glare Guard(msec): _____             Outgoing Glare Guard(msec): _____

                                      Outgoing Rotary Dial Interdigit (msec): ____
       Ringing Monitor(msec): _____                  Incoming Seizure(msec): ____
   Outgoing End of Dial(sec): ___          Outgoing Seizure Response(sec): ___
 Programmed Dial Pause(msec): _____            Disconnect Signal Error(sec): ___
          Flash Length(msec): ____
                                                        Busy Tone Disconnect?

END TO END SIGNALING
    Tone (msec): ____      Pause (msec): 150

OUTPULSING INFORMATION
     PPS: 10    Make(msec): 40    Break(msec): 60  PPM? y  Frequency: 50/12k
```

CAUTION:
Customers: Do not change fields on this page without assistance from Avaya or your network service provider.

Answer Send (msec)

This field appears only if the **Trunk Type** field is **dis** and the trunk group **Direction** field is **incoming** or **two-way**. Only TN2140 and TN2199 ports receive this timer.

Valid entries	Usage
10 to **2550** in increments of 10	Enter the duration of the answer signal pulse.

Busy Tone Disconnect

The field appears when **Enable Busy Tone Disconnect for Analog loop-start Trunks** is **y** on the System Parameters Country-Options screen.

Valid entries	Usage
y/n	Enter **y** to allow Communication Manager to recognize a busy tone signal as a disconnect on this trunk group.

Cama Outgoing Dial Guard (msec)

This field appears when **Group Type** is **cama** (the trunk group type used for emergency 911 service).

Valid entries	Usage
25 to **6375** in increments of 25	Enter the minimum interval between the seizure acknowledgment on the receiving server or switch and the outpulsing of digits by this server.

Cama Wink Start Time (msec)

This field appears when **Group Type** is **cama**.

Valid entries	Usage
20 to **5100** in increments of 20	Specifies the duration (the wait-for-wink-to-end time) for a wink-start CAMA trunk. The wink must begin before the Outgoing Seizure Response timer expires.

Disconnect Signal Error (sec)

This field appears for ground-start trunk groups.

Valid entries	Usage
1 to **255** in increments of 1	Enter the maximum interval that Communication Manager will wait to receive a disconnect signal from the far-end after the local party (a telephone or tie trunk) goes on-hook. If the timer expires, Communication Manager assumes a disconnect failure and takes appropriate action, such as creating an error message.

Flash Length (msec)

This timer is sent to TN436B, TN459B, TN464C (or later), TN465B (or later), (TN753 if Country is **23**), TN2146, TN2147, TN2184, and TN2199 circuit boards.

Valid entries	Usage
10 to **2550** in increments of 10	Enter the duration of a flash signal generated toward the central office.

Glare

This field is only administrable if the **Trunk Type** field is **cont** and the trunk group **Direction** field is **two-way** or **outgoing**. Only TN2140 ports receive this timer.

Valid entries	Usage
40 to **100** in increments of 10	Enter the minimum acceptable interval (in msec) between the moment your server running Communication Manager sends an outgoing seizure and the moment it receives a seizure acknowledgment. If acknowledgment is received before the timer expires, glare is assumed.

Incoming Dial Guard (msec)

Valid entries	Usage
10 to **2550** in increments of 10	Enter the minimum acceptable interval between the detection of an incoming seizure and the acceptance of the first digit. Communication Manager will not accept digits before this timer expires. This timer is never sent to TN429 ports.

Incoming Disconnect (msec)

The field appears only when the **Direction** field is **incoming** or **two-way** and the **Trunk Type** field is either blank or **cont**.

Valid entries	Usage
50 to **2550** in increments of 10	Enter the minimum valid duration of a disconnect signal for an incoming call. Communication Manager will not recognize shorter disconnect signals. This field cannot be blank. For Brazil pulsed E&M signaling, use **600**.

Incoming Disconnect Send (msec)

This field is only administrable if the **Trunk Type** field is **dis** and the trunk group **Direction** field is **incoming** or **two-way**. Only TN2140 ports receive this timer.

Valid entries	Usage
500 to **1200** in increments of 100	Enter the duration of the backward release signal your server running Communication Manager sends at the end of an incoming call.

Incoming Glare Guard (msec)

This field only appears when the trunk group **Direction** field is **two-way**.

Valid entries	Usage
100 to **25500** in increments of 100	Enter the minimum interval that must elapse between a trunk's release from an incoming call and its seizure for an outgoing call. This field cannot be blank. This delay gives the far-end time to release all equipment after the trunk is released.

Incoming Incomplete Dial Alarm (sec)

Only the TN436 (all), TN459 (all), TN464C (or later), TN767, TN2140, TN2146, TN2184, TN2199, and TN2242 circuit packs use this timer.

Valid entries	Usage
1 to **255** in increments of 1	Enter the maximum acceptable interval between an incoming seizure and receipt of all digits. Intervals greater than this limit generate an inline error.

Incoming Partial Dial (sec)

This timer appears only if the **Incoming Dial Type** field is **rotary**.

Valid entries	Usage
5 to **255** in increments of 1	Enter the maximum time allowed between incoming rotary digits.

Note:
This timer is never sent to TN429 ports.

Incoming Seizure (msec)

This field appears when the **Direction** field is **incoming** or **two-way**, and, when applicable, the **Trunk Type** field is **cont**. Only TN429, TN438 (any release), TN 447, TN464C (or later), TN465 (any release), TN767, TN2138, TN2140, TN2147, TN2184, and TN2199 ports receive this timer. For DID trunks, only TN2199 and TN429D (or later) receive this timer.

Valid entries	Usage
20 to **2550** in increments of 10	Enter the duration of the shortest incoming seizure signal your server running Communication Manager can recognize. For ICLID, set this field to 120. The field cannot be blank.

Normal Outgoing Seize Send (msec)

This field appears only if the **Trunk Type** field is **dis** and the trunk group **Direction** field is **two-way** or **outgoing**. Only TN2140 ports receive this timer.

Valid entries	Usage
10 to **990** in increments of 10	Enter the duration of the signal your server running Communication Manager sends for an outgoing seizure.

Outgoing Dial Guard (msec)

Valid entries	Usage
100 to **25500** in increments of 100	Enter the minimum interval between seizure acknowledgment of a trunk and the outpulsing of digits. This field cannot be blank. For trunks that do not provide seizure acknowledgment, the timer specifies the minimum time between seizure and the outpulsing of digits. Any digit the caller dials after they lift the receiver, but before the timer expires, is not outpulsed until the timer expires.

Outgoing Disconnect (msec)

Valid entries	Usage
50 to **2550** in increments of 10	Enter the minimum valid duration of a disconnect signal for an outgoing call. Communication Manager will not recognize shorter disconnect signals. This field cannot be blank. This timer begins timing when a disconnect signal is detected on an outgoing call and resets when the signal is no longer detected. If the timer expires, the trunk drops. For Brazil pulsed E&M signaling, use **600**.

Outgoing Disconnect Send (msec)

This field is administrable only if the **Trunk Type** field is **dis** and the trunk group **Direction** field is **two-way** or **outgoing**. Only TN2140 ports receive this timer.

Valid entries	Usage
100 to **9900** in increments of 100	Enter the duration of the forward release signal your server running Communication Manager sends at the end of outgoing calls.

Outgoing End of Dial (sec)

This field controls firmware answer supervision timers on circuit packs that have them. It appears when the **Direction** field is **outgoing** or **two-way** and the **Receive Answer Supervision** field is **n**. If the **Receive Answer Supervision** field is **y**, this field does not appear and the firmware timer on the appropriate circuit pack is automatically disabled.

Note:
The **Answer Supervision Timeout** field on the Trunk Group screen provides timed answer supervision for circuit packs without administrable timers. Since trunk groups might contain ports on more than one circuit pack, it's possible you might need to use both timers with the same trunk group. If so, set the **Outgoing End of Dial** field and the **Answer Supervision Timeout** field to the same value.

During a cut-through operation, timing begins after Communication Manager sends each outgoing digit and ceases when answer supervision is received. If the timer expires, Communication Manager acts as if it has received answer supervision. On senderized operation, the timer begins after the switch sends the last digit collected. The timer ceases when answer supervision is received. If the timer expires, Communication Manager acts as if it has received answer supervision.

Valid entries	Usage
1 to **254** in increments of 1	Enter the maximum time, in seconds, that Communication Manager will wait to receive answer supervision for outgoing calls on the ports controlled by firmware timers. For Brazil pulsed E&M signaling, use **40**.

Outgoing Glare Guard (msec)

This field only appears for **outgoing** and **two-way** trunk groups.

Valid entries	Usage
100 to **25500** in increments of 100	Enter the minimum interval that must elapse between a trunk's release from an outgoing call and its seizure for another outgoing call. This field cannot be blank. This delay gives the far-end time to release all equipment after the outgoing trunk is released.

Outgoing Last Digit (sec)

This field is only administrable if the **Trunk Type** field is **dis** or **cont** and the trunk group **Direction** field is **two-way** or **outgoing**. Only TN497 and TN2140 ports receive this timer.

Valid entries	Usage
1 to **40** in increments of 1	Enter the maximum time that Communication Manager will wait for the next digit dialed. After the timer expires, no more digits are accepted by the circuit pack.

Outgoing Rotary Dial Interdigit (msec)

This field only appears when:

1. the trunk group **Group Type** field is **access**, **aplt**, **co**, **diod**, **dmi-bos**, **fx**, **rlt**, **tandem**, or **wats** and the **Outgoing Dial Type** field is **rotary**.
2. the **Group Type** field is **tie**, the **Trunk Type** field is blank, **cont**, or **dis**, and the **Outgoing Dial Type** field is **rotary**.
3. the **Group Type** field is **tie**, and the **Trunk Type** field is **tge**, **tgi**, or **tru** (the **Outgoing Dial Type** field does not appear but is implied to be **rotary**).

Valid entries	Usage
150 to **2550** in increments of 10	Enter the minimum time between outpulsed digits on outgoing rotary trunks.

Outgoing Seizure (msec)

Appears when the **Country** field is **15**, the **Direction** field is **outgoing** or **two-way**, and the **Trunk Type** field is **2-wire-ac**, **2-wire-dc**, or **3-wire**. This timer is sent only to the TN2199 circuit pack.

Valid entries	Usage
20 to **2550** in increments of 10	Enter the duration of the outgoing seizure signal.

Outgoing Seizure Response (sec)

This timer is sent to the TN438B, TN439, TN447, TN458, TN464B (or later), TN465B (or later), TN767, TN2140, TN2147, TN2184, TN2199, and TN2242 circuit packs.

Valid entries	Usage
1 to **255** in increments of 1	Enter the maximum interval that Communication Manager should wait after sending a seizure signal to receive seizure acknowledgment from the far-end. If the acknowledgment is not received in this time, a seizure failure response is uplinked. For Brazil pulsed E&M signaling, use **255**.

Programmed Dial Pause (msec)

This timer is administrable for all outgoing and two-way trunk groups. This timer works with the TN464B (or later), TN767, TN458, TN2140, and TN2242 tie circuit packs. All CO circuit packs that accept administrable timers accept this timer.

Valid entries	Usage
100 to **25500** in increments of 100	Set the exact duration of pauses used during abbreviated dialing, ARS outpulsing, and terminal dialing operations.

Release Ack Send (msec)

After your server running Communication Manager receives a forward release signal, it must send a forward release acknowledgment signal. This field appears only if the **Trunk Type** field is **dis** and the trunk group **Direction** field is **incoming** or **two-way**. Only TN2140 ports receive this timer.

Valid entries	Usage
500 to **1200** in increments of 100	Enter the duration of the signal your server running Communication Manager sends for a forward release acknowledgment.

Ringing Monitor (msec)

This timer is sent to TN464C (or later), TN767, TN438 (all), TN447, TN465 (all), TN2138, TN2147, TN2184, and TN2199 CO circuit packs.

Valid entries	Usage
200 to **51000** in increments of 200	Enter the minimum time Communication Manager requires to determine if a trunk disconnects. The field cannot be blank. If the ringing signal disappears for a duration longer than the time specified in this field, Communication Manager assumes the call has been disconnected.

Seize Ack Delay (msec)

This field appears only if the **Trunk Type** field is **dis** and the trunk group **Direction** field is **incoming** or **two-way**. Only TN2140 ports receive this timer.

Valid entries	Usage
40 to **120** in increments of 10	Enter the maximum interval your server running Communication Manager will wait after receipt of an incoming seizure to send seizure acknowledgment.

Seize Ack Send (msec)

This field appears only if the **Trunk Type** field is **dis** and the trunk group **Direction** field is **incoming** or **two-way**. Only TN2140 ports receive this timer.

Valid entries	Usage
10 to **990** in increments of 10	Enter the duration of the seizure acknowledgment signal your server running Communication Manager sends in response to an incoming seizure.

Send Incoming/Outgoing Disconnect Timers to TN465 Ports

The field appears only for a co, fx, or wats trunk group.

Valid entries	Usage
y/n	Enter **y** if you want to send the incoming disconnect and outgoing disconnect timer values to the trunk group ports that are on a TN465 board.

END TO END SIGNALING

Pause (msec)

THis field is administrable only if the **Trunk Type** field is blank. All CO, DIOD, and tie circuit packs that accept administrable timers accept this timer. However, this timer is sent only to the following circuit packs: TN464B (or later), TN767, TN436B, TN459B, TN2146, TN2199, and TN2242, and TN429 and TN2184 ports in a DID trunk group.

Valid entries	Usage
20 to **2550** in increments of 10	Enter the minimum acceptable interval (pause) between DTMF tones sent from a hybrid telephone.

Tone (msec)

This field appears only if the **Trunk Type** field is blank. All CO, DIOD, and Tie circuit packs that accept administrable timers accept this timer. This timer is also sent to the following circuit packs: TN464B (or later), TN767, TN436B, TN459B, TN2146, TN2199, TN429, TN2184 ports in a DID trunk group.

Valid entries	Usage
20 to **2550** in increments of 10	Enter the duration of a DTMF tone sent when a button on a hybrid telephone is pressed.

OUTPULSING INFORMATION

Break (msec)

Valid entries	Usage
Enter the duration of the break interval (the pulse duration) while the system is outpulsing digits using dial pulse signaling. The field cannot be blank.	
20 to **80** in increments of 5	If PPS field is **10**, the sum of the **Make (msec)** and **Break (msec)** fields must equal 100.
10 to **40** in increments of 5.	If the **PPS** field is **20**, the sum of the **Make (msec)** and **Break (msec)** fields must equal 50.

Frequency

This field identifies the PPM pulse frequency, or frequencies, sent by the public network. It appears if the **Direction** field is **outgoing** or **two-way** and **PPM** is **y**. Circuit packs can detect up to three different frequencies, (12kHz, 16kHz, and 50Hz), plus two frequency combinations, (50Hz/12kHz and 50Hz/16kHz). This field controls TN465B, TN2138, and TN2184 circuit packs.

Valid entries	Usage
12k	The TN465B (or later) and TN2184 can only detect 12k and 16kHz PPM. Therefore, if **12k** is administered, the circuit pack will be set to detect 12kHz.
16k	The TN465B (or later) and TN2184 can only detect 12k and 16kHz PPM. Therefore, if **16k** is administered, the circuit pack will be set to detect 16kHz.
	1 of 2

Valid entries	Usage
50	The TN465B (or later) and TN2184 can only detect 12k and 16kHz PPM. Therefore, if **50** is administered, the circuit pack will be set to detect 16kHz.
50/12k	The TN465B (or later) and TN2184 can only detect 12k and 16kHz PPM. Therefore, if **50/12k** is administered, the circuit pack will be set to detect 12kHz.
50/16k	The TN465B (or later) and TN2184 can only detect 12k and 16kHz PPM. Therefore, if **50/16k** is administered, the circuit pack will be set to detect 16kHz.
	2 of 2

Make (msec)

Valid entries	Usage
Enter the duration of the make interval (the pause between pulses) while the system is outpulsing digits using dial pulse signaling. The field cannot be blank.	
20 to **80** in increments of 5	If the **PPS** field is **10**, the sum of the **Make (msec)** and **Break (msec)** fields must equal 100.
10 to **40** in increments of 5	If the **PPS** field is **20**, the sum of the **Make (msec)** and **Break (msec)** fields must equal 50.

PPM

For CO, DIOD, FX, PCOL, and WATS trunks. This field appears when the **Direction** field is **outgoing** or **two-way**.

Valid entries	Usage
y/n	Enter **y** if Periodical Pulse Metering (PPM) pulses should be collected from the public network to determine call cost. If this field is **y**, the **Frequency** field appears.

PPS

Valid entries	Usage
10 **20**	Enter the rate (pulses per second) at which outgoing rotary pulses are sent over this trunk group. **Note:** The TN439, TN458, TN497, TN747Bv12 (or later), and TN767 circuit packs send rotary pulses at 10 pps only.

Field descriptions for ATMS Thresholds page

This screen appears when the **Direction** field on page 1 is **outgoing** or **two-way** and the **ATMS** field is **y** on the Feature-Related System Parameters screen.

The figure below shows a common configuration for the ATMS Thresholds page of the Trunk Group screen. This screen is only an example, and the fields shown below might change or disappear according to specific field settings.

Note:
This section does not cover ISDN trunks. For descriptions of the screens and fields that are unique to ISDN trunks, see ISDN Trunk Group on page 539.

Figure 342: ATMS Thresholds screen

```
add trunk-group next                                            Page 4 of x
                              ATMS THRESHOLDS
TTL Type: _________                  Far End Test No: __________________
  TTL Vendor: ______________             TTL Contact: ______________________
Trunk Vendor: ____________             Trunk Contact: ___________________
Trunk Length: _____
                         MARGINAL          UNACCEPTABLE
                         Min   Max            Min   Max
           1004 Hz Loss: __     __            __    __

                         -Dev  +Dev           -Dev   +Dev
            404 Hz Loss:   _     _              _      _
           2804 Hz Loss:   _     _              _      _

Maximum C Message Noise:     __                   __
Maximum C Notched Noise:     __                   __
         Minimum SRL-HI:     __                   __
         Minimum SRL-LO:     __                   __
            Minimum ERL:     __                   __

      Allow ATMS Busyout, Error Logging and Alarming? _
  Maximum Percentage of Trunks Which Can Be Removed from Service by ATMS: __
```

CAUTION:
Customers: Do not change fields on this page without assistance from Avaya or your network service provider.

Far-End Test No.

Valid entries	Usage
1 to 16 digits	Enter the access number dialed to reach the terminating test line (TTL).

Trunk Contact

Valid entries	Usage
0 to 25 alphanumeric characters	Enter the name and/or telephone number of someone from the trunk vendor who can be contacted in the event of problems with the trunks.

Trunk Length

This field is not required. Since noise on a trunk increases with the length of the trunk, however, this information might be useful,

Valid entries	Usage
Use this field to record the length of the trunk group in kilometers or miles.	
0 to 4 digits followed by **k**	Shows the length in kilometers.
0 to 4 digits followed **m**	Shows the length in miles.

Trunk Vendor

Valid entries	Usage
0 to 22 alphanumeric characters	Enter the name of the vendor providing service over this trunk group (the company to notify in the event of problems with the trunks in this trunk group).

TTL Contact

Valid entries	Usage
0 to 25 alphanumeric characters	Enter the name and/or telephone number of someone from the TTL vendor who can be contacted in the event of problems with the terminating test line.

TTL Type

Specifies the type of terminating test line (TTL) selected for testing trunks. The TTL type determines what ATMS tests can be completed and thus which threshold values need to be administered.

Valid entries	Usage
105-w-rl	105 with return loss
105-wo-rl	105 without return loss
high-lts	high-level tone source
low-lts	low-level tone source

Valid entries	Usage
100	100 type
102	102 type

The following table explains the differences between types of terminating test lines:

Type TTL	**Description**	**Example**
105-w-rl	Full range of 18 measurements or some defaults for return loss used (56A)	TN771B, ZLC12 and SN261B circuit packs and new 56A mini-responder
105-wo-rl	Cannot return default values for far-end return loss	Older 56A mini-responder
high-level-tone	Sends a fixed sequence of tones at 0 dBm	ZLC12 and SN261B circuit packs
low-level-tone	Sends a fixed sequence of tones at -16dBm	SN261B circuit pack
100	Up to 5 measurements that sends a 1004 Hz tone then a quiet termination	
102	One measurement that sends a 1004 Hz tone	

The far-end server or switch containing the TTL might be any of the following:

- System 85 R2 switch, equipped with the Maintenance/Test Board (TN771B)
- System 75 R1V2 and beyond, all of which contain the circuitry required to perform the TTL function
- System 85 R2 switch, equipped with the Analog/Digital Facility Test Circuit (ADFTC, SN261)
- DIMENSION FP8, equipped with the Analog Facility Test Circuit (AFTC, ZLC-12)
- Central Office switches, equipped with various TTL equipment that provide 100, 102, or 105 test line capabilities (56A)

Other vendors' switches might be supported if compatible test lines are provided by these switches.

Four different versions of the ATMS Threshold Administration page can appear depending upon the measurements allowed by the TTL type selected. The four possibilities are:

1. 105-w-rl and 105-wo-rl - All thresholds appear.
2. high-lts and low-lts - All thresholds (except maximum C-notched noise) appear.

3. 100 - All thresholds (except maximum c-notched noise, 404Hz loss, and 2804 Hz loss) appear.
4. 102 - Only 1004 Hz loss threshold appears.

TTL Vendor

Valid entries	Usage
0 to 22 alphanumeric characters	Enter the name of the vendor supplying the terminating test line (TTL).

MARGINAL / UNACCEPTABLE

Allow ATMS Busyout, Error Logging and Alarming

Valid entries	Usage
y/n	Enter **y** to allow ATMS error logging and alarming (subject to filtering depending on the service organization used to deal with alarms).

Marginal Threshold - -Dev - 404 Hz Loss

Valid entries	Usage
0 to **9**	Enter the maximum negative deviation of measured loss at 404 Hz from the 1004 Hz test tone noise level (in dB) allowed before reporting a trunk as out of tolerance. Smaller dB values are more restrictive.

Marginal Threshold - +Dev - 404 Hz Loss

Valid entries	Usage
0 to **9**	Enter the maximum positive deviation of measured loss at 404 Hz from the 1004 Hz test tone loss level (in dB) allowed before reporting a trunk as out of tolerance. Smaller dB values are more restrictive.

Marginal Threshold - -Dev - 2804 Hz

Valid entries	Usage
0 to **9**	Enter the maximum negative deviation of measured loss at 2804 Hz from the 1004 Hz test tone loss level (in dB) allowed before reporting a trunk as out of tolerance. Smaller dB values are more restrictive.

Marginal Threshold - +Dev - 2804 Hz

Valid entries	Usage
0 to **9**	Enter the maximum positive deviation of measured loss at 2804 Hz from the 1004 Hz test tone loss level (in dB) allowed before reporting a trunk as out of tolerance. Smaller dB values are more restrictive.

Marginal Threshold - Max - 1004 Hz Loss

Valid entries	Usage
0 to **21**	Enter the maximum signal loss allowed for a 1004 Hz test tone (in dB) before a trunk is reported as out of tolerance. A smaller dB value is more restrictive.

Marginal Threshold - Min -1004 Hz Loss

Valid entries	Usage
-2 to **21**	Enter the minimum signal loss allowed for a 1004 Hz test tone (in dB) before a trunk is reported as out of tolerance. A larger dB value is more restrictive.

Marginal Threshold - Minimum ERL

Valid entries	Usage
0 to **40**	Enter the minimum low-frequency echo return loss in dB allowed before reporting a trunk as out of tolerance. Larger values are more restrictive.

Marginal Threshold - Maximum C Message Noise

Valid entries	Usage
15 to **55**	Enter the maximum C-message noise telephone as measured within the voice band frequency range (500 to 2500 Hz) allowed before reporting a trunk as out of tolerance. Smaller values are more restrictive.

Marginal Threshold - Maximum C Notched Noise

Valid entries	Usage
34 to **74**	Enter the maximum C-notched signal dependent noise interference in dBmC allowed before reporting a trunk as out of tolerance. Smaller values are more restrictive.

Marginal Threshold - Minimum SRL-HI

Valid entries	Usage
0 to **40**	Enter the minimum high-frequency signaling return loss in dB allowed before reporting a trunk as out of tolerance. Larger values are more restrictive.

Marginal Threshold - Minimum SRL-LO

Valid entries	Usage
0 to **40**	Enter the minimum low-frequency signaling return loss in dB allowed before reporting a trunk as out of tolerance. Larger values are more restrictive.

Maximum Percentage of Trunks Which Can Be Removed From Service by ATMS

Appears when the **Allow ATMS Busyout**, **Error Logging and Alarming** field is **y**.

Valid entries	Usage
0, **25**, **50**, **75**, **100**	Enter the highest percentage of trunks from the trunk group that can be removed from service at one time because of unacceptable transmission measurement results.

Unacceptable Threshold - -Dev - 404 Hz

Valid entries	Usage
0 to **9**	Enter the maximum negative deviation of measured loss at 404 Hz from the 1004 Hz test tone loss level (in dB) allowed before reporting a trunk as unacceptable. Smaller dB values are more restrictive.

Unacceptable Threshold - +Dev - 404 Hz

Valid entries	Usage
0 to **9**	Enter the maximum positive deviation of measured loss at 404 Hz from the 1004 Hz test tone loss level (in dB) allowed before reporting a trunk as unacceptable. Smaller dB values are more restrictive.

Unacceptable Threshold - -Dev - 2804 Hz

Valid entries	Usage
0 to **9**	Enter the maximum negative deviation of measured loss at 2804 Hz from the 1004 Hz test tone loss level (in dB) allowed before reporting a trunk as unacceptable. Smaller dB values are more restrictive.

Unacceptable Threshold - +Dev - 2804 Hz

Valid entries	Usage
0 to **9**	Enter the maximum positive deviation of measured loss at 2804 Hz from the 1004 Hz test tone loss level (in dB) allowed before reporting a trunk as unacceptable. Smaller dB values are more restrictive.

Unacceptable Threshold - Max - 1004 Hz Loss

Valid entries	Usage
0 to **21**	Enter the maximum signal loss allowed for a 1004 Hz test tone (in dB) before a trunk is reported as unacceptable. A smaller dB value is more restrictive.

Unacceptable Threshold - Min - 1004 Hz Loss

Valid entries	Usage
-2 to **21**	Enter the minimum signal loss allowed for a 1004 Hz test tone (in dB) before a trunk is reported as unacceptable. A larger dB value is more restrictive.

Unacceptable Threshold - Maximum C Message Noise

Valid entries	Usage
15 to **55**	Enter the maximum C-message noise interference in dBmC above reference noise terminating on a telephone as measured within the voice band frequency range (500 to 2500 Hz) allowed before reporting a trunk as unacceptable. Smaller values are more restrictive.

Unacceptable Threshold - Maximum C Notched Noise

Valid entries	Usage
34 to **74**	Enter the maximum C-notched signal dependent noise interference in dBmC allowed before reporting a trunk as unacceptable. Smaller values are more restrictive.

Unacceptable Threshold - Minimum ERL

Valid entries	Usage
0 to **40**	Enter the minimum low-frequency echo return loss in dB allowed before reporting a trunk as unacceptable. Larger values are more restrictive.

Unacceptable Threshold - Minimum SRL-HI

Valid entries	Usage
0 to **40**	Enter the minimum high-frequency signaling return loss in dB allowed before reporting a trunk as unacceptable. Larger values are more restrictive.

Unacceptable Threshold - Minimum SRL-LO

Valid entries	Usage
0 to **40**	Enter the minimum low-frequency signaling return loss in dB allowed before reporting a trunk as unacceptable. Larger values are more restrictive.

Field descriptions for Protocol Variations page

This screen appears only when the **Group Type** is **sip**.

Figure 343: Protocol Variations screen

```
add trunk-group next                                            Page   4 of   x
                              PROTOCOL VARIATIONS

                        Mark Users as Phone? n
              Prepend '+' to Calling Number? n
        Send Transferring Party Information? n

                      Send Diversion Header? n
                    Support Request History? y
               Telephone Event Payload Type:
```

Mark Users as Phone

Valid entries	Usage
y/n	When this field is set to **y**, URIs in call control signaling messages originated at the gateway are encoded with the "user=phone" parameter. No subscription messages are encoded with the "user=phone" parameter, even when the field is set to **y**. Default is **n**. **Note:** Do not change the default of **n** for this field unless you are sure that every recipient of SIP Enablement Services (SES) calls using this trunk can accept and properly interpret the optional "user=phone" parameter. Enterprise users without support for "user=phone" in their SIP Enablement Services (SES) endpoints will experience adverse effects, including rejected calls.

Network Call Redirection

Use this field to control which trunk groups the Network Call Redirection (NCR) service is signaled over. NCR only works on trunk groups connected to Service Providers that support NCR.

Valid entries	Usage
y/n	Enter **y** to specify this trunk group for Network Call Redirection. Default is **n**.

Prepend "+" to Calling Number

Appears when the **Group Type** is **sip**. When set to **y**, the calling party number in the header of the SIP message is prepended with a plus sign (+).

Valid entries	Usage
y/n	Set this field to **y** to add a plus sign (+) to the beginning of a number to accommodate international calls. Default is **n**.

Send Transferring Party Information

Valid entries	Usage
y	Enter **y** to send the transferring party information on a transferred call.
n	Default. Transferring party information is not sent.

Telephone Event Payload Type

Use this field to control the default payload type offered by Communication Manager for SIP trunks. The payload type number encoding for originating (offering) the RFC 2833 RTP "telephone-event" payload format is based on the administered number from this field. This value is used only for Communication Manager originations (outgoing offers).

Valid entries	Usage
96 to **127**, or blank	Enter the RTP payload type. Default is **127**.

Field descriptions for Group Member Assignments page

The total number of pages of the Trunk Group screen, and the page number of the first page of Group Member Assignments, vary depending on whether the Administrable Timers and ATMS Thresholds pages display. Note that the Group Member Assignments screen is repeated, as needed, to allow assignment of all group members. This section does not cover ISDN trunks. For descriptions of the screens and fields that are unique to ISDN trunks, see ISDN Trunk Group on page 539.

Note:
For SIP Enablement Services (SES) trunks, the group member-assignment pages are ***not*** individually administrable. The system automatically populates and displays these fields based on the number of members of SES trunk groups specified on page 1. Note that these display-only group member-assignment pages of the **Trunk Group** screen are repeated, as needed, to support all the trunk group's members.

Figure 344: Group Member Assignments screen

```
add trunk-group next                                                 Page y of x
                                 TRUNK GROUP
                                        Administered Members(min/max): xxx/yyy
                                           Total Administered Members: xxx
GROUP MEMBER ASSIGNMENTS
       Port      Code  Sfx  Name          Night          Mode      Type    Ans Delay
    1: _______              ___________   _____          ________  _______ ____
    2: _______              ___________   _____          ________  _______ ____
    3: _______              ___________   _____          ________  _______ ____
    4: _______              ___________   _____          ________  _______ ____
    5: _______              ___________   _____          ________  _______ ____
    6: _______              ___________   _____          ________  _______ ____
    7: _______              ___________   _____          ________  _______ ____
    8: _______              ___________   _____          ________  _______ ____
    9: _______              ___________   _____          ________  _______ ____
   10: _______              ___________   _____          ________  _______ ____
   11: _______              ___________   _____          ________  _______ ____
   12: _______              ___________   _____          ________  _______ ____
   13: _______              ___________   _____          ________  _______ ____
   14: _______              ___________   _____          ________  _______ ____
   15: _______              ___________   _____          ________  _______ ____
```

Administered Members (min/max)

This display-only field shows the minimum and maximum member numbers that have been administered for this trunk group.

Ans Delay

CAUTION:
Customers should not attempt to administer this field. Contact your Avaya technical support representative for assistance.

Valid entries	Usage
20 to **5100** in increments of 20	Specifies the length of time (in ms) your server running Communication Manager will wait before it sends answer supervision for incoming calls on tie trunks using the TN722A or later, TN760 (B, C, or D), TN767, TN464 (any suffix), TN437, TN439, TN458, or TN2140 circuit packs.
blank	Same as setting the field to zero.

This delay serves two purposes:

- It ensures that the answer supervision signal is valid and not a secondary delay-dial or wink-start signal.
- It ignores momentary off-hook signals resulting from connections made off-network through certain No. 5 Crossbar CCSA switches as the connection is being established. Therefore, calls aren't dropped inappropriately.

Code

This display-only field shows the type of circuit pack physically installed or logically administered at the location to which this member is assigned. If no circuit pack is installed or administered at the port address you enter, the field is blank.

Mode

This field specifies the signaling mode used on tie trunks with TN722A or later, TN760B or later, TN767, TN464 (any suffix), TN437, TN439, TN458, or TN2140 circuit packs. This entry must correspond to associated dip-switch settings on the circuit pack.

CAUTION:
Customers should not attempt to administer this field. Contact your Avaya technical support representative for assistance.

Valid entries	Usage
e&m	Enter **e&m** for 6-wire connections that pair 2 signaling wires with 4 voice wires. You'll use **e&m** in the vast majority of systems in the U.S.
simplex	Enter **simplex** for 4-wire connections that do not use an additional signaling pair. This configuration is very rare in the U.S.
protected	

Name

Your vendor, as well as Avaya technical support staff, sometimes need to identify specific trunks to work with your system. Therefore, the name you give to a trunk should identify the trunk unambiguously.

Valid entries	Usage
Up to 10 characters	Examples: • The telephone number assigned to incoming trunks • The Trunk Circuit Identification number assigned by your service provider

Night

Use this field only if you want to assign a night service destination to individual trunks that is different from the group destination entered in the **Night Service** field on page 1. Incoming calls are routed to this destination when the system is placed in night service mode.

Valid entries	Usage
a valid extension	Enter the extension of the night destination for the trunk.
attd	Enter **attd** if you want calls to go to the attendant when night service is active.
blank	

Port

If this trunk is registered as an endpoint in an IP application, this field displays T00000.

Valid entries	Usage
1 to **64** (S87XX/S8300 Servers)	First and second characters are the cabinet number.
A to **E**	Third character is the carrier.
0 to **20**	Fourth and fifth characters are the slot number.
01 to **04** (Analog TIE trunks) **01** to **31** (CSI, S87XX Servers)	Six and seventh characters are the circuit number.
1 to **250** (S87XX/S8300 Servers)	Gateway
V1 to **V9** (DEFINITY CSI, S87XX Servers)	Module
01 to **31** (DEFINITY CSI, S87XX Servers)	Circuit

Note:
In DCS networks, trunks must be assigned the same member number at both nodes.

Members assigned to IP trunk groups displays a value of **T00001**.

When administering analog trunks connected to a TIM518, physical ports 17-24 are administered as ports 9 to 16 in Communication Manager.

Sfx

This display-only field shows the model suffix for the type of circuit pack physically installed at the location to which this member is assigned. If no circuit pack is installed at the port address you enter, the field is blank.

Total Administered Members

This display-only field shows the total number of members administered in the trunk group.

Type

The **Type** column appears when the **Trunk Type** field is blank or **cont**. The **Type** column does not display if the **Trunk Type** field is **dis**.

This field specifies the signaling type to be used with TN760B (or later release), TN722 (with any suffix), TN767, TN2140 (when the **Trunk Type** field is **cont**), TN437, TN439, TN464 with any suffix, or TN458 circuit packs.

CAUTION:
Customers should not attempt to administer this field. Contact your Avaya technical support representative for assistance.

Valid entries	Usage
t1-stan	t1-stan (DEFINITY, S87XX Series IP-PNC)
t1-comp	t1-comp (DEFINITY, S87XX Series IP-PNC)
t5-rev	(S87XX Series IP-PNC) The value of **t5 rev** is allowed only for the TN760D vintage 10 or later. When Type is **t5 rev**, Mode must be **e&m**.
type-5	type-5 (S87XX Series IP-PNC)

Related topics

See Trunks and Trunk Groups in *Avaya Aura™ Communication Manager Feature Description and Implementation,* 555-245-205, for more information on all types of trunk groups except ISDN.

See ISDN Service in *Avaya Aura™ Communication Manager Feature Description and Implementation,* 555-245-205, for more information.

Uniform Dial Plan Report

Figure 345: Uniform Dial Plan Report screen

```
list uniform-dialplan                                             Page 1 of x
                                UNIFORM DIAL PLAN REPORT

 Matching Pattern    Len    Del     Insert Digits      Net      Conv     Node Num
    2                 4      0          817            aar       n
    4                 5      1          334            aar       n
    43659             5      1          928            aar       y
    6                 6      1                         ext       n
    73012             5      1                         enp       n         31
    74100             5      0          81             ars       y
```

Field descriptions for page 1

Matching Pattern

The number you want Communication Manager to match to dialed numbers.

Len

The number of user-dialed digits the system collects to match to the dialed string.

Del

The number of digits deleted before routing the call.

Insert Digits

Use this field to specify the digits or the number of digits of the location prefix that Communication Manager inserts in the dialed string before routing a call.

Valid entries	Usage
0 to **9**	The digits that replace the deleted portion of the dialed number. You can enter up to 10 digits. Leave the field blank to delete the digits.
L*x*	The variable ***x*** is a number from 1 to 11 and represents the number of leading digits of the administered location prefix. Communication Manager prepends the location prefix digits to the dialed string. The value of x must be less than the value of the Prefix field on the Location screen.

Net

The server or switch network used to analyze the converted number.

Conv

Indicates whether additional digit conversion is allowed.

Node Num

The Extension Number Portability (ENP) node number.

Uniform Dial Plan Table

The **Uniform Dialing Plan** field must be **y** on the System Parameters Customer-Options (Optional Features) screen before you can administer this table.

The UDP provides a common dial plan length — or a combination of extension lengths — that can be shared among a group of Avaya S8XXX Servers. Additionally, UDP can be used alone to provide uniform dialing between two or more private switching systems without ETN, DCS, or Main/Satellite/Tributary configurations.

See Uniform Dial Plan in *Avaya Aura™ Communication Manager Feature Description and Implementation,* 555-245-205, for more information.

Field descriptions for page 1

Figure 346: Uniform Dial Plan Table screen

```
change uniform-dialplan 0                                       page 1 of x
                        UNIFORM DIAL PLAN TABLE
                                                          Percent Full:   2
Matching Pattern Len Del Insert Digits Net Conv Node Num
1234567890123..  12  1   1234567890    123 n    123
```

Conv

Valid entries	Usage
y/n	Enter **y** to allow additional digit conversion

Del

Valid entries	Usage
0 to **9**	Enter the number of digits to delete before routing the call. This number must be less than or equal to the number entered in the **Len** field.

Insert Digits

Use this field to specify the digits or the number of digits of the location prefix that Communication Manager must insert in the dialed string before routing the call.

Valid entries	Usage
0 to **9**	Enter the digits that must replace the deleted portion of the dialed number. You can enter up to 10 digits. Leave the field blank to delete the digits
L*x*	The variable ***x*** is a number from 1 to 11 and represents the number of leading digits of the administered locations prefix on the **Locations** screen. Communicarion Manager repends the digits to the dialed string. The value for ***x*** must less than the number of digits in the location prefix.

Len

Valid entries	Usage
1 to **18**	Enter the number of user-dialed digits the system collects to match to this Matching Pattern. This number must be greater than or equal to the number entered in the **Matching Pattern** field. The value **2** can be used only when **Insert Digits** contains an **L*x*** value, where *x* is the number of leading digits to prepend for the location of an originating call.

Matching Pattern

Communication Manager matches the value of this field to the dialed numbers.

Valid entries	Usage
0 to **9**	Communication Manager matches the value of this field to the dialed numbers. You can enter up to 13 digits.
blank	Leave this field blank if you do not want to specify a pattern.

Net

Enter the server or switch network used to analyze the converted number.

Valid entries	Usage
aar **ars** **enp** **ext**	The converted digit-string will be routed either as an extension number or via its converted AAR address, its converted ARS address, or its ENP node number. If you enter **enp**, you must enter the ENP node number in the **Node Num** field. The **Insert Digits** field must be blank, and **Conv** must be **n**.

Node Num

This is the ENP (Extension Number Portability) Node Number.

Valid entries	Usage
1 to **999**	Enter the ENP node number.

Percent Full

Displays the percentage (0 to 100) of the memory resources allocated for the uniform dial plan data that are currently being used.

Acceptable Service Level (sec)

Only appears when, on the System Parameters Customer-Options (Optional Features) screen, the **BCMS/VuStats Service Level** field is **y** and the **Measured** field is **internal** or **both**.

Valid entries	Usage
0 to **9999** seconds	Enter the number of seconds within which calls to this VDN should be answered. This allows BCMS to print out a percentage of calls that were answered within the specified time.

User Profile

This screen is described in *Maintenance Commands for Avaya Aura™ Communication Manager, Media Gateways and Servers*, 03-300431.

For more information on administering user profiles and logins, see AAA Services in *Avaya Aura™ Communication Manager Feature Description and Implementation*, 555-245-205.

Variables for Vectors

Use this screen to create variables and define the necessary parameters for each variable type. You can specify the variable type, the name to use for the variable, the size of the variable, how the variable gets set/assigned and whether the variable is local or global. Up to 702 variables can be supported using A to Z and AA to ZZ rows.

Field descriptions for page 1

Figure 347: Variables for Vectors screen - page 1

```
change variables                                                 Page   1 of x
                               VARIABLES FOR VECTORS

Var   Description       Type   Scope Length Start   Assignment          VAC
A:    _______________   _____ __     ____   ___
B:    _______________   _____ __     ____   ___
C:    _______________   _____ __     ____   ___
D:    _______________   _____ __     ____   ___
E:    _______________   _____ __     ____   ___
F:    _______________   _____ __     ____   ___
G:    _______________   _____ __     ____   ___
H:    _______________   _____ __     ____   ___
I:    _______________   _____ __     ____   ___
J:    _______________   _____ __     ____   ___
K:    _______________   _____ __     ____   ___
L:    _______________   _____ __     ____   ___
M:    _______________   _____ __     ____   ___
```

Figure 348: Variables for Vectors screen - page 2

```
change variables                                                 Page   2 of x
                               VARIABLES FOR VECTORS

Var   Description       Type   Scope Length Start   Assignment          VAC
N:    _______________   _____ __     ____   ___
O:    _______________   _____ __     ____   ___
P:    _______________   _____ __     ____   ___
Q:    _______________   _____ __     ____   ___
R:    _______________   _____ __     ____   ___
S:    _______________   _____ __     ____   ___
T:    _______________   _____ __     ____   ___
U:    _______________   _____ __     ____   ___
V:    _______________   _____ __     ____   ___
W:    _______________   _____ __     ____   ___
X:    _______________   _____ __     ____   ___
Y:    _______________   _____ __     ____   ___
Z:    _______________   _____ __     ____   ___
```

Assignment

This field only allows entry when the **Type** is **value** or **collect G**. Entry of an **Assignment** for **value** or **collect G** is optional. That is, it can be left blank. The current value/assignment for each global variable is always displayed in the **Assignment** column when you access the Variables for Vectors screen. This includes the **doy**, **dow**, and **tod** types which show the current values from the switch clock as a display-only entry in the **Assignment** column.

Valid entries	Usage
digits	Enter a number to pre-assign to the variable. This field displays the current value for global values

Description

Valid entries	Usage
up to 27 alphanumeric characters, or blank	Optionally enter an identifying name or description of the vector variable. Default is blank.

Length

This field specifies the maximum number of digits from the data to assign to the variable. Length does not apply to the **tod**, **doy**, **dow** or **vdn** variables. When **Type** is **value**, the length is pre-populated with **1**. A length entry is required for all types to which it applies.

Valid entries	Usage
1 to **16**	Enter the maximum length of digits to use in the variable.

Scope

This field only allows an entry for variables that can be *either* local or global. For those variables that can only be one or the other, the **L** or **G** value is pre-populated automatically after you enter the **Type**.

Valid entries	Usage
G/L	Indicate whether the variable is to be used locally (**L**) or globally (**G**).

Start

This field specifies the beginning character position of the data digits string to be used for assigning to the variable. The combination of the **Start** position and maximum length of the digits string defines what is to be assigned to the variable. If the number of digits to be used is less than the maximum length specified, only that portion is assigned to the variable. **Start** only allows entry when **Type** is **collect** or **asaiuui**.

Valid entries	Usage
1 to **96**	Enter the character position of the first digit to be stored in the variable.

Type

Valid entries	Usage
ani **asaiuui** **collect** **dow** **doy** **stepcnt** **tod** **value** **vdn** **vdntime**	Enter the vector variable type.

Var

Valid entries	Usage
A to **Z**, **AA-ZZ**	Display only. The letter identifying the row of a specific vector variable.

VAC

The **VAC** (Variable Access Code) column only allows entry (**1** to **9** or blank) when the **Type** is **value**. Entry is not required for this type. If VAC is left as a blank, assignment is done using the **Assignment** column. The **VVx** entry is one of the Vector Variable feature items on the **Feature Access Code (FAC) screen** that can be assigned a feature access code (FAC).

Valid entries	Usage
1 to **9** or blank	Displays the Vector Variable Feature Access Code (FAC) to use for changing the value.

Vector

See **Call Vector**.

Vector Directory Number

This screen defines vector directory numbers (VDN) for the Call Vectoring feature. A VDN is an extension number used to access a call vector. Each VDN is mapped to one call vector.

VDNs are software extension numbers (that is, not assigned to physical equipment). A VDN is accessed via direct dial CO trunks mapped to the VDN (incoming destination or night service extension), DID trunks, and LDN calls. The VDN can be Night Destination for LDN.

See *Avaya Aura™ Call Center 5.2 Call Vectoring and Expert Agent Selection (EAS) Reference*, 07-600780, for more information.

Field descriptions for page 1

Figure 349: Vector Directory Number screen

```
change vdn 20001                                                Page   1 of   x
                            VECTOR DIRECTORY NUMBER

                             Extension: 20001
                                 Name*: California Customer Service
                           Destination: Vector Number        500
                   Attendant Vectoring? n
                  Meet-me Conferencing? n
                    Allow VDN Override? n
                                   COR: 1
                                   TN*: 1
                              Measured: none


        VDN of Origin Annc. Extension*:
                            1st Skill*:
                            2nd Skill*:
                            3rd Skill*:



* Follows VDN Override Rules
```

1st/2nd/3rd Skill

Only appears when, on the System Parameters Customer-Options (Optional Features) screen, the **Expert Agent Selection (EAS)** field is **y** and the **Meet-me Conferencing** field is **n**. When **Meet-me Conferencing** is **n**, an asterisk (*) appears next to the field name, indicating that this field follows VDN override rules when the system changes the "active" VDN for a call.

Valid entries	Usage
1 to **999** or blank	Enter the desired Skill numbers in each field (or leave blank). The default is blank.

Acceptable Service Level (sec)

Only appears when, on the System Parameters Customer-Options (Optional Features) screen, the **BCMS/VuStats Service Level** field is **y** and the **Measured** field is set to **internal** or **both**.

Valid entries	Usage
0 to **9999** seconds or blank	Enter the number of seconds within which calls to this VDN should be answered. This allows BCMS to report the percentage of calls that were answered within the specified time. The default is blank.

Allow VDN Override

This field appears if the **Meet-me Conferencing** field is **n**. The **Allow VDN Override** field allows the system to change the "active" VDN for a call. The “active” VDN is the VDN to be used for parameters associated with the call such as VDN name, skills, tenant number, BSR application, VDN variables, etc.

Parameters (VDN fields) for the call that are defined by the “active” VDN include the fields in the following list. When **Meet-me Conferencing** is **n**, an asterisk (*) appears next to each field name, indicating that the field follows VDN override rules when the system changes the "active" VDN for a call.

- VDN Name
- Tenant Number (TN)
- VDN of Origin Announcement Extension
- VDN skills (1st, 2nd, 3rd)
- Return Destination
- VDN Timed ACW Interval
- BSR Application
- BSR Available Strategy
- BSR Tie Strategy
- Display VDN for Route-to DAC
- ISDN Trunk ASAI Messages (depending on field setting)
- BSR Local Treatment
- VDN Variables
- VDN Time Zone Offset

Note:
The "active" VDN can be specified in some vector commands as a keyword. When a vector step with the keyword "active" is executed, the extension for the call's "active" VDN as defined by VDN override rules is substituted for the keyword when processing the vector command. The keyword "active" can be used as the VDN extension for the **goto** command "counted-calls" conditional, the **goto** command "rolling-asa for VDN" conditional, the messaging command mailbox extension, or can be defined as the "vdn" vector variable type assignment. The keyword "latest," (the last VDN routed to), can also be assigned in these same vector commands or variables, but the "latest" VDN is not changed by VDN Override settings.

Valid entries	Usage
y	Entering **y** in this field allows a routed-to VDN (by a route-to number or route-to digits vector command) to become the “active” VDN. The first VDN reached by the call becomes the “active” VDN.
n	The routed-to VDN does not become the active VDN. The parameters of the original VDN are used. This is the default.

Attendant Vectoring

This field appears when, on the System Parameters Customer-Options (Optional Features) screen, **Attendant Vectoring** is **y**. This field indicates if the vector you are defining is an attendant vectoring VDN.

Valid entries	Usage
y	Enter **y** so the vector is an attendant vector. This entry will dynamically change the rest of the screen to eliminate field options available with other types of vectors.
n	Default.

COR

Specifies the class of restriction (COR) of the VDN.

Valid entries	Usage
0 to **995**	Enter a 1 or 2-digit number. This field cannot be blank.

Destination

Specify if the calls are routed using a Vector Number or Policy Routing Table. Valid entries are **Vector Number** and **Policy Routing Table**.

Extension

This is a display-only field showing the extension number of the VDN. The extension is a number that starts with a valid first digit and length as defined by the system's dial plan.

Measured

This field appears if the **Meet-me Conferencing** field is **n**. Used to collect measurement data for this VDN. Data can be collected for reporting by BCMS or CMS.

Note:
On the System Parameters Customer-Options (Optional Features) screen, the **BCMS** field must be **y** for the **Measured** field to be set to **internal** or **both**. In addition, the appropriate CMS release must be administered on the Feature-Related System Parameters screen if this field is being changed to **external** or **both**.

Valid entries	Usage
internal	Data is measured internally by BCMS.
external	Data is measured internally by CMS.
both	Data is measured internally by both BCMS and CMS.
none	Data is not measured. This is the default.

Meet-me Conference

This field appears only if, on the System Parameters Customer-Options (Optional Features) screen, the **Enhanced Conferencing** field is **y**. This field determines if the VDN is a Meet-me Conference VDN.

Note:
If the VDN extension is part of your DID block, external users will be able to access the conference VDN. If the VDN extension is not part of your DID block, only internal callers on the your network (including DCS or QSIG) or remote access callers can access the conference VDN.

Valid entries	Usage
y/n	Enter **y** to enable Meet-me Conference for this VDN. If Meet-me Conference is **y**, only **Extension**, **Name**, **Vector Number**, **Meet-me Conference**, **COR**, and **TN** fields display and the fields for page 2 change. Both **Attendant Vectoring** and **Meet-me Conference** cannot be enabled at the same time. If **Enhanced Conferencing** is **y**, but no other vectoring options are enabled, only Meet-me Conference vectors can be assigned.

Note:
If the vector for Meet-Me conferencing allows a new party to join a conference immediately, and that party is joining as an H.323 ip trunk user, the caller might not have talkpath with the others in the conference. To prevent this, include in the vector a short delay before a new party joins the Meet-Me conference, such as a step to collect digits, a 1-second delay, or play an announcement. Since Meet-Me vectors are always configured with announcements and digit collections, this should rarely be an issue.

Name

This is an optional field that need not contain any data. It is the name associated with the VDN. When **Meet-me Conferencing** is **n**, an asterisk (*) appears next to the field name, indicating that this field follows VDN override rules when the system changes the "active" VDN for a call.

Valid entries	Usage
Enter up to a 27-character alphanumeric name that identifies the VDN.	The name might be truncated on agents' displays depending on the application. When information is forwarded with an interflowed call, only the first 15 characters are sent. **Note:** For 4610SW, 4620SW, 4621SW, and 4622SW, Sage, Spark, and 9600-series Spice telephones, the **Name** field has an associated optional native name field that is supported by the Unicode language display. The native name field is accessible through the Integrated Management Edit Tools such as Avaya Site Administration (ASA). Unicode is also an option for the 2420J telephone when **Display Character Set** on the **System Parameters Country-Options** screen is **katakana**. For more information on the 2420J, see *2420 Digital Telephone User's Guide*, 555-250-701. Avaya BRI stations support ASCII characters only. They do not support non-ASCII characters, such as Eurofont or Kanafont. Therefore, if you use non-ASCII characters in any Communication Manager **Name** field, such characters do not display correctly on a BRI station.

Number

Enter the number of the Vector or the Policy Routing Table (PRT) using which the calls are allocated. Valid entries are 1-2000.

Note:
This field does not appear on the screen, but this is a separate field next to the **Destination** field. When you specify a PRT number, the **Attendant Vectoring** and **Meet-Me Conferencing** fields do not appear, as Policy Routing does not support **Attendant Vectoring** or **Meet-Me Conferencing**.

Service Objective

Use this field to assign a service level to the VDN. The system displays this field only if you set the **Dynamic Advocate** field to **y**. The range of the acceptable value is between 1 to 9999. The default value is 20. For more information on **Service Objective**, see *Avaya Business Advocate User Guide*.

TN

Specifies the Tenant Partition number for this VDN. When **Meet-me Conferencing** is **n**, an asterisk (*) appears next to the field name, indicating that this field follows VDN override rules when the system changes the "active" VDN for a call.

Valid entries	Usage
1 to **100**	For S87XX Series IP-PNC.

VDN of Origin Annc. Extension

Use this field to specify the extension number of the VDN of Origin announcement. A VDN of Origin announcement is a short recording that identifies something about the call originating from the VDN. The agent hears the recording just prior to the delivery of the call. Data for this field appears only when, on the System Parameters Customer-Options (Optional Features) screen, the **VDN of Origin Announcement** field is **y** and, on this screen, the **Meet-me Conferencing** field is **n**. When **Meet-me Conferencing** is **n**, an asterisk (*) appears next to the field name, indicating that this field follows VDN override rules when the system changes the "active" VDN for a call.

Valid entries	Usage
VDN extension	Enter the extension number of the VDN of Origin announcement.

Vector Number

Valid entries	Usage
1 to **system max**	Enter an identifying number that specifies a particular call vector that is accessed through the VDN. This field cannot be blank.

Field descriptions for page 2 (Meet-me Conference is n)

The second page of the Vector Directory Number screen contains the name of the corresponding Audix server (if present), the BSR available agent strategy, whether the VDN is displayed in route to direct agent call situations, and settings for other optional features.

Figure 350: Vector Directory Number screen

```
change vdn nnnn                                                 Page  2 of  x
                            VECTOR DIRECTORY NUMBER

                                 AUDIX Name:


                    VDN Timed ACW Interval*:
                           BSR Application*:
              BSR Available Agent Strategy*: 1st-found
                          BSR Tie Strategy*: system

                    Observe on Agent Answer? y


              Display VDN for Route-To DAC*? y
 VDN Override for ISDN Trunk ASAI Messages*? y

                       BSR Local Treatment*? y

          Reporting for PC Predictive Calls? n
           Pass Prefixed CPN to VDN/Vector*? system
* Follows VDN Override Rules
```

AUDIX Name

Only appears for S87XX Series IP-PNC configurations. If this VDN is associated with the AUDIX vector, enter the name of the AUDIX machine as it appears in the IP Node Names screen.

BSR Application

To use multi-site Best Service Routing with this VDN, enter a one to three-digit number to specify an application plan for the VDN. This field appears if, on the System Parameters Customer-Options **(Optional Features)** screen, the **Lookahead Interflow (LAI)** and **Vectoring (Best Service Routing)** fields are **y**. When **Meet-me Conferencing** is **n**, an asterisk (*) appears next to the field name, indicating that this field follows VDN override rules when the system changes the "active" VDN for a call.

Valid entries	Usage
1 to **255** or blank	Enter a 1 to 3-digit number. For DEFINITY CSI.
1 to **511** or blank	Enter a 1 to 3-digit number. For S8300/S87XX Servers.

BSR Available Agent Strategy

The available agent strategy determines how Best Service Routing identifies the best split or skill to service a call in an agent surplus situation. To use Best Service Routing with this VDN, enter an agent selection strategy in this field.

This field only appears if, on the System Parameters Customer-Options **(Optional Features)** screen, the **Vectoring (Best Service Routing)** field is **y**. When **Meet-me Conferencing** is **n**, an asterisk (*) appears next to the field name, indicating that this field follows VDN override rules when the system changes the "active" VDN for a call.

Valid entries	Usage
1st-found	BSR uses the first selection for routing; that is, the current best selected from the previous consider commands.
UCD-LOA	The call is routed to the least occupied agent, without regard to skill level. Can be set only if, on the System Parameters Customer-Options (Optional Features) screen, the **Least Occupied Agent (LOA)** or **Business Advocate** field is **y**.
UCD-MIA	The call is routed to the most idle agent, without regard to skill level. This type of call distribution ensures a high degree of equity in agent workloads even when call-handling times vary.
EAD-LOA	The call is routed to the highest skill level agent with the lowest occupancy. Can be set only if, on the System Parameters Customer-Options (Optional Features) screen, the **Least Occupied Agent (LOA)** or **Business Advocate** field is **y**.
EAD-MIA	The call is routed to the highest skill level, most idle agent. Can be set only if, on the System Parameters Customer-Options (Optional Features) screen, the **Expert Agent Selection (EAS)** field is **y**.

BSR Local Treatment

In a multi-site BSR configuration, a call that arrives at a local communication server can be rerouted to a remote server located in a different part of the world. This feature maintains control at the local server and allow you to provide local audio feedback for IP and ISDN calls, or to take back the call while the call waits in queue on a remote server. When **Meet-me Conferencing** is **n**, an asterisk (*) appears next to the field name, indicating that this field follows VDN override rules when the system changes the "active" VDN for a call.

Valid entries	Usage
y/n	A **y** entry in this field allows you to provide local audio feedback for IP and ISDN calls while a call waits in queue on a remote server. **Note:** The **BSR Local Treatment** field must be set to **y** on both the local and remote vdns, or else call interflow attempts might result in dropped calls.

BSR Tie Strategy

This field appears only when **Vectoring (Best Service Routing)** on the **System Parameters Customer-Options (Optional Features)** screen is **y**. When **Meet-me Conferencing** is **n**, an asterisk (*) appears next to the field name, indicating that this field follows VDN override rules when the system changes the “active” VDN for a call.

Valid entries	Usage
system	The setting of the **BSR Tie Strategy** field on the **Feature-Related System Parameters screen** applies.
1st-found	BSR uses the previously selected best choice as the best skill or location. This is the default setting.
alternate	Alternates the BSR selection algorithm when a tie in EWT or available agent criteria occurs. Every other time a tie occurs for calls from the same VDN, the consider step with the tie is selected to send the call instead of the first selected split, skill, or location. This helps balance the routing when the cost of routing remotely is not a concern.

Display VDN for Route-To DAC

This field can be set to **y** only if, on the System Parameters Customer-Options (Optional Features) screen, the **Expert Agent Selection (EAS)** field is **y**. This field applies when either:

- A route-to number with coverage = y or route-to digits with coverage = y vector command routes a call to an agent as an EAS direct agent call
- Adjunct routing routes a direct agent call to the agent

When **Meet-me Conferencing** is **n**, an asterisk (*) appears next to the field name, indicating that this field follows VDN override rules when the system changes the “active” VDN for a call. For more information, see *Avaya Aura™ Call Center 5.2 Call Vectoring and Expert Agent Selection (EAS) Reference*, 07-600780.

Valid entries	Usage
y/n	Enter **y** to allow display of the VDN.

Observe on Agent Answer

Valid entries	Usage
y/n	This field allows for a service observer to delay observing a call to the VDN until the call is delivered to the agent/station.

Return Destination

When **Meet-me Conferencing** is **n**, an asterisk (*) appears next to the field name, indicating that this field follows VDN override rules when the system changes the "active" VDN for a call.

Valid entries	Usage
VDN extension or blank	Enter the VDN extension number to which an incoming trunk call is routed if it returns to vector processing after the agent drops the call.

VDN Override for ISDN Trunk ASAI Messages

When **Meet-me Conferencing** is **n**, an asterisk (*) appears next to the field name, indicating that this field follows VDN override rules when the system changes the "active" VDN for a call.This field appears only if the following conditions are set on the Communication Manager license file:

- On the System Parameters Customer-Options (Optional Features) screen, the **ASAI Link Core Capabilities** field is **y**.
- On the System Parameters Customer-Options (Optional Features) screen, the **G3 Version** field is set to **V10** or later

Additionally, you can set this field to **y** only when the **Allow VDN Override** field on this screen is also set to **y**. When **Meet-me Conferencing** is **n**, an asterisk (*) appears next to the field name, indicating that this field follows VDN override rules when the system changes the "active" VDN for a call.

Valid entries	Usage
y	When an incoming call routes, the "Called Number" information sent in the "Call Offered," "Altering," "Queued," and "Connect" ASAI events and the "Adjunct Route Request" ASAI message, is the "active VDN" extension associated with the routed call.
n	The "Called Number" information sent for the ASAI event notification and adjunct-request messages does not change for a ISDN-PRI trunk. It is always the number in the Called Number IE sent in the incoming ISDN call's SETUP message.

VDN Timed ACW Interval

When a value is entered in this field, an agent in auto-in work mode who receives a call from this VDN is automatically placed into After Call Work (ACW) when the call drops. When the administered time is over, the agent automatically becomes available. This field takes precedence over the **Timed ACW Interval** field on the Hunt Group screen. When **Meet-me Conferencing** is **n**, an asterisk (*) appears next to the field name, indicating that this field follows VDN override rules when the system changes the "active" VDN for a call.

Valid entries	Usage
1 to **9999** or blank	Enter the number of seconds the agent should remain in ACW following the call.

Field descriptions for page 2 (Meet-me Conference is y)

The fields on this screen are displayed when the **Meet-me Conference** field on page 1 of the Vector Directory Number screen is **y**.

Figure 351: Vector Directory Number screen (if the Meet-me Conference field is y)

```
change vdn nnnn                                                Page 2 of x
                         VECTOR DIRECTORY NUMBER

                      MEET-ME CONFERENCE PARAMETERS
    Conference Access Code:123456
     Conference Controller:
           Conference Type: expanded
           Route-to Number:
```

Note:
If the vector for Meet-Me conferencing allows a new party to join a conference immediately, and that party is joining as an H.323 ip trunk user, the caller might not have talkpath with the others in the conference. To prevent this, include in the vector a short delay before a new party joins the Meet-Me conference, such as a step to collect digits, a 1-second delay, or play an announcement. Since Meet-Me vectors are always configured with announcements and digit collections, this should rarely be an issue.

Conference Access Code

To ensure conference security, you should always assign an access code to a Meet-me Conference VDN.

Valid entries	Usage
6-digit number or blank	Enter a 6-digit access code for the Meet-me Conference VDN. If you do not want an access code, leave blank. Once an access code is assigned, an asterisk displays in this field for subsequent change, display, or remove operations by all users except the *init* superuser login.

Conference Controller

This field controls which user is allowed to change the access code for a Meet-me Conference VDN using a feature access code. This can be a local user or someone dialing in via remote access trunks.

Valid entries	Usage
extension number or blank	If an extension number is entered, only a user at that extension can change the access code for that VDN using a feature access code. If this field is blank, any station user that is assigned with console permissions can change the access code for that VDN using a feature access code.

Conference Type

Use this field to select the conference type that is appropriate for your call. For six or fewer participants, enter **6-party**. For a conference with more than six participants, select **expanded**.

Valid entries	Usage
6-party **expanded**	Enter **expanded** to enable the Expanded Meet-me Conference feature. Default is **6-party**.

Route-to Number

This field appears only if the **Conference Type** field is **expanded**. This field allows administration of the routing digits (the ARS/AAR Feature Access Code with the routing digits and the Conference ID digits for the VDN).

Valid entries	Usage
up to 16 digits	Enter the ARS or AAR Feature Access Code (FAC) followed by the routing digits. Alternately, you can enter the unique UDP extension. **Note:** The Route-to Number must be unique across all Expanded Meet-me Conference VDNs.

Field descriptions for page 3

Figure 352: Vector Directory Number screen

```
VECTOR DIRECTORY NUMBER                                        Page 3 of x

                                VDN VARIABLES*

                    Var  Description        Assignment
                    V1   _______________    ________________
                    V2   _______________    ________________
                    V3   _______________    ________________
                    V4   _______________    ________________
                    V5   _______________    ________________
                    V6   _______________    ________________
                    V7   _______________    ________________
                    V8   _______________    ________________
                    V9   _______________    ________________

VDN Time Zone Offset*: + HH:MM
                   Daylight Savings Rule*: system
* Follows VDN Override Rules
```

Assignment

The assignment field assigns an up to 16-digit unvalidated decimal number to each of the VDN variables V1 through V5. Valid entries for each assignment can be a string of up to 16 digits using **0** to **9**, or blank.

Daylight Savings Rule

Use this field to define the daylight saving time rule. This field is used with the VDN Time Zone Offset field. The daylight saving time rule and the time zone offset are applied to `goto` time-of-day commands in the vector that is assigned to the VDN. The time-of-day calculations are based on the local time of the receiving call's VDN. The assigned rule number applies start and stop rules that are administered on the system **Daylight Savings Rule** field for that rule number.

Tip:
Use the `list usage vdn-time-zone-offset` command to find VDNs containing an administered daylight saving time rule.

Valid entries	Usage
system	The system uses the same daylight saving time rule as the system clock shown in the **display/set time** field.
0	No daylight saving rule is applied. If the system time has a daylight saving rule specified, this rule is removed before evaluating the `goto if time-of-day` conditional.
1 to 15	Indicates the rule as defined on the Daylight Savings Rule field. When you use a number other than 0, the rule associated with the main server clock display time and the main server offset are not used. The offset and rule assigned to the active VDN for the call are applied to the operating system standard time so that local time for the VDN is used to test the time-of-day step.

Description

This field is displayed only if VDN Variables is active. The description field allows users to describe the VDN variable using up to 15 characters.

Var

The number assigned to the VDN variable. You can assign up to 9 VDN variables.

VDN Time Zone Offset

This field is applied against the switch clock when a time of day vector command is executed. Daylight savings time changes are handled by the switch clock using the existing operation. Based on a syntax of +HH:MM, the valid entries are:

- **+**, **-**
- **00-23** - hour
- **00-59** - minute

The default is **+00:00**. When the default is set, the system switch time is used without modification.When **Meet-me Conferencing** is **n**, an asterisk (*) appears next to the field name, indicating that this field follows VDN override rules when the system changes the "active" VDN for a call.

For more information about this feature, see *Avaya Aura™ Call Center 5.2 Automatic Call Distribution (ACD) Reference*, 07-602568.

VDN Variables

When **Meet-me Conferencing** is **n**, an asterisk (*) appears next to the heading, indicating that variables **V1** through **V9** follow VDN override rules when the system changes the "active" VDN for a call.

Video Bridge

Use the Video Bridge screen to configure available ad-hoc conferencing resources. You can administer up to 40 video bridges.

For more information on Ad-hoc Conferencing, see *Administering Avaya Aura™ Communication Manager*, 03-300509. For more detailed information on Avaya Video Telephony, see *Avaya Video Telephony Solution Networking Guide*, 16-601423.

Field descriptions for page 1

Figure 353: Video Bridge screen

```
add video-bridge next                                            Page 1 of x
                                VIDEO BRIDGE
Bridge ID: 3
     Name:

Max Ports:

Trunk Groups: (Must have at least one incoming and one outgoing, or a two-way)
1: ____
2: ____
3: ____

  Far End Resource Info:          Type:           Call Rate:     : Kbits

               ID Range: _________ to _________
Priority Factory Number:_________
Standard Factory Number:_________
```

Bridge ID

Valid entries	Usage
1 to 40	Display only. Shows the ID number for this video bridge.

Call Rate

This field appears only when **Far End Resource Info** is **n** and **Type** is not changed from the default value **Maximum**.

Valid entries	Usage
1 to system max	Enter the maximum allowable call rate for the conference.

Far End Resource Info

Valid entries	Usage
y	The far end tracks port usage and provides updates on resource availability.
n	No resource information is provided from the far end.

ID Range Start/End

These fields appear when the **Group Type** field on the Trunk Group screen is **h.323**.

Valid entries	Usage
Enter 1-9 digits (0,9)	Enter a range of conference IDs that this video bridge can use. There must be enough IDs so that all of the ports can be used – one ID for every six ports. Default is blank.

Max Ports

Valid entries	Usage
3 to system max	Enter the maximum number of video conferencing ports for this video bridge. Default is none.

Name

Valid entries	Usage
up to 30 alphanumeric characters	Enter a name for identifying this video bridge. Default is blank.

Priority Factory Number

This field appears when the **Group Type** field on the Trunk Group screen is **h.323** or **sip**, and the **Far End Resource Info** field on the Video Bridge screen is **y**. When creating an ad-hoc conference call, Communication Manager first contacts the conference factory, which allocates the ad-hoc conference ID, and establishes an audio channel between the video bridge and Communication Manager audio resources. Priority vs. Standard factory number depends on who creates the conference; if a user with Priority Video permissions creates it, the priority factory is used, which may have better bandwidth or a dedicated video bridge. Note that the Priority/Standard distinction only applies when the **Far End Resource Info** field on the Video Bridge screen is **y**.

Valid entries	Usage
1-9 digits (0,9), or blank	At least one of **Priority Factory Number** or **Standard Factory Number** must be filled in. If **Priority Factory Number** is blank, priority calls can use the bridge, but will prefer a bridge with a priority factory. Standard and Priority factory numbers can be the same. Default is blank.

Standard Factory Number

This field appears when the **Group Type** field on the Trunk Group screen is **h.323** or **sip**. For h.323, the **Far End Resource Info** field on the Video Bridge screen must be **y**. When creating an ad-hoc conference call, Communication Manager first contacts the conference factory, which allocates the ad-hoc conference id, and establishes an audio channel between the video bridge and Communication Manager audio resources. Priority vs. Standard factory number depends on who creates the conference; if a user with Priority Video permissions creates it, the priority factory is used, which may have better bandwidth or a dedicated video bridge. Once established as a priority conference, the call remains priority even if the priority user drops off. Note that the Priority/Standard distinction only applies when the **Far End Resource Info** field on the Video Bridge screen is **y**.

Valid entries	Usage
1-9 digits (0,9), or blank	At least one of Priority Factory Number or Standard Factory Number must be filled in. If **Standard Factory Number** is blank, non-priority conferences are unable to use this video bridge. Standard and Priority factory numbers can be the same. Default is blank.

Trunk Groups

Use this field to assign trunk groups to this video bridge. You must have at least one incoming and one outgoing trunk, or a two-way trunk. Note that all trunks on a given video bridge must be the same type; you cannot mix H.323 and SIP.

Valid entries	Usage
1 to **2000**	Enter administered SIP or ISDN H.323 trunk groups. Default is blank.

Type

This field appears when **Far End Resource Info** is **n**. When **Type** has the default value of **Maximum**, the **Call Rate** field is enabled.

Valid entries	Usage
Exact, **Maximum**, or **Region**.	When **Type** is **Maximum**, any call rate up to the configured rate is allowed in a conference. When **Type** is **Exact**, all participants in the conference must use the exact rate specified or be in the *audio-only* mode. Default is **Maximum**. For an existing video bridge, **Type** initially appears as **Region** and the **Call Rate** field is not displayed. You can set these values for a new video bridge also.

Virtual MAC Addresses

The Virtual MAC Addresses screen lists the virtual Media Access Control (MAC) addresses on your system.

Field descriptions for page 1

Figure 354: Virtual MAC Addresses

```
display virtual-mac-address 1                                     Page 1 of x
                   VIRTUAL MAC ADDRESSES - TABLE: 1
      MAC Address     Used                        MAC Address     Used
    00:04:0d:4a:53:c0   y                       00:04:0d:4a:53:cf   n
    00:04:0d:4a:53:c1   n                       00:04:0d:4a:53:d0   n
    00:04:0d:4a:53:c2   n                       00:04:0d:4a:53:d1   n
    00:04:0d:4a:53:c3   n                       00:04:0d:4a:53:d2   n
    00:04:0d:4a:53:c4   n                       00:04:0d:4a:53:d3   n
    00:04:0d:4a:53:c5   n                       00:04:0d:4a:53:d4   n
    00:04:0d:4a:53:c6   n                       00:04:0d:4a:53:d5   n
```

MAC Address

Valid entries	Usage
15 alpha-numeric characters	Virtual MAC address shared by duplicated TN2602AP circuit packs. **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.

Used

Valid entries	Usage
y/n	This field is autopopulated. If **y**, the associated virtual MAC address has been assigned in the system

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