Lucent Technologies
Bell Labs Innovations

MultiPoint Conferencing Unit (MCU)

Release 6.0

Administration

555-027-502
Comcode 108394677
Issue 1
April 1999

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Part 68: Network Registration Number. This equipment is registered with the FCC in accordance with Part 68 of the FCC Rules. This equipment is identified by the FCC registration number AS5USA-75530-MF-E.

Prior to connecting this device, the following information should be given to the Telephone Company.

Facility Interface Code: 04DU9-DN

Service Order Code: 6.0P

Notice
The Canadian Department of Communications label identifies certified equipment. This certification means that the equipment meets certain telecommunications network protective, operational and safety requirements. The Department does not guarantee the equipment will operate to the user's satisfaction.

Before installing this equipment, users should ensure that it is permissible to be connected to the facilities of the local telecommunications company. The equipment must also be installed using an acceptable method of connection. In some cases, the company's inside wiring associated with a single line individual service may be extended by means of a certified connector assembly (telephone extension cord). The customer should be aware that compliance with the above conditions may not prevent degradation of service in some situations.

Repairs to certified equipment should be made by an authorized Canadian maintenance facility designated by the supplier. Any repairs or alterations made by the user to this equipment, or equipment malfunctions, may give the telecommunications company cause to request the user to disconnect the equipment. Users should ensure for their own protection that the electrical ground connections of the power utility, telephone lines, and internal metallic water pipe system, if present, are connected together. This precaution may be particularly important in rural areas.

⚠ CAUTION:

Users should not attempt to make such connections themselves, but should contact the appropriate electric inspection authority, or electrician, as appropriate.

Notice
The Load Number (LN) assigned to each terminal device denotes the percentage of the total load to be connected to a telephone loop which is used by the device, to prevent overloading. The termination on a loop may consist of any combination of devices subject only to the requirement that the total of the Load Numbers of all the devices does not exceed 100.

Load Number: Not applicable

Standard Connecting Arrangement: CA81A

Canadian Equipment Maintenance Facility:
Lucent Technologies, Inc. Canada
3650 Victoria Park Avenue, Suite 700
Willowdale, Ontario M2H3P7
Telephone Number 416 756-5222

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This document was prepared by BCS Product Publications, Lucent Technologies, Inc., Middletown, NJ 07748-9972.

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About This Document

This document describes the Lucent Technologies MultiPoint Conferencing Unit (MCU) reservation, administration, and troubleshooting functions, along with the reports available to a telecommunications manager or an on-site MCU system administrator.

Prerequisites

It is assumed that before using this document you have knowledge about telecommunications equipment and an understanding of trunks and network services. A basic familiarity with point-to-point or multipoint video conferencing is helpful for prequalifying video endpoints and establishing conference parameters.

Related Information

Use this document in conjunction with the following Lucent Technologies MCU documents and information sources:

- *Lucent Technologies MultiPoint System Guide for the MultiPoint Conferencing Unit/Conference Reservation and Control System*, 555-027-214

  Provides an overview of the MultiPoint Conferencing Unit (MCU) and Conference Reservation and Control System (CRCS), including their capabilities, features, and advantages to the customer.

- *Lucent Technologies MultiPoint Conferencing Unit Maintenance,* 555-027-107

  Provides information about the tests, alarms, and maintenance of a Lucent Technologies MCU.

- *Lucent Technologies MultiPoint Conferencing Unit Seminar,* BM1068U

  A Lucent Technologies course offering that provides an overview of the Lucent Technologies MCU administration and scheduling functions.

- *Lucent Technologies Conference Reservation and Control System User's Guide and Quick Tour,* 555-027-755

  Online document that provides detailed information and procedures that a CRCS administrator or reservations agent can use to install CRCS, set up and manage CRCS databases, reserve conferences, and perform CRCS system administration.

- *BCS Products Security Handbook*, 555-025-600

  Discusses security risks and measures that help prevent external telecommunications fraud involving Lucent Technologies products.

- *BCS Products: Insights into Securing Against Toll Fraud,* BG9054W

  Provides an individualized learning program on insights into securing against toll fraud involving Lucent Technologies products. Using passwords and monitoring various reports are discussed, among other topics.

- *Lucent Technologies BCS Toll Fraud Overview,* 015-338-100

  Provides an explanation of the industry-wide toll fraud problem and the actions needed to protect your telecommunications system.

Organization

This document contains the following chapters and appendices:

- Chapter 1, "MCU Conference Reservation Tools," discusses the paper-based and PC-based (CRCS) scheduling systems and also the Conference Record screen and the Availability of Ports screens.
- Chapter 2, "MCU System Administration," explains how to perform ongoing administration procedures, such as adding trunks and enabling features.
- Chapter 3, "Checking MCU Performance," provides a list of MCU reports that can be displayed or printed to track MCU performance and the status of trunks.
- Chapter 4, "Conference Control Features," discusses Universal Conference Control (UCC), conference control commands from the MCU-MT, and OpCenter, which is a component of the Conference Reservation and Control System (CRCS).
- Chapter 5, "Diagnosing and Solving Problems," explains how to troubleshoot problems within MCU conferences.
- Appendix A, "Conference Command Summary," discusses various conference scheduling, troubleshooting, and control commands.

- Appendix B, “Feature Descriptions,” provides a detailed explanation of the MCU/CRCS features, including the appropriate administration and relevant interactions and requirements.
- Appendix C, “Administration Commands,” provides a summary of the available MCU administration commands and their associated objects.

This document also has a glossary and index.

Typographic Conventions

Several conventions are used to convey information quickly. These conventions are as follows:

- This *typeface* is used for references to titles of other documents and when referring to fields on screens.
- This **typeface** is used to identify commands and values for fields.
- This `typeface` is used when a word or phrase must be written on a paper form and when a message is returned by the MCU.
- The following icon:

WARNING:

emphasizes information that is important to your safety.

- The following icon:

CAUTION:

indicates information you need to prevent equipment damage.

- The following icon:

NOTE:

identifies additional information pertinent to the text preceding it.

Getting Help

If you need help with the procedures or other information in this document, and if you are a customer in the United States or Canada, be sure to call the Video Technical Center (VTC)/Customer Care Center at 800 242-2121. If you are a customer in Australia or the United Kingdom, call the International Technical Assistance Center (ITAC) at 303 804-3777.

MCU Conference Reservation Tools

1

Overview

This chapter briefly discusses the methods for scheduling MCU conferences. The chapter also discusses the Conference Record Form, which is a tool that allows you to schedule MCU conferences. For complete details on PC-based conference scheduling, refer to the appropriate Conference Reservation and Control System (CRCS) documentation.

Conference Scheduling Methods

There are two ways to schedule and reserve conferences on the MCU: PC-based scheduling and paper-based scheduling.

PC-Based Scheduling System

This optional PC-based scheduling system, the Lucent Technologies Conference Reservation and Control System (CRCS), automates the scheduling tasks and performs conflict resolution to ensure MCU resources are available and that selected endpoints have the ability to participate. It also allows time to be reserved on the MCU in advance of the scheduled conference date, and it automatically activates the conference on the day it is scheduled to begin. For details on CRCS, refer to the appropriate CRCS documentation.

Paper-Based Scheduling System

The paper-based scheduling system tracks concurrent and overlapping conferences. Through this system, the number of MCU ports that are available is known and MCU time can be reserved in advance (more than 24 hours ahead of the planned start time) of the conference date.

The paper-based scheduling system requires accurate records of specific conference information, such as conferees' names and contact numbers, conference mode, date of conference, and start and stop times. A usage chart is included to track MCU resource availability and to prevent overbooking. It also requires a way to remind the agent when to reserve the conference (within 24 hours of the start and stop times).

Conferences are reserved by using the MCU Scheduling Terminal (MCU-ST). If a conference convener calls within 24 hours of the conference, the conference can be reserved while the convener is on the phone (provided the MCU resources and Meet-Me numbers are available for all conferees).

Another option is to dedicate resources of the MCU around-the-clock. This method reserves ports on the MCU so selected locations always have the ability to convene a conference at any time without making a reservation.

Adding and Changing Conferences

The Conference Record form is used to capture the information required by the MCU to reserve a conference.

Conference Record Form

An example of the Conference Record form appears as follows:

```
change conference 1                                    Page  1 of  8

                              CONFERENCE RECORD

     Conference ID: 1                                    Status: inactive
   Conference Name: Exec 384k                  Conference Mode: voice-activated
   Conference Type: _______________
                                                  Cascade Mode: _________
        Billing ID: _________________               Audio Mode: auto-w/upgrade
    Password Scope: ____________                     Bandwidth: 384k_
                                                   Speed Match? y
             Class: reserved___                 Entry/Exit Alert? tone
        Start Time: 13:00                          Warning Alert? tone
         Stop Time: 15:00
   OneNumber Access? n
Continuous Presence? n  Executive Conference? y    Video Mode: H.263/1-w/upgrade
 Site1 Ext: _____  Site2 Ext: _____
 Site3 Ext: _____                      H.263/H.261 Transcoding? n
  Starting Image: executive_
 Force Video Now to: ____________              T.120 Data Mode: none
  Lip Sync  <= 128K: custom  Value: 200_              Far End Camera Control? n
  Lip Sync 224/256K: none
  Lip Sync  >= 336K: auto
```

Figure 1-1. Conference Record Screen (Page 1 of 8)

NOTE:
All of the fields that can appear in the previous form are not shown. Also, all of the fields shown in the previous form do not necessarily appear each time the Conference Record is accessed and used. Refer to the following list for details.

```
add conference next                                    Page  2 of  8

                              CONFERENCE RECORD
CONFEREES
 1:   Type: P64_    MCU-Ext: 55555   Rate: _____   Meet-Me Number: xxxxxxxxxxxxxx
  (Re)Dial? _      App Comp? _  Basic/Enh: basic___     Site Name: Site 1
        Dial Out #1: _________________          Dial-Out #2: _________________
      Password Reqd? y
CONFEREES
 2:   Type: P64_    MCU-Ext: 66666   Rate: _____   Meet-Me Number: xxxxxxxxxxxxxx
  (Re)Dial? _      App Comp? _  Basic/Enh: basic___     Site Name: Site 2
        Dial Out #1: _________________          Dial-Out #2: _________________
      Password Reqd? y
CONFEREES
 3:   Type: P64_    MCU-Ext: _____   Rate: _____   Meet-Me Number: xxxxxxxxxxxxxx
  (Re)Dial? _      App Comp? _  Basic/Enh: basic___     Site Name: Site 3
        Dial Out #1: _________________          Dial-Out #2: _________________
      Password Reqd? y
CONFEREES
 4:   Type: P64_    MCU-Ext: _____   Rate: _____   Meet-Me Number: xxxxxxxxxxxxxx
  (Re)Dial? _      App Comp? _  Basic/Enh: basic___     Site Name: Site 4
        Dial Out #1: _________________          Dial-Out #2: _________________
      Password Reqd? y
```

Figure 1-2. Conference Record Screen (Page 2 of 8)

NOTE:
Pages 2 through 7 of the screen have the same format of four entries per page. However, page 8 has just one entry.

The following list provides descriptions of the fields on the Conference Record form and their related values:

- *Conference ID*

  Display-only field that shows the number that identifies this conference.

- *Conference Name*

  Indicates the purpose of the meeting for identification purposes. Enter a maximum of 20 alphanumeric characters.

- *Billing ID*

  If applicable, enter the billing number for the conference. The number may contain a maximum of 15 alphanumeric characters.

- *Password Scope*

  Entries include **all** and **audio**. **Audio** indicates that the per-conference password is in effect and is required only for an audio-only endpoint. For value **all** or **audio**, the *Password* field on page 1 of the form is displayed. For value **all**, the per-user *Password Reqd?* field on pages 2 through 8 of the form is displayed.

- *Password*

  Enter a maximum of 15 alphanumeric characters.

- *Class*

  There are three conference classes: **reserved**, **dedicated**, and **file**. A **reserved** conference is scheduled to begin within 24 hours of the current time. A **dedicated** conference is always available and can be joined at any time. A **file** conference is filed as a record for future use. The default is **reserved**.

- *Start Time*

  If value **reserved** or **file** is selected in the *Class* field, this field appears so you can enter the time the conference begins. The time is entered in ***hhmm*** format where ***hh*** is the hour (***00*** to ***23***) and ***mm*** is the minutes (***00*** to ***59***). This field is optional for file conferences.

- *Stop Time*

  If value **reserved** or **file** is selected in the *Class* field, this field appears so you can enter the time the conference ends. The time is entered in ***hhmm*** format where ***hh*** is the hour (***00*** to ***23***) and ***mm*** is the minutes (***00*** to ***59***). This field is optional for file conferences.

- *OneNumber Access*

  Use this field to administer the OneNumber Access feature, which allows multiple endpoints to use the same MCU-extension within the same conference. Enter **y** to enable the feature or **n** to disable it.

  NOTE:
  Changing the field value from **n** to **y** can be done only while Dynamic Conference Resizing is activated and the conference is not active. Changing the field value from **y** to **n** is not permitted whenever the conference is active or in use.

  If an MCU-extension is used for OneNumber Access, it cannot be used as a control extension for a Presentation or Broadcast/scan conference.

  If *Executive Conference* is set to **y**, this feature can use only audio-only endpoints.

- *Continuous Presence*

  This field is used to allocate Continuous Presence *Plus* resources to a conference. Whenever the field is set to **y,** the video processor resources are allocated once the conference starts, and the following six fields discussed in this list are displayed. Whenever the field is set to **n**, Continuous Presence *Plus* is not a part of the administered conference.

  NOTE:
  This field cannot be administered unless the *Confinuous Presence/Executive Conf* or *Conferencing Suite* field in the Customer-Options form is set to a value greater than **0**. Also, this field cannot be administered unless the *Executive Conference* field is set to **n**.

  If *Speed Match* contains **y**, the field value cannot be changed from **y** to **n** in an active or in-use conference.

  If *Far End Camera Control* contains **y**, this field must be disabled.

  Whenever *Continuous Presence* is set to **y** and *Speed Match* is set to **n**, the *Lip Sync* field appears.

The following six fields appear whenever the *Continuous Presence?* field is set to **y**.

- *Quad (1-4) Ext*

  These fields are used to assign specific endpoints to a quadrant in the Continuous Presence *Plus* display. To assign an endpoint to a specific quadrant, enter its MCU-extension into the appropriate field. The MCU-extension for the quad must be valid and properly administered. If any field is left blank, the corresponding quadrant is considered “unassigned.” Endpoints are VAS-switched into the unassigned quadrants corresponding to such fields, as appropriate.

  NOTE:
  The MCU-extension for the quad cannot be a UCC, Audio Add-On, or OneNumber Access extension. Also, it cannot be a control extension in a Presentation mode conference.

  If *Continuous Presence* is set to **n**, and if *Executive Conference* is then set to **y**, the extensions that were previously entered into the *Quad (1-3) Ext* fields remain in the *Site (1-3)* fields.

- *Starting Image*

  Specifies the starting video composition for a CPP or Executive Conferencing conference. Values for CPP include **full**, **dual-panoramic**, **lower-panoramic**, **upper-panoramic**, and **quad** (default). Values for Executive Conferencing include **full** and **executive** (default). Value **dual-panoramic** presents a two-image screen layout where the top half of the screen displays one image and the bottom half of the screen also displays one image. Value **lower-panoramic** presents a three-image screen layout where the bottom half of the screen displays one image while the top half of the screen is segmented into two screen displays. Value **upper-panoramic** presents a three-image screen layout where the top half of the screen displays one image while the bottom half of the screen is segmented into two screen displays. Value **executive** presents a dual-panormaic screen layout to the three broadcaster endpoints in the conference, and it presents an upper-panoramic screen layout to the other endpoints in the conference.

  This field appears only whenever the *Continuous Presence* field or the *Executive Conference* field is set to **y**.

- *Force Video Now To*

  This field is used to toggle between three-image, quadrature, dual-screen, and full-screen video within a conference. This field appears only whenever the *Continuous Presence* field or the *Executive Conference* field contains **y**.

  Values for CPP include **full**, **dual-panoramic**, **lower-panoramic**, **upper-panoramic**, **quad**, and blank (default). Values for Executive Conferencing include **full** and **executive** (default). Whenever the Conference Record is submitted, the display change is applied to the conference and remains in effect until it is changed by another change conference command, Universal Conference Control (UCC), or the Conference Reservation and Control System (CRCS).

- *Empty Quadrant Extension*

  This field is used to provide a fill pattern in the Continuous Presence *Plus* display whenever there are fewer than four parties in the conference. The fill endpoint is an endpoint in the conference. This endpoint's MCU-extension is entered into this field. The MCU-extension for the empty quad must be valid and properly administered. If there are only two active endpoints in the conference, the video from the fill endpoint is displayed in the remaining two quadrants.

  NOTE:
  The MCU-extension for the empty quad cannot be a Universal Conference Control (UCC), Audio Add-On, or OneNumber Access extension. Also, it cannot be a control extension in a Presentation mode conference.

- *Executive Conference*

  This field is used to allocate Executive Conferencing resources to the conference. This feature supports only the following rates: **1x56k**, **1x64k**, **2x56k**, **2x64k**, **112k**, **128k**, **224k**, **256k**, **336k**, **384k**, **448k**, **512k**, **672k**, and **768k**. Also, it supports only the H.261 and H.263 video modes; CTX, CTX+, MRV, and SG4 video modes are not supported.

  NOTE:
  This field cannot be administered unless the *Executive Conference, Continuous Presence/Executive Conf,* or *Conferencing Suite* field in the Customer-Options form is set to a value greater than **0**. Also, this field cannot be administered unless the *Continuous Presence* field in the Conference Record form is set to **n**.

  This field cannot be administered unless *Conference Mode* is set to **voice-activated** or **chair**; it cannot be administered for value **presentation** or **broadcast/scan**.

  If this field is set to **y**, the *H.263/H.261 Transcoding*, *Speed Match*, and *Far End Camera Control* fields must be set to **n**; also, the *Entry/Exit Alert* and *Warning Alert* fields can be set to no other value than **tone** or **none**.

  OneNumber Access can be used with this feature only for audio-only endpoints.

  Executive Conferencing cannot be dynamically enabled for an active conference. However, dynamic changes can be made to Sites 1 through 3 within an Executive Conferencing conference.

The *Site (1-3) Ext* fields, which are discussed next, and the *Starting Image* and *Force Video Now to* fields, which were discussed previously, appear whenever the *Executive Conference* field is set to **y**.

- *Site (1-3) Ext*

  These fields are used to assign the broadcaster endpoints (if desired) in an Executive Conferencing display. To assign an endpoint, enter its MCU-extension into the appropriate field. The MCU-extension must be valid and properly administered. If any field is left blank, a fill pattern is provided for the appropriate screen display.

  NOTE:
  Dynamic changes can be made to Sites 1 through 3 within an Executive Conferencing conference.

  If *Executive Conference* is set to **n**, and if *Continuous Presence* is then set to **y**, the extensions that were previously entered into the *Site (1-3) Ext* fields remain in the *Quad (1-3)* fields.

- *Far End Camera Control*

  Enter **y** to enable this feature. Enter **n** to disable it.

  NOTE:
  If the feature is enabled, the *Continuous Presence*, *Executive Conference*, and *T.120 Data Mode* fields must be set to **n**.

- *Lip Sync*

  NOTE:
  This field appears only whenever either *H.263/H.261 Transcoding* or *Continuous Presence* is set to **y** and *Speed Match* is set to **n**.

  Use this field to establish Lip Synchronization mode for either H.263/H.261 transcoded conferences or CPP conferences without SM. One of two modes, Preserved Lip Sync Mode or Audio Priority, may be enabled.

  To enable Preserved Lip Sync Mode, enter one of two values into the field. Value **auto** enables a preselected MCU delay of a length determined by the endpoint type and/or audio standard. Preselected audio delays are defined within and uplinked by the TN2237 circuit pack. Value **custom** enables a delay of **0** to **1000** msecs in 20 msec increments. Entering this value brings up an associated *Value* field, which allows you to enter the appropriate msec value.

  To enable Audio Priority, enter value **none** into the field.

- *Lip Sync <=128K,...224/256K,...>=336K*

  NOTE:
  These fields appear only whenever *Speed Match* is set to **y**.

  Use these fields to establish Lip Synchronization mode for SM conferences at any of the three rate ranges indicated. One of two modes, Preserved Lip Sync Mode or Audio Priority, may be enabled.

  To enable Preserved Lip Sync Mode, enter one of two values into the field. Value **auto** enables a preselected MCU delay of a length determined by the endpoint type and/or audio standard. Preselected audio delays are defined within and uplinked by the TN2237 circuit pack. Value **custom** enables a delay of **0** to **1000** msecs in 20 msec increments. Entering this value brings up an associated *Value* field, which allows you to enter the appropriate msec value.

  NOTE:
  As a default, each *Value* field is automatically populated with the corresponding value administered in the Feature-Related System Parameters form. For details, refer to "Features-Related System Parameters Form" on page 2-43.

  To enable Audio Priority, enter value **none** into the field.

- *Value*

  This field appears whenever any of the fields for Lip Synchronization identified in the previous two bullet items contain value **custom**. The field allows you to enter a Lip Sync delay value in msecs for an MCU conference. Values include **0** through **1000** in increments of 20. (Refer to the previous NOTE.)

- *Status*

  This display-only field reports the current status of a conference as it relates to the *Class* of the conference. A reserved conference can be **active, in-use**, **inactive**, or **complete**. **Active** means the conference is available to receive calls. **In-use** means the conference is in progress and one or more conferees has joined in. **Inactive** means the conference is not available to receive calls (for a reserved conference, the *Start Time* has not been reached). **Complete** means a conference has ended (the *Stop Time* has been reached). A dedicated conference is always **active** (available to receive calls). A file conference is always **inactive** (not available to receive calls).

- *Conference Mode*

  Depending on the MCU model, this field offers up to three conference modes. Value **voice-activated** allows conferees to view the current speaker. The video source changes when another speaker begins talking. Value **chair** allows the chairperson to determine which video-sending endpoint is broadcast to all of the other conferees. Value **presentation** allows one location to be viewed constantly by all conferees. If a conferee asks a question or makes a comment, the other conferees hear the speaker but continue to view the presenter. Value **broadcast/scan** allows one location to be viewed and heard constantly by all conferees. The broadcaster receives video (not audio) from all conferees on a rotating basis. Value **voice-activated** is the default.

  NOTE:
  These field values can be used only if the required field in the Customer-Options form is set to **y**. The following table provides details

Conference Mode Field Value	Required Field(s) in System-Parameters Customer-Options Form
broadcast/scan	*Presentation/Broadcast?*
chair	*Chair Control Suite?*
presentation	*Presentation/Broadcast?*
voice-activated	*VAS?* and/or one or more of the other fields identified in this column

  The *Type* field can be set to **UCC** only if *Conference Mode* is set to

voice-activated or **presentation**.

For values **broadcast/scan** and **presentation**, the *Control Ext* field must be populated correctly.

The field must contain **voice-activated** or **chair** to enable both Speed Match and Continuous Presence *Plus* within a conference.

- *Control Ext.*

  This field appears only when the *Conference Mode* field contains either **presentation** or **broadcast/scan**. The control extension is the MCU-extension of the location that will be the presenter or broadcaster. This extension must be valid and administered properly. The control extension cannot be a OneNumber Access, Audio Add-On, or UCC extension.

  NOTE:
  This field must be populated correctly whenever the *Conference Mode* field is set to **presentation** or **broadcast/scan**.
  If *Cascade Mode* is set to **primary** or blank, the field must contain an extension for an H.320 (*Type* is set to **P64**) or a BONDed (*Type* is set to **BOND**) endpoint. If *Cascade Mode* is set to **secondary**, the field must contain an extension for a cascaded (*Type* is set to **CAS**), Dedicated Access (*Type* is set to **DA**) or BONDed-cascaded (*Type* is set to **BCAS**) endpoint.

- *Scn Int.*

  This field appears only when *Conference Mode* contains **broadcast/scan**. This is the amount of time (in seconds) that the broadcaster will view each conferee before the MCU switches to the next conferee. The default is **15** seconds.

- *Cascade Mode*

  This mode must be either **primary** or **secondary** if the conference includes a CAS link and/or if a cascaded (*Type* is set to **CAS**) or BONDed-cascaded (*Type* is set to **BCAS**) endpoint is being added to the conference. On the other hand, if a cascaded endpoint cannot be administered, or if there are no cascaded or BONDed-cascaded endpoints in the conference, the mode must be **blank**. Finally, if the appropriate endpoints can be administered but the link is not enabled, the mode is **disabled**.

  NOTE:
  This field is not displayed unless the *Cascading* field in the Customer-Options form is set to **y**.

  The field cannot be set to **disabled**, **primary**, or **secondary** (and should be left blank) if *T.120 Data Mode* contains **any-mlp**.

- *Audio Mode*

  The default is **auto-w/upgrade**.

  If the Lucent Technologies ClearPresence™@716 Audio Coder right-to-use (RTU) option *is not enabled*, the default value indicates that the MCU will automatically select either the **G.711**, **G.722** or **G.728** audio mode, depending on the bandwidth of the conference and of the participating video endpoints. (These three rates are fixed values.) Also, this value indicates that the MCU will automatically upgrade a conference where the audio mode has been downgraded whenever the endpoint causing the downgrade drops out of the conference. Value **auto-down_only** indicates that the MCU will upgrade the audio mode only of a conference that has been downgraded and after all endpoints drop.

  To establish the endpoint as a ClearPresence @716 endpoint, enter value **ClearPres716**.

  NOTE:
  Value **ClearPres716** cannot be entered unless the ClearPresence @716 RTU is enabled on the Customer-Options form.

  Refer to Table B-11 on page B-64 for a description of the per-conference audio and video capabilities received by endpoints per the ClearPresence @716 RTU and other factors for all valid *Audio Mode* field values.

  The field value and whether or not the ClearPresence @716 RTU is optioned determine which audio modes are eligible for changing in Dynamic Resizing conferences. For details, refer to Table B-12 on page B-66 and Table B-13 on page B-66.

  For T.120 conferences, if the field contains either **G.711** or **G.722**, the *MLP Rate* field must contain either **6.4k** or **var_mlp**. Also, for such conferences, if the field contains either **auto-w/upgrade** or **auto-down_only**, the *MLP Rate* field must contain **6.4k**. Finally, for T.120 conferences, the field must contain **G.728** or **ClearPres716** if the *MLP Rate* field contains a value *other than **6.4k***.

  Speed-matched T.120 Data Conferencing conferences are allowed with audio mode **ClearPres716** and MLP rate “fixed” (and not var_ MLP).

- *Bandwidth*

  This field is used to specify the bandwidth selected for the conference. If **1x56k** is selected, the conference will be a 1-channel 56k conference. If **1x64k** is selected, the conference will be a 1-channel 64k conference. If **2x56k** is selected, the conference will be a 2B-channel 56k conference (112k bandwidth). If **2x64k** is selected, the conference will be a 2B-channel 64k conference (128k bandwidth). If **384k** is selected, the conference will be a 1-channel **384k** (H0 or BONDed) conference. Other 1-channel bandwidths include **112k, 128k, 168k, 192k, 224k, 256k, 280k, 320k, 336k, 448k, 512k, 672k, 768k, 1008k, 1152k, 1288k, 1344k, 1472k, 1536k, 1680k,** and **1920k**. Default is **2x56k**.

  NOTE:
  The field cannot be administered unless the corresponding field within the Maximum Port Capacity area of the Customer-Options form has a value greater than 0. Also, sufficient resources for the endpoint type must be available for the requested time period. The following bandwidths are not allowed in the conference if *Type* contains **BCAS** or **BOND**: **1x56k**, **1x64k**, **2x56k**, **2x64k**, or a bandwidth *greater than* ***768k***.

  The *Lo-/Hi-Interworking* field *cannot* be administered with the following bandwidths: **1x56k**, **1x64k**, **2x56k**, and **2x64k**.

  Speed Match (SM) and Continuous Presence *Plus* (CPP) support only the following rates: **1x56k**, **1x64k**, **2x56k**, **2x64k**, **112k**, **128k**, **224k**, **256k**, **336k**, **384k**, **448k**, **512k**, **672k**, and **768k.**

- *Speed Match*

  Controls the allocation of speed-matching resources for the conference. Values include **n** (default) and **y**. The field must be set to **n** whenever the *Cascade Mode* field contains **secondary**.

  The field cannot be administered unless the *Speed Match* or *Conferencing Suite* field in the Customer-Options form is set to a value in the range **1** through **8**. Sufficient speed-matching resources must be available for the requested time period.

  Also, the field cannot be administered unless various fields in the Conference Record form are populated correctly. The following list identifies these fields along with the appropriate values for Speed Match, and it provides any necessary commentary:

  — *Status* — **inactive** (The field cannot be set for active or in-use conferences.)

  — *Bandwidth* — **1x56k**, **1x64k**, **2x56k**, **2x64k**, **112k**, **128k**, **224k**, **256k**, **336k**, **384k**, **448k**, **512k**, **672k**, or **768k**

  — *Executive Conference* — **n**

— *Video Mode* — Values for H.261-associated video modes, including **H.261-down_only**, **H.261-w/upgrade**, **H.261-CIF-Floor**, **H.261-CIF-Now**, or **H.261-QCIF-Now**

— *CIF Frame Rate* — **7.5 fps**, **10 fps**, **15 fps**, or **auto**

— *QCIF Frame Rate* — **7.5 fps**, **10 fps**, **15 fps**, or **auto**

— *MLP Rate* — If T.120 Data Conferencing is enabled, and if the ClearPresence @716 RTU is not optioned, this field must contain one of the following values: **6.4k**, **8k**, **14.4k**, **16k**, **22.4k**, **24k**, **30.4k**, **32k**, or **38.4k**. If the ClearPresence @716 RTU is optioned, this field can contain **40k**, **46.4k**, or any of the rates indicated in the previous sentence. However, for this RTU, whenever *Audio Mode* contains **auto-w/upgrade** or **auto-down_only**, only MLP rate **6.4k** is permitted. Variable rates (indicated by value “var_mlp”) are not allowed.

The following points regarding the Conference Record form also apply whenever you are enabling Speed Match:

— OneNumber Access is supported but only for Audio Add-On endpoints. Multimedia endpoints are not supported.

— UCC and Audio Add-On endpoints cannot be speed matched. Therefore, if the *Type* field on the form contains **UCC** or **AUD**, the corresponding *Rate* field must be blank.

— Rate of the Dedicated Access (DA) endpoints must be compatible with the administered endpoint rate. The rate of the specified access endpoints for the dial-out numbers contained in the *Dial-Out #1* and *Dial-Out #2* fields must be equal to or greater than the rate at which the endpoint is administered.

— 56k endpoints cannot be speed matched with Nx64k endpoints. Therefore, if *Rate* contains an Nx64k value, *MCU-Ext* cannot contain a value representing a 56k endpoint.

— *Type* field must contain **BOND** or **BCAS** if the *Rate* field contains **112k**, **128k**, **224k**, **336k**, **448k**, **672k** or **768k**, with one exception: the field can contain **Px64** for RS366 dialing.

— If the *Rate* field contains **2x56k** or **2x64k,** the *Dial-Out #2* field as well as the *Dial-Out #1* field must be populated. If the *Rate* field does not contain either of these values, the *Dial-Out #2* field must be blank.

— If the ClearPresence @716 RTU is optioned, G.728 and G.722 can interwork in Speed Match conferences. Endpoints joining at a transfer rate less than or equal to 2B can experience G.728 audio; endpoints joining at a transfer rate greater than 2B transfer can experience G.722 audio.

NOTE:
Whenever *Speed Match* is set to **y**, the following fields relevant to Lip Synchronization appear on the form: *Lip Sync <= 128K*, *Lip Sync 224/256K*, and *Lip Sync >= 336K*.

- *Entry/Exit Alert*

  This field appears if your MCU model has the Notification Package feature. The field indicates via tones and/or on-screen display messages when conferees join or exit the conference. Enter **tone** (default) to provide the traditional audible tone. Enter **text** to provide the text message. Enter **ANI** to display the Automatic Number Information (ANI) code of the endpoint for the associated conferee. Enter **text/tone** to provide both the text message and the audible tone. Enter **ANI/tone** to provide both an audible tone and the ANI code of the endpoint for the associated conferee.

  NOTE:
  Values **text**, **text/tone, ANI,** and **ANI/tone** cannot be administered unless the conference is a non-SM CPP conference. Therefore, the *Speed Match* field must contain **n** and the *Continuous Presence* field must contain **y**. Also, these values cannot be administered whenever *Executive Conference* is set to **y**.

  Value **text/tone** for entry/exit alert is not allowed if OneNumber Access is being used in the conference for either Audio Add-On or multimedia endpoints.

  Enter **none** to provide no notification of a conferee joining or exiting the conference.

  NOTE:
  None of the values for the field can be changed dynamically during an active or in-use conference.

- *Warning Alert*

  This field appears if your MCU model has the Notification Package feature and if *Class* contains **reserved** or **file**. Enter **tone** (default) to provide the traditional 10-minute audible warning tone. Enter **text** to provide the text message for both the 10-minute and two minute warnings. Enter **text/tone** to provide both the text message and the audible tone for the 10-minute warning and just the text message for the two-minute warning.

  NOTE:
  Values **text** and **text/tone** cannot be administered unless the conference is a non-SM CPP conference. Therefore, the *Speed Match* field must contain **n** and the *Continuous Presence* field must contain **y**. Also, these values cannot be administered whenever *Executive Conference* is set to **y**.

Enter **none** to provide no conference end notification.

NOTE:
None of the values for the field can be changed dynamically during an active or in-use conference

If the conference end time is extended dynamically during a conference in response to the 10-minute notification, and if the field contains a valid value other than **none**, a subsequent 10 minute notification followed by a two-minute notification are provided starting 10 minutes prior to the newly established conference end time.

- *Rate Adaptation*

This field appears if the conference has a 2x64k bandwidth. It indicates whether or not a 56k/64k rate adaptation is allowed for the conference. Values include **yes** and **no**.

NOTE:
The field cannot be enabled whenever *T.120 Data Mode* is set to **any-mlp**. Whenever *Cascade Mode* contains a non-blank value or *Video Mode* contains a non-H.261- or non-H.263-associated value, the field must be set to **n**. If Dynamic Resizing is enabled while the conference is active, the field cannot be changed from **y** to **n**. Finally, the field cannot be changed while the conference is active and Dynamic Resizing is not enabled.

- *Lo-/Hi-Interworking*

This field indicates whether or not the conference supports Low-/High-Speed Interworking. Values include **yes** and **no**.

NOTE:
The field cannot be changed from **y** to **n** whenever the conference is active or in use.

The field *cannot* be administered with the following bandwidths: **1x56k**, **1x64k**, **2x56k**, and **2x64k**.

- *Video Mode*
  - Values include **H.261-down_only, H.261-w/upgrade, H.261-CIF-Floor, H.261-CIF-Now, H.261-QCIF-Now, H.CTX, H.CTX+, SG4, H.263/1-down_only, H.263/1-w/upgrade,** and **MRV-w/upgrade.** The default is **H.261-w/upgrade**. This value automatically upgrades a conference where the video mode has been downgraded whenever the endpoint causing the downgrade drops out of the conference. Value **H.261-down_only** upgrades the video mode of the conference only after all endpoints drop from the conference. **H.261-CIF-Floor** forces the video format to Common Intermediate Format (CIF). A CIF Frame Rate floor may be specified via administration. **H.261-CIF-Now** forces the video to CIF, and it forces the CIF Frame Rate to a value specified via administration. **H.261-QCIF-Now** forces the video to Quarter Common Intermediate Format (QCIF), and it forces the QCIF Frame Rate to a value specified via administration. **H.CTX** requires a conference bandwidth *not greater than* 384k. **H.CTX+** requires a conference bandwidth *not less than* 336k. **H.263/1-down_only** indicates the highest common video mode that allows either its frame rate or video to be downgraded to H.261 video mode. Whenever an endpoint that caused a video downgrade disconnects from the conference, no attempt is made to upgrade video quality. This video mode can downgrade to H.261. The format and associated frame rates for the H.261 and H.263 video modes can also be downgraded. **H.263/1-w/upgrade** indicates the highest common video mode that allows either its frame rate or video to be either downgraded (to H.261) or upgraded (to H.263) depending on the capabilities of the endpoints currently in the conference. This video mode can downgrade to H.261. The format and associated frame rates for the H.261 and H.263 video modes can also be downgraded.

NOTE:
H.CTX or **H.CTX**+ appears in this field only when the port capacity for the appropriate one of these video modes is nonzero.

Whenever SG4 is optioned (via an administered nonzero port capacity), the value **SG4** appears in this field.

NOTE:
H.263/1-down_only, H.263/1-w/upgrade**,** H.261-down_only, H.261-w/upgrade, H.261-CIF-Now, H.CTX, H.CTX+, and SG4 can be set only via administration. H.261-CIF-Floor and H.261-QCIF-Now can be set via administration and conference control.

Speed Match is supported only by H.261-associated video modes.

The field *cannot* be set to **H.263/1-w/upgrade** whenever *Continuous Presence* is set to **y**.

Dynamically changing from H.261 video mode to H.263 video mode is allowed only in non-SM full-screen conferences.

- *H.263/H.261 Transcoding*

  Enables H.263/H.261 transcoding for endpoints operating at the same transfer rate. Values include **y** and **n** (default). Value **y** indicates that H.263- and H.261-associated endpoints can coexist within the same conference. Value **n** indicates that H.263 and H.261 endpoints cannot coexist; therefore, the conference is downgraded to H.261 video mode if an endpoint having only H.261 video mode capability joins the conference at that rate.

  NOTE:
  The *Speed Match* and *Conferencing Suite* fields within the Customer Options form can be used to enable full-screen H.263/H.261 transcoding.

  This field appears only whenever the *Video Mode* field contains **H.263/1-down_only** or **H.263/1-w/upgrade**.

  The field can be administered only if the appropriate video processor resources are allocated via the *Speed Match* or *Conferencing Suite* field in the Customer-Options form and if both the *Continuous Presence* field and *Speed Match* field in the Conference Record form *do not contain* **y**. Also, the field must be set to **n** whenever *Executive Conference* is set to **y**.

  If the field is activated, the *Lip Sync Setting* field appears.

  Changing the field from **y** to **n** de-allocates the video resources for full-screen non-SM conferences. Changing the field from **n** to **y** for in-use conferences is not allowed.

  Whenever *H.263/H.261 Transcoding* is set to **y**, the *Lip Sync* field appears.

- *T.120 Data Mode*

  If this field contains **any-mlp**, the MCU opens a var-MLP channel to each endpoint in the conference. If this field contains **none**, the MCU does not open an MLP channel to any of the endpoints in the conference.

  NOTE:
  If the field contains **any-mlp**, the *T.120* field in the Customer-Options form must be set to a value greater than **0**. If the *T.120* field is set to **0**, the *T.120 Data Mode* field can be set only to **none**.

  The field cannot be changed while the conference is active or if Dynamic Resizing is enabled while the conference is active.

  If *Far End Camera Control* contains **y**, this field must be disabled, or vice versa.

- *MLP Rate*

  Indicates the data rate for T.120 Data Conferencing. Values include **var-mlp**, **6.4k, 8k, 14.4k, 16k, 22.4k, 24k, 30.4k, 32k, 38.4k, 40k,** and **46.4k**.

  NOTE:
  This field cannot be changed while the conference is active.

  If the ClearPresence @716 RTU is not optioned, MLP rates **40k** and **46.4k** for T.120 Data Conferencing are not allowed in Speed Match conferences. If the ClearPresence @716 RTU is optioned, these rates are allowed. However, rate **var-MLP** is not allowed in either case.

  For T.120 conferences, if the *Audio Mode* field contains **G.711**, **G.722**, **auto-w/upgrade**, or **auto-down_only**, the *MLP Rate* field must contain **6.4k**.

- *CIF Frame Rate*

  This field appears whenever *Video Mode* is set to **H.261-CIF-Now** or **H.261-CIF-Floor**. The field is used to specify the appropriate MPI rate. Values include **7.5 fps** (frames per second), **10 fps**, **15 fps**, **30 fps**, or **auto**. Value **auto** causes a negotiation to the highest frame rate.

  NOTE:
  Value **30 fps** is not supported in Speed Match conferences.

- *QCIF Frame Rate*

  This field appears whenever *Video Mode* is set to **H.261-QCIF-Now**. The field is used to specify the appropriate MPI rate. Values include **7.5 fps**, **10 fps**, **15 fps**, **30 fps**, or **auto**. Value **auto** causes a negotiation to the highest frame rate.

  NOTE:
  Value **30 fps** is not supported with Speed Match.

- *Type*

  This field indicates the type of conferee. Possible values include **P64** (multimedia), **AUD** (audio-only), **CAS** (cascade link), **BOND** (BONDing), **DA** (Dedicated Access), **BCAS** (cascade BONDing), and **UCC** (Universal Conference Control). Dedicated Access endpoints include DS1 and data interface (DI) endpoints. DI endpoints include EIA-530, RS449, and V.35 endpoints. For access endpoints that support RS366 dialing, value **DA** or **P64** is valid. Also, these endpoints may participate in conferences either as dial-in or (for Dedicated Access endpoints only) dial-out conferees.

NOTE:
This field must be populated if *MCU-Ext* contains a value.

Some of the field values can be used only if the required field(s) in the Customer-Options form are populated correctly. The following table provides details.

Type Field Value	Required Field(s) and Corresponding Value in System-Parameters Customer-Options Form
AUD	*Audio Add-On*: valid value *greater than* ***0***
BCAS	--- At least one of the *BONDing...* fields: valid value *greater than* ***0*** --- *Cascading*: **y**
BOND	At least one of the *BONDing...* fields: valid value *greater than* ***0***
CAS	*Cascading*: **y**
UCC	*UCC:* valid value *greater than* ***0***

Value **UCC** is valid for no more than one extension within a conference. Also, the field can be set to **UCC** only if *Conference Mode* is set to **voice-activated** or **presentation**. Whenever *Cascade Mode* contains **secondary**, value **CAS** or **BCAS** is valid for no more than one extension within the secondary MCU.

If the field contains **AUD** or **UCC**, the associated extension must be an audio-only extension. If the field does not contain either of these values, the associated extension cannot be an audio-only extension. The following bandwidths are not allowed in the conference for value **BCAS** or **BOND**: **1x56k**, **1x64k**, **2x56k**, **2x64k**, or a bandwidth *greater than* ***768k***.

- *MCU-Ext*

  An MCU-extension is required for each conferee in a conference unless the OneNumber Access feature is enabled. The extension must be valid and administered properly.

  NOTE:
  A different MCU-extension is required whenever *Type* contains **DA**, **BOND**, **CAS**, **BCAS**, or **UCC**.

  With Speed Match enabled, OneNumber Access is allowed only for MCU-extensions for audio-only endpoints. If an MCU-extension is used for OneNumber Access, it cannot be used as a control extension for a Presentation or Broadcast/scan conference.

  Different MCU-extensions must be used whenever *OneNumber Access* contains **n**. Also, MCU-extensions (with or without OneNumber Access) cannot be changed even with Dynamic Resizing enabled.

- *Rate*

  Specifies the rate at which the endpoint is joining the Speed Match conference. Values include **1x56k**, **1x64k**, **2x56k**, **2x64k**, **112k**, **128k**, **224k**, **256k**, **336k**, **384k**, **448k**, **512k**, **672k**, **768k**, and blank.

  The MCU allows a maximum of four rates at which an endpoint can join a Speed Match conference. Any rate entered into this field that exceeds this total is denied.

  NOTE:
  This field must be blank if *Speed Match* is set to **n**. The field must also be blank for UCC or Audio Add-On endpoints. The field may be blank for endpoints at the primary transfer rate as indicated in the *Bandwidth* field.

  If *Speed Match* is set to **y**, and if *Type* is set to **BOND** or **BCAS**, the field can be set only to one of the following rates: **112k**, **224k**, **336k**, **448k**, or **672k**. Also, if *Speed Match* is set to **y**, and if the field contains an Nx64k rate, 56k endpoints are not allowed.

- *Meet-Me Number*

  Whenever an MCU-extension has been entered in a previous Conference Record administration session, the MCU automatically displays the data entered on the MCU-Extension form for the associated Meet-Me Number field. The Meet-Me Number is usually the network number that the conferee dials to join a conference or the location of the conferee.

- *(Re)dial?*

  Values include **y**, **n** and blank. Value **n** indicates that the MCU will not dial out at the start of the conference. The field value can be changed if Dynamic Resizing is optioned. The field should be blank whenever the conference is not active or not in use, or if *Class* is set to **file**.

- *App Comp*

  Indicates the “Application Compliant” flag. If this flag is set to **y(es)**, and if the relevant endpoints support T.120 Data Conferencing and G.728 (LD-CELP), the MCU opens an MLP channel to the endpoint. If the flag is set to **n(o)**, and if the relevant endpoints support var-MLP and G.728 (LD-CELP), the MCU opens an MLP channel to the endpoint, but data sent to or received from the endpoint is blocked.

  NOTE:
  If the conference is active or in use, but Dynamic Resizing is not enabled, no changes can be made to the field. Also, if the conference is active, but Dynamic Resizing is not enabled, associated endpoints cannot be added or deleted.

  The field must be blank whenever *Type* and *MCU-Ext* are blank, or whenever *Type* is set to **AUD** or **UCC**. The field must be set to either **y** or **n** whenever *T.120 Data Mode* is set to a value *other than **none***. Finally, the fields for all the endpoints using the same MCU-extension (to enable OneNumber Access, for example) must have the same settings.

- *Basic/Enh*

  Value **basic** indicates that the endpoint supports only basic BAS commands/caps. Value **enhanced** indicates that the endpoint supports both enhanced and basic BAS commands/caps.

  NOTE:
  Value **enhanced** is required for at least one endpoint in a non-cascaded, Chair conference. This usually applies to the Chair conferee.

  The fields for all the endpoints using the same MCU-extension (to enable OneNumber Access, for example) must have the same settings.

- *Site Name*

  Name used to identify the user of the endpoint. Enter a maximum of 15 alphanumeric characters. Default is **Site *n***, where ***n*** is the conferee ID number (for example, **Site 3**).

  NOTE:
  This field may be blank except whenever *Entry/Exit Alert* is set to **text** or **text/tone**.

  This field can be overwritten by CRCS.

- *Dial-Out #1, Dial-Out #2*

  Network numbers that the MCU dials. These numbers are assigned to the endpoints. A maximum of 48 entries is allowed. The value must contain only digits **0**-**9**, *****, **#**, **~p** (pause), **~w** (wait), and/or **~m** (mark), and it must begin with a valid FAC, TAC, or extension.

  NOTE:
  If the Dial-Out feature is not enabled, these fields should be left blank.

  If *Dial-Out #2* is set, *Dial-Out #1* should also be set.

  The fields should not be populated with MCU-extensions that are repeated within a conference. Such MCU-extensions cannot be assigned as Dial-Out numbers.

  If *Bandwidth* and/or *Rate* contain **2x56k** or **2x64k**, *Dial-Out #2* must be specified. If *Bandwidth* and/or *Rate* contain a value other than **2x56k** or **2x64k**, *Dial-Out #2* should be left blank.

  If *Type* contains **AUD**, *Dial-Out #1* can contain a dial-out Audio Add-On number. If *Type* contains **DA**, *Dial-Out #1* must contain an access endpoint extension. If *Type* is **DA** and *Dial-Out #1* (or *#2*) contains a DI access endpoint extension, the value must be compatible with the conference bandwidth, and it must not exceed the system bandwidth limit for DI ports. If *Type* contains **DA** and the call occupies two channels, *Dial-Out #2* must contain a unique access endpoint extension. If *Type* contains **DA** and *Dial-Out #1* and *Dial-Out #2* contain access endpoints, these endpoints must be the same type of access endpoint (either DS1, V.35, RS-449, or EIA-530).

  Whenever *Speed Match* is set to **y**, the rate of the Dedicated Access (DA) endpoints must be compatible with the administered endpoint rate. The rate of the specified access endpoints for the dial-out numbers contained in the *Dial-Out #1* and *Dial-Out #2* fields must be equal to or greater than the rate at which the endpoint is administered.

- *Password Reqd?*

  Value **y** indicates that a per-conference password value is required. Value **n** indicates that such a value is not required.

MCU System Administration

2

Overview

This chapter describes the ongoing administration tasks performed by a telecommunications or video manager or an on-site MCU system administrator. The following table identifies these tasks alphabetically and provides the appropriate page reference.

Table 0-1. Checklist of MCU Administration Tasks

MCU Administration Task	Page Reference
Administering.....	
----- AAR and ARS Digit Analysis	page 2-159
----- AAR and ARS Digit Conversion	page 2-169
----- Access endpoints	page 2-196
----- Alphanumeric dialing	page 2-189
----- Automatic Alternate Routing	page 2-153
----- Automatic Route Selection	page 2-154
----- Call Detail Recording	page 2-39
----- Cascades	page 2-191
----- ClearPresence @716 Audio Coder	page 2-255
----- Continuous Presence *Plus*	page 2-221
----- Country options	page 2-46
----- Customer options	page 2-22
----- Data communications	page 2-190

Table 0-1. Checklist of MCU Administration Tasks — *Continued*

MCU Administration Task	Page Reference
----- Data modules ---------- 7500 ---------- Network control channel ---------- Processor ---------- Processor interface	 page 2-97 page 2-67 page 2-101 page 2-76
----- Dial-Out	page 2-36
----- Dial plans	page 2-153
----- Direct Connect endpoints	page 2-96
----- DS1 or UDS1 circuit packs	page 2-55
----- ESM	page 2-149
----- Executive Conferencing	page 2-230
----- Far-End Camera Control	page 2-245
----- Feature access codes	page 2-156
----- Feature-related system parameters	page 2-42
----- Full Tie-Trunk Capabilities Using BRI Trunks	page 2-252
----- H.263/H.261 Transcoding	page 2-240
----- International-based parameters	page 2-46
----- ISDN-BRI Trunk Interface	page 2-245
----- ISDN-PRI over PACCON	page 2-90
----- ISDN-PRI trunk groups	page 2-121
----- ISDN D-channels	page 2-76
----- ISDN services (new)	page 2-52
----- Lip Synchronization	page 2-235
----- Logins	page 2-13
----- Maintenance-related system parameters	page 2-90
----- Meet-Me numbers and MCU-extensions	page 2-104
----- Modem pool groups	page 2-225
----- Network facilities	page 2-52
----- Network service and access parameters	page 2-53
----- On-Screen Display	page 2-242
----- OneNumber Access	page 2-234
----- Route patterns	page 2-183

Table 0-1. Checklist of MCU Administration Tasks — *Continued*

MCU Administration Task	Page Reference
----- RS366 Dialing	page 2-244
----- Signaling groups	page 2-86
----- Speed Match conferences	page 2-237
----- Synchronization	page 2-67, page 2-206
----- System features	page 2-42
----- T.120 Data Conferencing	page 2-232
----- Tie trunk groups	page 2-135
----- Trunk groups	page 2-120
----- Uniform Dial Plan	page 2-155, page 2-112
----- Video modes	page 2-190
----- World Class BRI	page 2-255
Assigning.....	
----- MCU-Extensions	page 2-114
Changing.....	
----- Administration permissions	page 2-18
----- Passwords	page 2-8
Checking.....	
----- Access endpoints	page 2-213
----- MMI circuit packs (status)	page 2-218
----- Video Processor circuit packs (status)	page 2-223
Collecting.....	
----- Endpoint data	page 2-11
Establishing.....	
----- Administered connections	page 2-73
----- Network control channels	page 2-67
Logging in from.....	
----- MCU-MT	page 2-7
----- MCU-ST (remotely)	page 2-7
Performing.....	
----- System backups	page 2-217

Table 0-1. Checklist of MCU Administration Tasks — *Continued*

MCU Administration Task	Page Reference
----- Video endpoint inventories	page 2-10
Recording.....	
----- Meet-Me Numbers and MCU-Extensions	page 2-12
Saving.....	
----- Translations	page 2-217
Setting.....	
----- Class of Restriction	page 2-30
----- Class of Service	page 2-34
----- System clock	page 2-9
Supporting.....	
----- MCU-ST, system printer, or CRCS	page 2-159

NOTE:
The Conference Record form and/or the Customer-Options form are required to administer some of the features and capabilities identified in the previous list. Refer to the "Conference Record Form" section in Chapter 1 of this document for details on the Conference Record form, and refer to the "Customer-Options Form" section in this chapter for details on the Customer-Options form.

Diagnostic and maintenance procedures are described in the MCU Maintenance document. Chapter 5, "Diagnosing and Solving Problems" provides conference troubleshooting procedures.

Using the MCU Management Terminal

The MCU Management Terminal (MCU-MT) is a 715GBCS-2 Terminal that is primarily used to perform administration and report functions, but can also be used to reserve conferences and to help diagnose problems reported about conferences.

A maximum of four MCU-MTs can be used simultaneously to issue administration and maintenance commands; however, each command must access a unique record. Therefore, for example, the **add conference 100** command and the **change conference 100** command cannot be issued simultaneously.

Keys and Functions

The keyboard of the MCU-MT is equipped with cursor keys, transaction keys and editing keys to enable data to be sent and received from the MCU. These keys and their functionality are described in the following list:

- **Cursor Keys**

  Used to move the cursor between fields and pages on a screen form. The cursor must be positioned in a field on a form before information can be added or changed in that field.

- **Editing Keys**

  BACKSPACE key and CLEAR are used to edit data in a field. BACKSPACE erases the character at the current cursor position. CLEAR must be used in combination with SHIFT to clear all data in a field. The cursor must be in the field you want to clear before CLEAR and SHIFT are pressed. You can use TAB to advance from field to field or use SHIFT and TAB together to reverse the direction.

- **Transaction Keys**

  CANCEL, ENTER, and HELP perform special functions. CANCEL erases a form or command. ENTER stores data from the screen into the system's memory. HELP displays more information about the values or commands that can be entered for a particular field or form.

  PAGE UP and PAGE DOWN display the previous page and the next page of screens that have more than one page.

Logins and Permissions

The MCU-MT and MCU Scheduling Terminal (MCU-ST) use the following customer logins (shown in hierarchical order).

Table 0-2. MCU Logins and Default Passwords

User	Login	Permissions
System administrator using MCU-MT	cust	Display administration and maintenance data Request system measurements Administer maintenance alarm terminals Administer trunks Administer MCU-extensions Administer features Administer conference records
System administrator using an MCU-MT or remote MCU-ST	rcust	Same as cust login
Reservations agent using an MCU-MT or remote MCU-ST	agent	Display administration data Administer Meet-Me numbers Administer conference records
Display only (using an MCU-MT or remote MCU-ST)	browse	Display administration and maintenance data

Logging In from the MCU-MT

To log in from the MCU-MT, follow these procedures:

1. Verify that the MCU-MT displays: *Login:*
2. Enter **cust**.
3. Verify that the screen displays: *password:*
4. Enter **your password**.

   NOTE:
   For security purposes, your password is not displayed as you enter it. The MCU verifies the login and password. If an invalid login or incorrect password name is entered, the screen displays: *login incorrect:*

5. The software version is displayed.
6. When the screen displays: *Terminal Type (Enter 513, 715, 4410, or 4425): [513],* enter **715** press RETURN.
7. The MT should now display: *Command:*

Logging In Remotely from the MCU-ST

You can perform administration and report functions from the remote MCU-ST. To log in to the MCU-ST, follow these procedures:

1. Enter **at**.
2. When the *OK* prompt appears, enter **atdt*code*** where ***code*** is the access code provided when the MCU is installed.
3. Verify that the MCU-ST displays: *Login:*
4. Enter **rcust**.
5. Verify that the screen displays: *password:*
6. Enter **your password**.

   NOTE:
   For security purposes, your password is not displayed as you enter it. The MCU verifies the login and password. If an invalid login or incorrect password name is entered, the screen displays: *login incorrect:*

7. The software version is displayed.
8. When the screen displays: *Terminal Type (Enter 513, 715, 4410, or 4425): [513],* enter **715** and press RETURN.
9. The MT should now display: *Command:*

Changing Passwords

To log in to the MCU and to change the associated passwords, follow these procedures:

1. Log in as **cust** on the MCU-MT.
2. The MT should now display: *Command:*
3. To change the password, enter **change password *login*** where ***login*** is **cust**, **rcust**, **agent**, or **browse**.

   NOTE:
   If it is necessary to void or deny access to a particular login, the login may be voided by changing the password of the login to **VOID**. After the login's password is **VOID**, the login is denied access to the MCU. The **VOID** password can then be changed or reinstated by a higher-level login (the order given in Table 2-1).

4. The Password Change form is displayed.
5. At the *Your Current Password* field, enter the current password you used to log in and press RETURN.
6. In the *New Password For Login Name* field, enter the new password (see **NOTE**) and press RETURN.

   NOTE:
   Valid passwords have four to seven alpha or numeric characters or a combination of both.

7. Reenter the new password in the *New Password (enter again)* field and press ENTER.
8. Verify that the MT displays: *Command successfully completed* followed by *command:*
9. To change the passwords for the other logins, repeat these steps.

Setting the System Clock

The time set on the MCU is critical for reserving conferences. In areas of the country that change to daylight savings time, the system clock must be adjusted by one hour twice a year (advance one hour for daylight savings time; back one hour for standard time).

The 24-hour conference scheduling system uses the MCU system clock to know when to begin and end conferences. If the MCU is set for the wrong time, it may restore conference records from backup memory that are incorrectly scheduled. If the optional CRCS is installed, the date and time on the MCU and CRCS must be synchronized. Also, for a conference involving T.120 data, the clock on the ESM must be synchronized with the clock on the MSM via a heartbeat message sent by the MSM to the ESM.

The system clock is affected only when power is interrupted for more than two minutes. The optional Uninterrupted Power Supply (UPS) may be used to protect against power outages to the MCU.

Set Date and Time Form

The date and time are set using the Set Date and Time Form. To set the date and time, follow these procedures:

NOTE:
Be sure to obtain the accurate time by calling the National Bureau of Standard Time at **202 844-1234**.

1. At the *command:* prompt, enter **set time**.
2. At the *Day of the Week* field, enter the day of the week.
3. At the Month field, enter the current month.
4. In the *Day of the Month* field, enter the current day (1 through 31).
5. In the *Year* field, enter the current year.
6. In the *Hour* field, enter the current hour in military time. (00 is midnight, noon is 12, and 23 is 11 p.m. The p.m. time is hour plus 12.)
7. In the *Minute* field, enter the current minute (0 through 59).

   NOTE:
   Seconds are reset to 00 for the minute selected.

8. Press ENTER.
9. When the MT displays: *Command successfully completed* followed by *command*:, enter **display time** to verify the setting.

Performing Endpoint Inventories

Performing an inventory of features and functionality for video endpoints that will participate in multipoint conferences helps to provide the optimum MCU video conference experience. Currently, video endpoints support different features offered in the H.320 standards. In some cases, such as with the Lucent Technologies Group Video System 4000, additional H.320 functionality has been added with each new software release of the same model. For example, the model 4000 software version 3.0P.03 does not support all the H.320 features supported in version 4.1P.

The following data should be collected for all video endpoints that will be joining conferences on the MCU:

NOTE:
There is no official ITU-T conformance testing body, nor is there a guide to arbitrate different interoperations of the standards by manufacturers of video products. Thus, as manufacturers implement the standards or add features and options, their products may change. Lucent Technologies will have no knowledge or control over any such changes made by other manufacturers in their H.320 implementation. As such, Lucent Technologies cannot guarantee the interoperability of its ITU-T implementation with untested versions of ITU-T compliant endpoints.

See the “Collecting Endpoint Data” section that follows for a list of information you should obtain about each participating endpoint. If your reservations agent is using the paper-based scheduling system, you should provide some information on a Site Profile for the reservations agent to use.

Table 0-3. Sample Site Profile

Site Profile					
Location Name	**Time Zone**	**Bandwidths Available**	**Network Type**	**Room Location**	**Contact Name/Phone**
Site A	EST	all	private	G123	J. Gibson/215 555-1200
Site B	EST	56k 64k	private	1B-24	T. Lee/301 555-3231
Site C	MST	336k	private	457	C. Charles/303 555-9872
Site D	EST	768k	private	3G-104	D. Gilberg/908 555-7777
Site E	CST	1472k	public	28A	R. Prasad/312 555-1818
Site F	PST	224/256k	private	1150	A. Snow/619 555-2222

Collecting Endpoint Data

To prequalify an endpoint to participate in a multipoint video conference, the following information is needed:

- Type of video endpoint (such as Lucent Technologies Group Video System or PictureTel™ Group Video System)
- Model name or number (such as Model 400, Model 70)
- Options installed
- Time zone where the video endpoint is located
- Encoding algorithms supported (for example, H.320 or a proprietary algorithm, such as SG4, H.CTX, etc.)
- Specific software version (such as 4.1, 1.1)
- Bandwidths supported
- Data capabilities of the system
- Network access (public through direct connection or private through a PBX)
- Network services supported (such as SDDN, ACCUNET SDS)

  NOTE:
  The MCU and video endpoints joining a conference must be subscribers to the same network services. The MCU can be a subscriber to multiple services.

- Name of the location or person associated with the video endpoint
- Name and number of the person responsible for setup and maintenance
- Name by which the video endpoint will be identified (such as the Sales Department in a particular city)

Currently, most video endpoints have the ability to select between originating a 2B-channel 56k call or a 2B-channel 64k call. In general, video endpoints with ISDN-PRI access to the network are not constrained on the types of calls they can make. However, endpoints with robbed-bit signaling (RBS) network access trunks are restricted to originating and receiving only 56k or 56k-increment BONDed calls. Endpoints originating 64k calls are blocked if the MCU has access only to 56k facilities or when all 64k facilities are in use when the call arrives. Currently, 384k calls to the MCU require ISDN-PRI H0/multirate capability by both the MCU and each participating video endpoint. Also, any multirate or 64k-increment BONDed calls require ISDN-PRI.

Recording Meet-Me Numbers and MCU-Extensions

The MCU-Extensions Form should be completed after the trunks and network numbers are ordered for the MCU.

The form shows all MCU-extensions assigned to the MCU and their corresponding Meet-Me numbers (the phone number the conferee dials to join a conference).

The MCU-Extensions Form also includes the bandwidth per channel available with each MCU-extension and the Meet-Me numbers categorized by network type, either private or public. It should be updated whenever a Meet-Me number is added or removed (see the *Administering Meet-Me Numbers and MCU-Extensions* sections for more information).

An MCU-extension is added to the system by entering at the prompt **add mcu-extension** and the appropriate MCU-extension (for example, **add mcu-extension** *75000*). The following version of the MCU-Extension form appears:

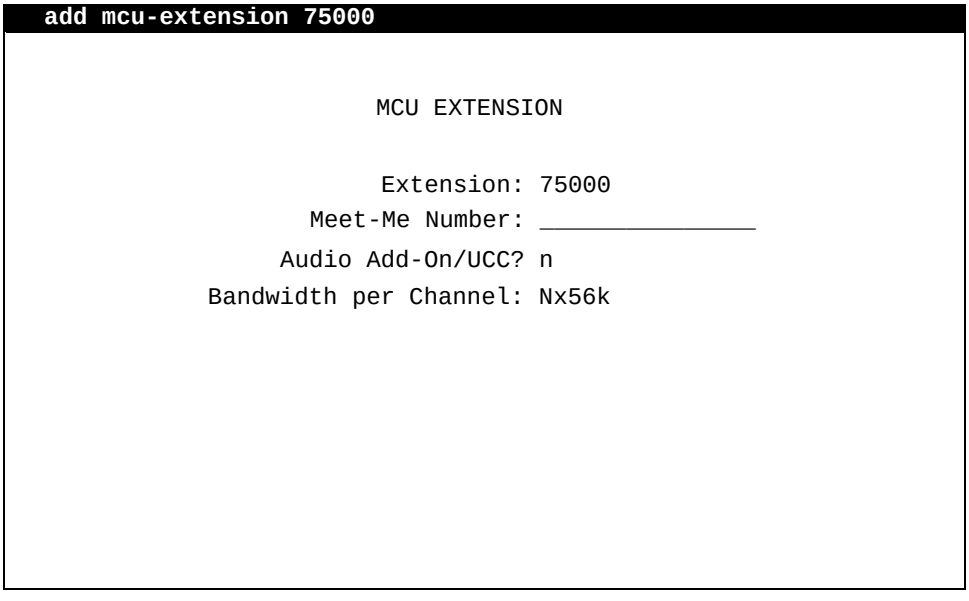

Figure 0-1. MCU-Extension Form for Add/Change/Remove/Display

NOTE:
Value “Nx56k” indicates that the endpoint is restricted to 56k-increment calls.

The following version of the MCU-Extension form appears when the **list mcu-extension** command is entered:

```
list mcu-extension

                        MCU EXTENSIONS

Ext.       Meet-Me Number       Bandwidth    Audio Add-On/UCC?

   72009       (700)737-5009        all              n
   75000       (700)737-5111        Nx56k            n
   75001       (700)737-5001        all              n
   75002       (908)560-5002                         y
   75004                            all              n
   75006                            all              n
   75007       (700)561-5007        Nx56k            n
   75008       (700)640-5008        all              n
   75010       (700)737-5010        all              n
   75106                            all              n
   75107       (700)561-5107        Nx56k            n
   75108       (700)640-5108        all              n
   76001       (700)737-5003        all              n
   78005       (700)737-5005        all              n
   78105       (700)737-5105        all              n
```

Figure 0-2. MCU-Extension Form for the List Command

Login Administration

When your MCU is delivered, one customer super-user login and password combination is already defined. You must administer additional logins and passwords, if needed. If you are the super-user, you have full customer permissions and can customize any login you create. The maximum number of customer logins is 11.

As super-user, you can establish permissions for the other logins in your system. You can block access to any object that may compromise switch security.

NOTE:
You cannot use a shortened form of the login ID for any of the following actions.

The following login administration commands are available.

Table 0-4. Login Administration Commands

Action	Object	Qualifier
add	login	xxxxxx (login-ID)
change	login	xxxxxx (login-ID)
change	permissions	xxxxxx (login-ID) [print or schedule]
display	login	xxxxxx (login-ID) [print or schedule]
list	logins	
remove	login	xxxxxx (login-ID)
status	login	

Adding a Login

To add a customer login, you must be a super-user with authority to administer permissions. To establish a login, first use the Login Administration form. An example of the form follows.

```
add login                                                    Page  1

                              LOGIN ADMINISTRATION

        Password of Login Making Change:


        LOGIN BEING ADMINISTERED
                          Login's Name:xxxxxxx
                            Login Type:
                         Service Level:
    Disable Following a Security Violation?
                     Access to INADS Port?


       LOGIN'S PASSWORD INFORMATION
                      Login's Password:
                Reenter Login's Password:
    Password Aging Cycle Length (Days):


   LOGOFF NOTIFICATION
     Facility Test Call Notification? y       Acknowledgment Required? y
          Remote Access Notification? y       Acknowledgment Required? y
```

Figure 0-3. Login Administration Form

Once the login is established, set permissions using the Command Permission Categories form (discussed later in this section).

If you are adding a login, enter the new login name as part of the **add** command. The name must contain between three and six characters, and it can contain the characters 0-9, a-z, A-Z. If you add or remove a login, the Security Measurement reports are not updated until the next hourly poll or until you enter the **clear measurements security-violations** command.

The form contains the following fields.

- *Password of Login Making Change*

  You must enter your password in order to change any field on this form.

- *Login's Name*

  Displays the login name that you assigned when you entered the **add login** command.

- *Login Type*

  Indicates whether this login belongs to a customer or to Lucent Technologies services personnel. The default and only valid value for customer logins is **customer**.

- *Service Level*

  Indicates whether this user has super-user or non-super-user permissions. Default is **non-super-user**. You cannot change your own service level.

  The super-user login has access to the **add, change, display, list,** and **remove** commands for all customer logins and passwords. The super-user can administer any mix of super-user and non-super-user logins up to a total of eleven customer logins.

  "Non-super-user" permissions are limited by restrictions specified by the super-user when administering the non-super-user login. A non-super-user can change his or her password with permission set by the super-user; however, once a password has been changed, the non-super-user must wait 24 hours before changing the password again. A non-super-user cannot change other user passwords, login characteristics, or permissions.

- *Disable Following a Security Violation*

  Enter **y** (default) to disable this login following a login security violation. This field appears only when the SVN Login Violation Notification feature is enabled.

- *Access to INADS Port*

  Enter **y** to allow access to the remote administration port. This field appears only if Lucent Technologies has enabled customer access to the INADS Remote Administration Port. Default is **n**.

- *Login's Password*

  You must assign the initial password for this login. The owner of the login should change the password immediately. Passwords must be four to 11 characters in length and contain at least one alphabetic and one numeric character. Valid characters include numbers, and **!&*?;'^(),.:-**. The password does not display on the screen as you type.

- *Reenter Login's Password*

  The login's password, as described in the previous bullet item, for verification. The password does not display on the screen as you type.

- *Password Aging Cycle Length*

  For security purposes, you should assign password aging to all logins. Enter the number of days that the password will remain valid. The system tracks a password from the day the login was created or the day the user last changed the password. If you enter a blank in this field, password aging does not apply to the login. Default is blank.

- *Facility Test Call Notification*

  If this field is **y**, the user receives notification at logoff that the facility test call feature access code is administered. Default is **y**. For security purposes, this field should be **y** for all logins.

- *(Facility Test Call Notification) Acknowledgment Required*

  If this field is **y**, the user is required to acknowledge that he or she wishes to log off while Facility Test Call is still administered. Default is **y**.

- *Remote Access Notification*

  If this field is set to **y**, the user receives notification at logoff that remote access is still administered. Default is **y**. For security purposes, this field should be **y** for all logins.

- *(Remote Access Notification) Acknowledgment Required*

  If this field is **y**, the user is required to acknowledge that he or she wishes to log off while remote access is still administered. Default is y.

Changing a Login's Attributes

To change a customer login's attributes you must be a super-user and have administration permissions.

Enter the **change login (login-ID)** command to access the Login Administration form. Make any necessary changes and press ENTER to record the changes in the system.

Displaying a Login

To display a specific login, enter the **display login (login-ID)** command and press the ENTER key. The system displays the Login Administration form with all information about the requested login except the password.

Listing Logins

To list all of the system logins and the status of each login, enter the **list logins** command. The system displays the following information for all current logins:

- Name
- Service level
- Status (active, inactive, disabled, svn-disabled, void)
- Password aging cycle length

The system displays only those logins with the same, or lower, service level as the user making the request.

Removing a Login

To remove a login from the system, you must use a super-user login.

1. Enter the **remove login (login-ID)** command. The system displays the Login Administration form with all of the information about the requested login except the password.
2. Press ENTER to remove the login, or press CANCEL to cancel the procedure without removing the login.

Forced Password Aging

Password aging is an option that you can activate while administering logins. The password for each login can be aged starting with the date the password was created, or changed, and continuing for a specified number of days (one to 99).

The user is notified at the login prompt seven days before the password expiration date that the password is about to expire. When the password expires, the user must enter a new password into the system before logging in. Once a non-super-user has changed his or her password, the user must wait 24 hours to change the password again.

Changing Administration Permissions

The Permission Administration form is used to change the Command Permission Categories that an authorized user (reservations agent, system administrator, and on-site technician) can access. Each is identified by a login name. Prior to installation, these passwords are voided; the Video Technical Center (VTC)/NetCare Center must reinstate them with new customer-selected passwords.

Only the VTC is authorized to change the *Administer Permissions* field. System administrators and reservation agents cannot change the default permissions for themselves or others unless the VTC changes the *Administer Permissions* field to **y** for the customer.

Table 0-5. Default Permissions for Each Login

Login	Default Permissions
craft	All permission categories except Administer Permissions
cust	Display administration and maintenance data, administer conferences, features, Meet-Me numbers, maintenance alarm terminals, and trunks
rcust	Display administration and maintenance data, administer conferences, Meet-Me numbers, maintenance alarm terminals, and trunks
agent	Display administration and maintenance data, administer conferences and Meet-Me numbers
browse	Display administration and maintenance data

Use the following procedure to change permissions:

1. At the *command:* prompt, type **change permission *login*** then press RETURN.
2. Verify that the screen displays the Permission Administration screen.
3. From the *Display Admin. and Maint. Data?* field, use TAB and RETURN to advance to the fields you want to change. Type **y** or **n** as appropriate, followed by ENTER.
4. When the form is completed, verify the MT displays: *Command successfully completed* followed by *command:*
5. To verify the form, type **display permission *login*** then press RETURN.

Command Permission Categories Form

An example of page 1 of the Command Permission Categories form appears as follows.

```
change permissions cust                                    Page 1 of 1

                          COMMAND PERMISSION CATEGORIES

                                Login Name: cust


COMMON COMMANDS
              Display Admin. and Maint. Data? y
                         System Measurements? n
              System Mgmt Data Transfer Only? n

ADMINISTRATION COMMANDS

         Administer Stations? y                      Administer Features? y
           Administer Trunks? y                   Administer Permissions? n
  Administer Meet-me Numbers? y              Administer Conference Records? y
     Additional Restrictions? n

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

Figure 0-4. Command Permission Categories Form (Page 1)

The following list identifies the fields on page 1 of the Command Permission Categories form. Enter **y** for each field you want the login to access; enter **n** for each field you do not want the login to access.

- *Login name*

  Displays the login for which the permissions are being changed.

- *System Measurements*

  Displays system measurement data.

- *Display Admin. and Maint. Data*

  Shows administration screens and maintenance data.

- *Administer Stations*

  Specifies whether or not this login is permitted to change maintenance phone assignments and extensions for the MCU.

- *Administer Trunks*

  Specifies whether or not this login is permitted to change trunk groups and assignments for the MCU.

- *Administer Meet-Me Numbers*

  Specifies whether or not this login is permitted to change Meet-Me number assignments.

- *Additional Restrictions*

  If this field is set to **y**, the second and third pages of this form appear. You can use these pages to add up to 40 objects that this user cannot manipulate. If an object appears on the Additional Restrictions page, users cannot display, add, change, or do anything with that object.

- *Administer Features*

  Specifies whether or not this login is permitted to add or remove features from the MCU.

- *Administer Permissions*

  Specifies whether or not this login is permitted to change the command permissions for logins of the MCU.

- *Administer Conference Records*

  Specifies whether or not this login is permitted to administer conference records, including scheduling conferences and updating conference reservations.

- *Maintain Stations*

  Specifies whether or not this login can perform maintenance commands on the MCU maintenance phones.

- *Maintain Trunks*

  Specifies whether or not this login can perform maintenance tests on the MCU trunks.

- *Maintain System*

  Specifies whether or not this login can perform system maintenance tests on MCU processor circuit packs.

- *Maintain Switch Circuit Packs*

  Specifies whether or not this login can perform maintenance tests on the MCU circuit packs.

- *Maintain Process Circuit Packs*

  Specifies whether or not this login can perform maintenance tests on MCU processor circuit packs.

- *Maintain Enhanced DS1*

  You can enter **y** into this field only if *DS1 MSP* is set to **y** on the Customer-Options form. Allows users to perform enhanced DS1 maintenance.

Pages 2 and 3 of the form are identical.

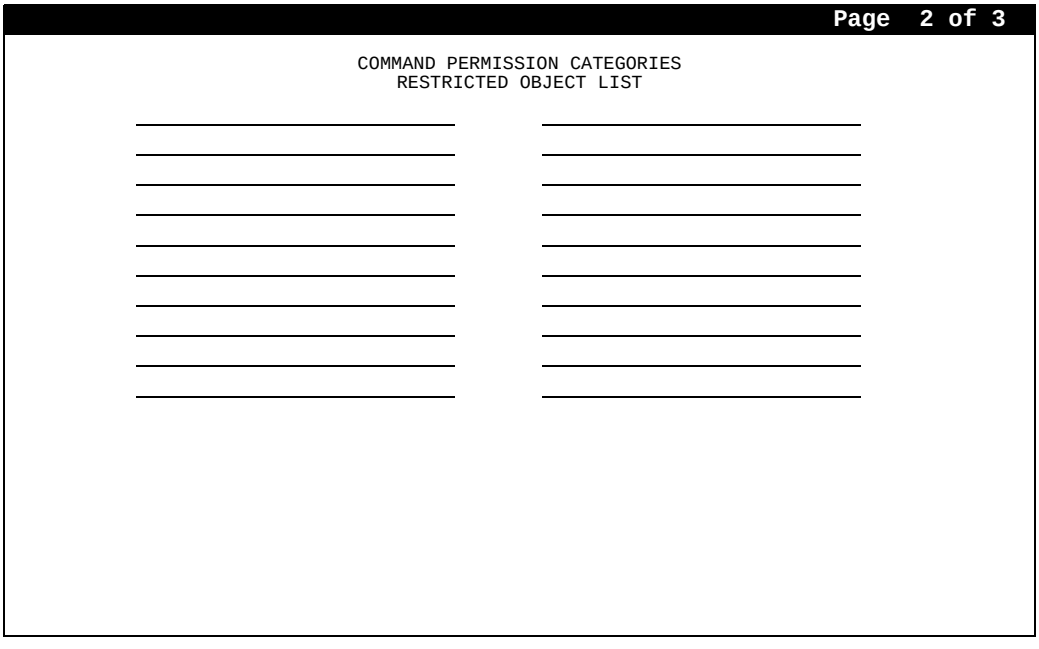

Figure 0-5. Command Permissions Categories Form (Page 2)

These pages allow you to specify certain objects that this user cannot access. If you want to limit a user's permissions beyond those on page 1, enter the objects into this list.

Selecting Customer Options

The Customer-Options form is used by approved Lucent Technologies personnel to administer MCU model changes. The form specifies the port capacity, optional features, and model name of the MCU.

Use the following procedure to change customer options:

1. At the *command:* prompt, type **change system-parameters customer-options** then press RETURN.
2. Verify that the screen displays the Customer-Options form.
3. From the *Model Name* field, use TAB and RETURN to advance to the fields you want to change.
4. When the form is completed, verify that the MT displays: *Command successfully completed* followed by *command:*
5. To verify the form, type **display system-parameters customer-options** and then press RETURN.

Customer-Options Form

The Customer-Options form appears as follows:

```
change system-parameters customer-options                    Page  1 of  1
                                CUSTOMER OPTIONS


            Model Name: EX
Presentation/Broadcast? y        Dynamic Resizing? y              ISDN-PRI over PACCON?: y
                  VAS? y      Notification Tones? y                 Conferencing Suite: 0
   Chair Control Suite? y               Cascading? y Continuous Presence/Executive Conf: 3
   BONDed Use Same DS1? n       Scheduler Adjunct? y                Executive Conference? 0
             Dial-out? y        OneNumber Access: all                         Speed Match: 3
    ClearPresence @716? n

 MAXIMUM PORT CAPACITY                                                BONDing 112/128K: 24
                    2B: 64                  512K: 24                  BONDing 168/192K: 24
                128K: 64                  768K: 12                  BONDing 224/256K: 24
                192K: 64            1008/1152K: 0                   BONDing 280/320K: 24
                256K: 60            1288/1472K: 12                  BONDing 336/384K: 24
                320K: 56            1344/1536K: 8                   BONDing 448/512K: 16
                384K: 52            1680/1920K: 8                   BONDING 672/768K: 8

         Audio Add-on: 32                  UCC: 12                BRI Direct Connect: 0
                T.120: 24                  SG4: 16                DCP Direct Connect: 0
                                        H.CTX/+: 16                 DI Direct Connect: 0



            (NOTE: You must logoff & login to effect the permission changes.)
```

Figure 0-6. Customer-Options Form

The following list identifies the fields on the Customer-Options form and their possible values.

NOTE:
This form can be administered only by personnel approved by Lucent Technologies

Additional hardware may be required to support the features and capabilities that you administer on this form. For details, refer to the “Hardware Requirements” section in Appendix B for the appropriate features and capabilities.

- *Model Name*

  Depending on the customer’s order (shown by the model PEC associated with an order), enter **CX**, **DX**, or **EX**.

- *Presentation/Broadcast*

  Shows a **y** to indicate that both Broadcast mode with Auto Scan and Presentation mode are supported. Broadcast mode allows one endpoint (known as the broadcaster) to be viewed for the entire conference by all participants. The broadcaster views a continual rotation of participating endpoints. No return audio is heard. Presentation mode is similar to broadcast mode except that the audio portion is a summation of all participating endpoints, not just the broadcaster.

- *VAS*

  Shows a **y** to indicate that the MCU supports Voice-Activated Switching, where the speaker is viewed by all parties except the speaking party; this party views the previous speaker.

- *Chair Control Suite*

  Shows a **y** to indicate that the MCU supports chair control or FECC where the conferee designated as the “chair” or the far-end camera controlling endpoint controls who is viewed by all parties.

- *BONDed Use Same DS1*

  Shows a **y** to indicate that higher BONDing port capacities can be used, assuming that all BONDed calls from the same endpoint arrive over the same DS1 facility.

- *Dial-out*

  Shows a **y** to indicate that the MCU can dial out to add video endpoints to a conference. The default setting is **n**.

- *Dynamic Resizing*

  Shows a **y** to indicate that the list of conferees scheduled to join a conference can be changed even during an active conference.

- *Notification Package*

  Set this field to **y** to enable the MCU to support the following:

  — Entry and exit text overlay messages as well as entry and exit tones whenever a conferee joins or disconnects from a multipoint conference

  — Warning text overlay messages and warning tones notifying conference participants whenever 10 minutes remain in an active conference

- *Cascading*

  Shows **y** to indicate that the MCU can be connected to another Lucent Technologies MCU to combine resources of up to 46 ports.

- *Scheduler Adjunct*

  Enter **y** if the customer ordered the optional Lucent Technologies Conference Reservation and Control System (CRCS) with or without Standalone OpCenter. This activates the Operations Support System Interface (OSSI). The default setting is **n**.

- *OneNumber Access*

  Enter the appropriate value to either disable this feature or enable it with specified parameters.

  The following list identifies the possible values and their meaning:

  — **none** — disables the feature.

  — **aud** — enables the feature only for Audio Add-On endpoints. H.320 and BONDed endpoints are not allowed.

  — **mrate/aud** — enables the feature for H.320 endpoints in multirate conferences and Audio Add-On endpoints in all conference types. H.320 endpoints in 2B conferences and BONDed endpoints are not allowed.

  — **all** — enables the feature for H.320 endpoints in 2B conferences, as well as for Audio Add-On and multirate endpoints in all conference types. BONDed endpoints are not allowed.

  NOTE:
  If a value to enable the feature is entered, any duplicate extensions administered for the feature must be of the type enabled via the value. Otherwise, such extensions are blocked via administration.

- *ISDN-PRI over PACCON*

  Display-only field shows a **y** to indicate the termination capability of ISDN-PRI over the Packet Control (PACCON) circuit pack.

NOTE:
This capability is supported only by the TN2207 circuit pack. It is not supported by the TN767 circuit pack.

- *Conferencing Suite*

  Enter the number of full-screen (with or without H.263/H.261 transcoding) CPP, CPP with SM, or Executive Conferencing (EC) conferences for the MCU. Values include **0** through **8**.

  NOTE:
  The MCU supports a maximum of 10 Video Processor circuit packs. (However, only eight circuit packs can be used.) These circuit packs can accommodate a mixture of full screen-only Speed Match conferences, CPP Speed Match conferences, CPP-only conferences, H.263/H.261 transcoding (H.320 Translator) conferences, and Executive Conferencing conferences.

  The number of simultaneous conferences using some form of video processing is controlled by the number of each of the three right-to-use permissions as displayed on the Customer-Options form. In particular, the sum of the CPP-only, full screen Speed Match (SM), combination SM/CPP, H.263/H.261 transcoding, and Executive Conferencing right-to-use (RTU) permissions may not exceed eight since one dedicated Video Processor circuit pack is required to support each of these conference types. CPP/EC permissions are valid only for non-Speed Match, single transfer rate CPP (or CPP-only) conferences or EC conferences. SM permissions are valid only for full screen SM conferences (with or without H.263/H.261 transcoding). However, Conferencing Suite permissions can be used to provide a CPP-only conference, a full screen-only SM conference, an SM conference that also provides CPP video to and from endpoints at various transfer rates, an H.263/H.261 transcoding conference, or an Executive Conferencing conference. Therefore, as long as all of the available Conferencing Suite right-to-use permissions have not been used, any such available permissions may be applied to administer a CPP-only, full screen-only SM, SM/CPP, H.263/H.261 transcoding, or Executive Conferencing conference.

  A maximum of *eight* CPP, full screen Speed Match, CPP Speed Match, H.263/H.261 transcoding, and Executive Conferencing conferences *combined* are allowed. If both *Continuous Presence/EC* and *Conferencing Suite* contain **0**, CPP cannot be administered; if both *Speed Match (SM)* and *Conferencing Suite* contain **0**, full screen Speed Match cannot be administered; if *Executive Conference, Continuous Presence/Executive Conf,* and *Conferencing Suite* contain **0**, Executive Conferencing cannot be administered.

In addition, starting with Release 6.0, the right-to-use (RTU) for Speed Match also applies to full-screen H.263/H.261 transcoding (in addition to full-screen Speed Match). Similarly, the RTU for Conferencing Suite also applies to full-screen H.263/H.261 transcoding (in addition to full-screen Speed Match, CPP, CPP with SM, and Executive Conferencing).

The following table summarizes the RTUs for the aforementioned fields on the System-Parameters Customer-Options form

Table 0-6. RTUs for Several Options on the Customer Options Form

Option (Field)	Meaning
Continuous Presence/ Executive Conf	--- Continuous Presence *Plus,* or --- Executive Conferencing
Speed Match (SM)	--- full-screen Speed Match, or --- full-screen H.263/H.261 Transcoding at Same Speed
Conferencing Suite	--- CPP, or --- full-screen Speed Match, or --- full-screen H.263/H.261 Transcoding at Same Speed, or --- CPP with Speed Match, or --- Executive Conferencing
Executive Conferencing	--- Executive Conferencing only

- *Continuous Presence/Executive Conf*

  Enter the number of CPP-only (non-speed-matched) or Executive Conferencing conferences for the MCU. Values include **0** through **8**.

- *Executive Conference*

  Enter the number of Executive Conferencing conferences for the MCU. Values include **0** through **8**.

- *Speed Match (SM)*

  Enter the number of full screen Speed Match conferences (with or without H.263/H.261 transcoding) for the MCU. Values include **0** through **8**.

The following fields are included within the *Maximum Port Capacity* section of the form.

- *2B*

  Enter the total number of ports in the configuration. Values include **0** through **96** in four port increments, along with value **6**. The default setting is **4** ports.

- *128k*

  Enter the number of ports available for 128k conferencing**.** Values include **0** through **96** in four port increments**.**

- *192k*

  Enter the number of ports available for 192k conferencing**.** Values include **0** through **96** in four port increments.

- *256k*

  Enter the number of ports available for 256k conferencing**.** Values include **0** through **96** in four port increments.

- *320k*

  Enter the number of ports available for 320k conferencing**.** Values include **0** through **96** in four port increments.

- *384k*

  Enter the number of ports available for 384k (H0) conferencing. These ports may be used for 2B conferences when they are not reserved for 384k conferences. The number of H0 ports is always less than or equal to the number of 2B ports installed. Values include **0** through **96** in four port increments. The default setting is **0**.

- *512k*

  Enter the number of ports available for 512k conferencing**.** Values include **0** through **72** in four port increments.

- *768k*

  Enter the number of ports available for 768k conferencing. Values include **0** through **48** in four port increments

- *1008/1152k*

  Enter the number of ports available for 1008k and/or 1152k conferencing. Values include **0** through **28** in four port increments. The default setting is **0**.

- *1288/1472k*

  Enter the number of ports available for 1288k and/or 1472k conferencing. Values include **0** through **24** in four port increments. The default setting is **0**.

- *1344/1536k*

  Enter the number of ports available for 1344k and/or 1536k conferencing. Values include **0** through **24** in four port increments. The default setting is **0**.

- *1680/1920k*

  Enter the number of ports available for 1680k and/or 1920k conferencing. Values include **0** through **24** in four port increments. The default setting is **0**.

- *BONDing 112/128k*

  Enter the number of BONDing ports available for 112/128k BONDing. For flexible Dial-Out conferencing, enter a value from **0** through **96** in four port increments; for dial-in/non-CRCS conferencing, enter a value from **0** through **64** in four port increments.

- *BONDing 168/192k*

  Enter the number of BONDing ports available for 168/192k BONDing. For flexible Dial-Out conferencing, enter a value from **0** through **96** in four port increments; for dial-in/non-CRCS conferencing, enter a value from **0** through **64** in four port increments.

- *BONDing 224/256k*

  Enter the number of BONDing ports available for 224/256k BONDing. For flexible Dial-Out conferencing, enter a value from **0** through **96** in four port increments; for dial-in/non-CRCS conferencing, enter a value from **0** through **60** in four port increments.

- *BONDing 280/320k*

  Enter the number of BONDing ports available for 280/320k BONDing. For flexible Dial-Out conferencing, enter a value from **0** through **96** in four port increments; for dial-in/non-CRCS conferencing, enter a value from **0** through **56** in four port increments.

- *BONDing 336/384k*

  Enter the number of BONDing ports available for 336/384k BONDing. For flexible Dial-Out conferencing, enter a value from **0** through **96** in four port increments; for dial-in/non-CRCS conferencing, enter a value from **0** through **52** in four port increments.

- *BONDing 448/512k*

  Enter the number of BONDing ports available for 448/512k BONDing. For flexible Dial-Out conferencing, enter a value from **0** through **72** in four port increments; for dial-in/non-CRCS conferencing, enter a value from **0** through **44** in four port increments.

- *BONDing 672/768k*

  Enter the number of BONDing ports available for 672/768k BONDing. For flexible Dial-Out conferencing, enter a value from **0** through **48** in four port increments; for dial-in/non-CRCS conferencing, enter a value from **0** through **36** in four port increments.

- *Audio Add-On*

  Enter the maximum number (**0**, **2, 4, 6, 8, 10, 12, 14, 16, 18, 20, 22, 24, 26, 28, 30,** or **32**) of audio-only ports per system. The default setting is **0**.

- *T.120*

  Enter the maximum number of T.120 ports (**0**, **4**, **8**, **12**, **16**, **20**, or **24**) that can be scheduled for a conference at the same time. The default is **0**. If the value is set to **0**, the T.120 feature is not optioned for the MCU. If the value is set to any valid number other than **0**, the T.120 feature is optioned for the MCU.

  NOTE:
  The maximum 2B port capacity must be greater than or equal to the maximum T.120 port capacity.

- *UCC*

  Enter the number of UCC ports (**0**, **1**, **2**, **4**, **8**, **12**, **16**, **20**, or **24**) available for UCC conferencing.

- *SG4*

  Enter the number of SG4 ports available for SG4 conferencing. Values include **0** through **96** in 16 port increments.

- *H.CTX/+*

  Enter the number of H.CTX or H.CTX+ ports available for H.CTX or H.CTX+ conferencing. Values include **0** through **96** in 16 port increments.

- *BRI Direct Connect*

  Enter the number of BRI Direct Connect ports (**0**, **12, 24,** or **36**) available for conferencing. Default is **0**. If this field contains value **0** and BRI ports are installed, system resources are reserved to allow nonblocking point-to-point calls for this port type .

- *DCP Direct Connect*

  Enter the number of DCP Direct Connect ports (**0** or **4**) available for conferencing.

- *DI Direct Connect*

  This field reserves either time slots for each administered DI port using the 384k transfer rate or 12 time slots per DI port for estimating. Values for this field equate to the number of DI circuit packs (ports) possible for an MCU, and they include the range of **0** through **32** in increments of one. The number of time slots to reserve per circuit pack (port) depends on the transfer rates in effect at the time.

  NOTE:
  The DI port capacity that is available at any one time is the same as the multirate capacity value (for example, a DI port rate of 1920k allows a maximum of 16 ports at a time).
  One TN2236B, version 5 or greater circuit pack is required for each DI access-endpoint that is administered in the MCU unless the DI access-endpoint requires 2B calling (2x56k or 2x64k); in such a case, one TN2236B, version 5 or greater circuit pack is required for each call.

- *ClearPresence @716*

  Indicates whether the right-to use (RTU) fee required for ClearPresence @716 has been paid by the customer. Values include **y** and **n** (default). Also, indicates the set of audio modes available for conferences and the rationale for each mode. For value **y**, the audio modes supported include ClearPresence @716, G.722, G.728, G.711, auto-down_only, and auto-w/upgrade. Value **y** also indicates that G.728/G.722 interworking is available in Speed Match conferences. For value **n**, ClearPresence @716 and G.728/G.722 interworking are not supported.

  NOTE:
  If a TN788B circuit pack is inserted after value **y** is entered into the *ClearPresence @716* field, an alarm occurs.

Setting the Class of Restriction

The Class of Restriction (COR) feature is used to define the calling capabilities of MCU trunk groups, hunt groups, data modules, data extensions, and maintenance alarm terminals. You can assign one COR to all of these groups, or you can assign as many as necessary (up to 64) to effect the desired restriction.

For example, you may assign to the maintenance alarm terminals a COR with no outside calling privileges. Likewise, you could restrict MCU trunk groups to incoming calls only.

The following commands can be used to access the Class of Restriction (COR) form.

Table 0-7. Class of Restriction Administration Commands

Action	Object	Qualifier
change	cor	0 to 63
display	cor	0 to 63, print or schedule

Use the following procedure to change the COR:

1. At the *command:* prompt, type **change cor #** and then press RETURN.
2. Verify that the screen displays the Class of Restriction screen.
3. Use TAB or RETURN to advance to the fields you want to change.
4. When the form is completed, verify the MT displays: *Command successfully completed* followed by *command:*
5. To verify the form, type **display cor #** and then press RETURN.

Class of Restriction Form

The Class of Restriction form appears as follows:

```
change cor 1                                              Page  1 of  2

                            CLASS OF RESTRICTION

              COR Number: 1
         COR Description: _

                     FRL: 7                                    APLT? y
                                          Calling Party Restriction: none
                                           Called Party Restriction: none
Partitioned Group Number: 1              Forced Entry of Account Codes? n
        Priority Queuing? n
                                            Facility Access Trunk Test? n
    Restricted Call List? n


    Category For MFC ANI: 7
        Send ANI for MFE? n
           MF ANI Prefix:
```

Figure 0-7. Class of Restriction Form (Page 1)

```
change cor 1                                              Page  2 of  2

                                CLASS OF RESTRICTION

CALLING PERMISSION (Enter "y" to grant permission to call specified COR)

  0? y      12? y     24? y     36? y     48? y     60? y     72? y     84? y
  1? y      13? y     25? y     37? y     49? y     61? y     73? y     85? y
  2? y      14? y     26? y     38? y     50? y     62? y     74? y     85? y
  3? y      15? y     27? y     39? y     51? y     63? y     75? y     87? y
  4? y      16? y     28? y     40? y     52? y     64? y     76? y     88? y
  5? y      17? y     29? y     41? y     53? y     65? y     77? y     89? y
  6? y      18? y     30? y     42? y     54? y     66? y     78? y     90? y
  7? y      19? y     31? y     43? y     55? y     67? y     79? y     91? y
  8? y      20? y     32? y     44? y     56? y     68? y     80? y     92? y
  9? y      21? y     33? y     45? y     57? y     69? y     81? y     93? y
 10? y      22? y     34? y     46? y     58? y     70? y     82? y     94? y
 11? y      23? y     35? y     47? y     59? y     71? y     83? y     95? y
```

Figure 0-8. Class of Restriction Form (Page 2)

The following list provides the fields on the COR form and their related values.

NOTE:
Keep the default setting for all the fields in this form.

Class of Restriction Form Page 1 of 2

- *COR Number*

  Display-only field when the form is accessed via a **change cor** or **display cor** command. Shows the COR number entered.

  NOTE:
  There is no need at this time to make changes to the COR form because administered extensions do not currently use the Class of Restriction features. Whenever you are administering the MCU, leave this field set at the default value of **1** for any administration forms (such as those for trunk groups, data extensions, etc.) that have access to the COR feature.

- *COR Description*

  Enter a description of the COR containing a maximum of 35 characters.

- *FRL*

  Provides a method of restricting users from originating calls to specified areas. There are eight levels of calling restrictions, “0” through “7.” “0” is the most restricting for a station, and “7” is the least restricting. A station user who has an FRL of “3” has all the calling privileges of FRLs “0” through “2” plus what is allowed on FRL “3.”

  Except for special cases, changes are not required on the MCU. Leave at default value of **7** on all COR forms.

- *Partitioned Group Number*

  This field is not used. Leave it at the default of **1**.

- *Priority Queuing*

  This field is not currently used. Leave it at the default setting of **n**.

- *Restricted Call List*

  This field is not currently used. Leave it at the default setting of **n**.

- *APLT*

  N/A for the MCU. Leave set to **y**.

- *Calling Party Restriction*

  N/A for the MCU. Leave it at the default of **none**.

- *Called Party Restriction*

  N/A for the MCU. Leave it at the default of **none**.

- *Forced Entry of Account Codes*

  N/A on the MCU. Leave it at the default setting of **n**.

- *Category for MFC ANI*

  N/A for the domestic MCU. Leave at default value **7**.

- *Send ANI for MFE*

  N/A for the domestic MCU; therefore, enter value **n**. If desired in Spain, enter value **y**.

- *MF ANI Prefix*

  Defines the prefix to apply to an extension number when ANI is sent to the CO. This prefix overrides any ANI prefix that is administered on the Multifrequency-Signaling-Related System Parameters form, except whenever ANI is tandemed through the PBX on tandem calls or whenever ANI for a PBX is sent. Values include one to seven digits. Default is blank.

- *Facility Access Trunk Test*

  This field is not currently used. Keep the default value **y**.

Class of Restriction Form Page 2 of 2

- *Calling Permissions*

  Keep the default value **y**.

Setting the Class of Service

The Class of Service (COS) feature is used to administer access permissions for call processing features that require a dial code and/or feature access button permission. COS does not apply to trunk groups. All COSs (**0** through **15**) have the same settings; all fields are defaulted to **n** except *Data Privacy*, which is set to **y**. Therefore, you can select **0** through **15** for any of the features that require a COS.

NOTE:
Do not change the defaults.

The following commands can be used to access the Class of Service (COS) form.

Table 0-8. Class of Service Administration Commands

Action	Object	Qualifier
change	cos	
display	cos	print or schedule

Use the following procedure to change the COS:

1. At the *command:* prompt, type **change cos** and then press RETURN.
2. Verify that the screen displays the Class of Service screen.
3. From the *Auto Callback* row, use TAB and RETURN to advance to the fields you want to change.
4. When the form is completed, verify that the MT displays: *Command successfully completed* followed by *command:*
5. To verify the form, type **display cos** and then press RETURN.

Class of Service Form

The Class of Service Form appears as follows.

```
change cos                                                  Page  1 of  1


                              CLASS OF SERVICE

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

Auto Callback                 n  y  y  n  y  n  y  n  y  n   y   n   y   n   y   n
Call Fwd-All Calls            n  y  n  y  y  n  n  y  y  n   n   y   y   n   n   y
Data Privacy                  n  y  n  n  n  y  y  y  y  n   n   n   n   y   y   y
Priority Calling              n  y  n  n  n  n  n  n  n  y   y   y   y   y   y   y
Console Permissions           n  n  n  n  n  n  n  n  n  n   n   n   n   n   n   n
Off-hook Alert                n  n  n  n  n  n  n  n  n  n   n   n   n   n   n   n
Client Room                   n  n  n  n  n  n  n  n  n  n   n   n   n   n   n   n









QSIG Call Offer Originations  n  n  n  n  n  n  n  n  n  n   n   n   n   n   n   n
```

Figure 0-9. Class of Service Form

The following list provides the fields on the COS form and their related values.

Class of Service Form Page 1 of 1

- *Auto Callback*

  This feature is not presently available. Leave all entries set to **n**.

- *Call Fwd-All Calls*

  This feature is not currently available. Leave all entries set to **n**.

- *Data Privacy*

  The MCU should be set for data privacy. All entries should be set to **y**.

- *Priority Calling*

  This feature is not currently available. Leave all entries set to **n**.

- *Console Permissions*

  This feature is not used. Leave all entries set to **n**.

- *Off-hook Alert*

  This feature is not currently available. Leave all entries at the default of **n**.

- *Client Room*

  This field is not currently used. Leave all entries set to **n**.

- *QSIG Call Offer Originations*

  For outgoing call offer call originations, set to **y**. Default is **n**.

Dial-Out Administration

The Dial-Out feature allows the Lucent Technologies MCU to automatically originate calls to video conferees to join a conference. The conferees' video terminals are not directly connected to the MCU; they gain access via data trunks. Administered connections are used to establish and terminate the call from the MCU to the endpoints. Therefore, the Dial-Out feature requires coordination among the video endpoints, trunks, and MCU.

Dial-out administration involves outgoing trunk groups, AAR/ARS routing patterns, and administered connections.

Use the following procedure to change the COR:

1. At the *command:* prompt, type **display administered-connection** and then press RETURN.
2. Verify that the screen displays the Administered Connection screen.
3. From the *FRL* field, use TAB and RETURN to advance to the fields you want to change.
4. When the form is completed, verify the MT displays: *Command successfully completed* followed by *command:*
5. To verify the form, type **display cor** # and then press RETURN.

Establishing Administered Connections

For dialing out, use the Administered-Connection form to establish a connection that is automatically reestablished whenever the MCU is restarted.

The following commands can be used to administer connections:

Table 0-9. Administered-Connection Administration Commands

Action	Object	Qualifier
add	administered-connection	1-600 or next
change	administered-connection	1-600 or next
display	administered-connection	1-600, next, print or schedule
duplicate	administered-connection	1-600 or next
list	administered-connection	1-600 or next
remove	administered-connection	1-600, count, print or schedule

Use the following procedure to establish administered connections:

1. At the *command:* prompt, type **add administered-connection x**, and then press RETURN.
2. Verify that the screen displays the Administered-Connection form.
3. From the *Enable* field, use TAB and RETURN to advance to the fields you want to change.
4. When the form is completed, verify the MT displays: *Command successfully completed* followed by *command:*
5. To verify the form, type **display administered-connection x**, and then press RETURN.

Administered-Connection Form

The following list identifies the fields on the Administered-Connection form and their related values:

- *Connection Number*

  This display-only field shows the Administered-Connection number.

- *Enable*

  Enter **y** to indicate that an attempt will be made to establish the administered connection whenever the MCU is restarted or the connection is due to be active. The default is **y**.

- *Originator*

  Displays the extension of the Meet-Me user who is currently using this administered connection.

- *Destination*

  Enter the address of the destination access or data endpoint (**0** through **9)**. This endpoint does not have to be local to the MCU. If a local endpoint is entered, it must be assigned to either an access or data endpoint. Up to 36 characters can be entered. The default is blank.

  NOTE:
  A FAC is restricted to AAR, ARS, and CDR account code dialing.

- *Name*

  Enter an optional name (up to 15 characters) for the connection, such as **MCU-ST.**

- *Continuous*

  Enter **y** if the administered connection is enabled whenever the MCU is restarted. Enter **n** if you want to schedule the connection to be active only during certain times. If you enter **n**, the following three fields are displayed.

- *Start Days (Sun through Sat)*

  Displays only when **n** is entered in the *Continuous* field. Enter a **y** for each day you want to establish the administered connection; otherwise, enter n. If you enter **y**, the administered connection begins on the day specified and continues for the duration set in the *Duration* field.

- *Start Time*

  Displays only when **n** is entered in the *Continuous* field. Enter the time of day when an attempt to make the connection should occur. Valid entries are **00:00** to **23:59**. The default is **00:00**.

- *Duration*

  Displays only when **n** is entered in the *Continuous* field. Enter the period of time that the scheduled administered connection should remain active. The maximum duration is one minute less than a week. Valid entries are **000** through **167** for hours; **00** through **59** for minutes. The default is **000:00**.

- *Alarm Type*

  Enter the type of alarm to be generated if the administered connection cannot be established or fails and cannot be reestablished and the number of failures equals the alarm threshold. The valid entries are **major**, **minor**, **warning**, or **none**. The default is **warning**.

- *Alarm Threshold*

  Enter the number of times an attempt to establish or reestablish a connection should be tried before an alarm is raised. Valid entries are **1** through **10.** The default is **5**.

- Retry Interval

  Enter the number of minutes between attempts to establish or reestablish the connection. Valid entries are **1** through **60**. The default is **2**.

- *Auto Restoration*

  Enter **y** to indicate that an attempt is to be made to reestablish the connection that has failed; otherwise, enter **n**. Auto restoration is only available for an administered connection that is established over ISDN-PRI and SDDN trunk groups. The default is **n.**

- *Priority*

  Enter a number in a range from **1** to **8** to determine the order in which the administered connections are to be established (the highest priority is **1**).

Administering the Call Detail Recording (CDR) Link

Call Detail Recording (CDR) allows specific information to be recorded about the multimedia conference. This information is useful for call costing, diagnostics, abuse detection, and network optimization. To enable CDR, the CDR link must be administered.

The following commands can be used to administer the CDR link:

Table 0-10. CDR Link Commands

Action	Object	Qualifier
change	system-parameters cdr	_____
display	system-parameters cdr	_____

Use the following procedure to establish administered connections:

1. At the *command:* prompt, type **change system-parameters cdr**, and then press RETURN.
2. Verify that the screen displays the CDR System Parameters form.
3. From the *Node Number* field, use TAB and RETURN to advance to the fields you want to change.
4. When the form is completed, verify that the MT displays: *Command successfully completed* followed by *command:*
5. To verify the form, type **display system-parameters cdr**, and then press RETURN.

CDR System Parameters Form

An example of the CDR System Parameters Form appears as follows:

```
change system-parameters cdr                                  Page  1 of  1


                                CDR SYSTEM PARAMETERS

 Node Number (Local PBX ID):                          CDR Date Format: month/day
      Primary Output Format: printer                Primary Output Ext: eia

           Use ISDN Layouts? n                     EIA Device Bit Rate: 9600
       Use Enhanced Formats? n
Modified Circuit ID Display? n

Suppress CDR for Ineffective Call Attempts? y          Outg Trk Call Splitting? y
    Disconnect Information in Place of FRL? n




                                  Digits to Record for Outgoing Calls: dialed

    Primary - Digits to Hide: 0
```

Figure 0-10. CDR System Parameters Form

The following list discusses the fields in the CDR System Parameters form.

- *Node Number (Local PBX ID)*

  The DCS switch node number in a network of switches. This field is display-only.

- *CDR Date Format*

  Enter month/day or day/month (default) to select the format for recording CDR date information for timestamp date records of all formats. If using a customized format, make this a six-digit field to ensure the proper output.

- *Primary Output Format*

  Controls data presentation sent to the primary output device. Entering **customized** generates the second page of this form for the customized record administration. Other values are **59-char**, **expanded**, **lsu**, **lsu-expand**, **int-direct**, **int-isdn**, **int-process**, **printer**, **teleseer**, **unformatted**, and blank (default).

- *Primary Output Ext*

  Appears and is required when the *Primary Output Format* field contains a valid value. Enter the extension number assigned to the primary output device. Default is blank.

  NOTE:
  The first digit of the CDR extension number and the length of that number must agree with the first digit table of the dial plan.

- *Use ISDN Layouts*

  Enter y to use ISDN record layouts. This field applies to both the primary and secondary formats, if applicable. The “lsu” and “printer” output formats will be affected as well as any format with ISDN layouts, such as “teleseer.” Default is **n**. This field and *Use Enhanced Formats* cannot both be set to **y**.

- *EIA Device Bit Rate*

  Enter the bit rate of the CDR device connected to the EIA port. Valid entries are **300**, **1200**, **2400**, and **9600**.

- *Use Enhanced Formats*

  Enter **y** to enable the use of the enhanced version of the specified primary output format. This field applies to both the primary and secondary output formats if applicable; however, only the “expanded,” “teleseer,” “lsu,” “printer,” and “unformatted” output formats are affected. This field and Use *ISDN Layouts* cannot both be set to **y**.

- *Modified Circuit ID Display*

  Enter **y** to modify the format of the incoming and outgoing circuit ID from the 10’s, units, 100’s format (the default) to 100’s, 10’s, units. Only the “printer,” “teleseer,” and “59-character” output formats are affected.

- *Suppress CDR for Ineffective Call Attempts*

  Enter **y** (default) to ignore the reporting of ineffective call attempts. Ineffective call attempts are calls originated by a voice terminal user 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 and incoming calls rejected by the switch due to NSF mismatch. Enter **n** to report ineffective call attempts.

- *Outg Trk Call Splitting*

  Enter **y** to enable splitting of call records for outgoing calls that are transferred or conferenced.

- *Disconnect Information in Place of FRL*

  Enter **y** to record disconnect information or **n** (default) to record the call’s FRL.

NOTE:
This field changes the Facility Restriction Level (FRL) field of the CDR record to record the reason for a disconnect of a call rather than the FRL. It is used for isolating problems between the MCU and the network.

- *Digits to Record for Outgoing Calls*

  Enter **dialed** or **outpulsed** to record the dialed or outpulsed digits in the Dialed Number field of the CDR.

- *Privacy — Digits to Hide*

  Enter the number of digits (**0** through **7**) to be blanked from the back (right to left) of the *Dialed Number* field on the CDR record. Default is **0**. For example, if **4** is entered into this field and the user dials “555-1234,” only “555” appears in the *Dialed Number* field of the CDR record. CDR Privacy must be enabled on the Station form.

Administering System Features

The Feature-Related System Parameters form sets parameters for various system-wide features. These parameters are administered at system initialization, and they may be changed for ongoing administration.

The following table provides the relevant commands.

Table 0-11. System Parameters Features Administration Commands

Action	Object	Qualifier
change	system-parameters features	
display	system-parameters features	print or schedule

To change the Feature-Related System Parameters form, follow these procedures:

1. At the *command:* prompt, type **change system-parameters features** and then press RETURN.
2. Verify that the screen displays the Feature-Related System Parameters screen.
3. Use TAB or RETURN to advance to the fields you want to change.
4. When the form is completed, verify that the MT displays: *Command successfully completed* followed by *command:*
5. To verify the form, type **display system-parameters features** and then press RETURN.

Features-Related System Parameters Form

The first page of the Features-Related System Parameters form appears as follows:

```
change system-parameters features                          Page 1 of 7

                    FEATURE-RELATED SYSTEM PARAMETERS

          Conference Password Query Time-out (seconds): 60
             Dial-out Failure Retry Interval (minutes): 1
                  Dial-out Overall Failure Retry Limit: __
              Dial-out Post Answer Failure Retry Limit: 9
                         Default Custom Lip Sync  <= 128K: 0
                         Default Custom Lip Sync 224/256K: 0
                         Default Custom Lip Sync  >= 336K: 0


                             AAR/ARS Dial Tone Required? y
                                System Audio Preference: Mu-law
                              MCU Pacer Interval (msec): 2000

                                                Forward AIM: y
           Automatic Circuit Assurance (ACA) Enabled? n
```

Figure 0-11. Feature-Related System Parameters Form (Page 1)

The following list identifies and discusses the fields on the Features-Related System Parameters form.

Feature-Related System Parameters Form, Page 1 of 2

- *Conference Password Query Timeout (seconds)*

  Enter the number of seconds (**30** through **300**) that the MCU will wait for a conference password response before timing out. The default is **60**.

  NOTE:
  A value greater than **60** may be required for some BONDing conferences, which may take a long time to set up.

- *Dial-Out Failure Retry Interval (minutes)*

  Enter the number of minutes (**1** through **5**) that the MCU will wait between dial-out attempts. The default is **1**.

- *Dial-Out Overall Failure Retry Limit*

  Enter the sum of the pre-answer and post-answer retries (**0** through **200**) by the MCU. Once this number is reached, the MCU stops calling. The default is a blank entry, which indicates that there is no pre-answer limit.

- *Dial-Out Post-Answer Failure Retry Limit*

  Enter the number of times (**0** through **9**) that the MCU should attempt to redial an endpoint that answers but is then dropped. The default is **4**.

- *Default Custom Lip Sync <=128K*

  Enter in msec the desired Lip Synchronization delay for MCU conferences operating at the rate of 128k or lower. Values include **0** (default) through **1000** in intervals of 20.

- *Default Custom Lip Sync 224/256K*

  Enter in msec the desired Lip Synchronization delay for MCU conferences operating at the 224/256k rate. Values include **0** (default) through **1000** in intervals of 20.

- *Default Custom Lip Sync >=336*

  Enter in msec the desired Lip Synchronization delay for MCU conferences operating at the rate of 336k or higher. Values include **0** (default) through **1000** in intervals of 20.

- *AAR/ARS Dial Tone Required*

  Enter **y** when a second dial tone is required on an incoming tie trunk call that is routed by AAR/ARS.

- *System Audio Preference*

  Values include **mu-law** (default) and **a-law**. The effect of this field is system-wide and affects all cascade links.

  Enter **a-law** whenever the MCU is used to cascade to non-Lucent Technologies MCUs at the lower bandwidths (for example, 2x64k) to avoid media problems on the cascade link. The field value may need to change for conferences sending G.711 audio mode across a cascade link with a non-Lucent MCU. If both Lucent-to-Lucent MCUs and Lucent-to-non-Lucent MCUs are being used, setting the value to **a-law** does not cause any problems on the Lucent-to-Lucent MCU.

- *MCU Pacer Interval (msec)*

  Enter in msec the time delay for dropping MCU calls in general and for sending Dial-Out calls. Values include **2000** (default) through **10000**.

- *Forward AIM*

  Values include **y** (default) and **n**. Certain releases of endpoints produce an audible tone whenever they receive audio indicated mute (AIM). To prevent this audible tone, set this field to **n**. For value **y**, AIM is sent to an endpoint when all other connected sites in the conference have sent AIM to the MCU. The effect of the value in this field is system-wide and affects all conferences.

- *Automatic Circuit Assurance (ACA) Enabled*

  N/A for the MCU. Leave the default value **n**.

```
change system-parameters features                          Page 2 of 2

                        FEATURE-RELATED SYSTEM PARAMETERS

SYSTEM PRINTER PARAMETERS
              System Printer Extension:                Lines Per Page: 60
                   EIA Device Bit Rate: 9600


SYSTEM-WIDE PARAMETERS

                             Switch Name:
```

Figure 0-12. Feature-Related System Parameters Form (Page 2)

Feature-Related System Parameters Form, Page 2 of 2

NOTE:
Be sure to refer to the “Call Detail Recording” section in Appendix B for information on Call Detail Recording.

- *Printer Extension*

  Enter the data module extension number associated with the system printer, or enter **eia** when the DCE jack is used to interface with the printer.

- *EIA Device Bit Rate*

  Enter the required printer speed setting. Values include **1200**, **2400**, **4800**, and **9600**.

- *Lines Per Page*

  Specify the number of lines per page required for the report (**24** through **132**). The default is **60**.

- *Switch Name*

  Enter the name of the switch.

Administering International-Based Parameters

Parameters associated with specific international call characteristics can be implemented by completing the System Parameters Country-Options form. The following table lists the commands that can access the form.

Table 0-12. International-Based Parameters Commands

Action	Object	Qualifier
change	system-parameters country-options	____
display	system-parameters country-options	['print' or 'schedule']*

* Brackets ([ ]) indicate that the qualifier is optional. Single quotes (' ') indicate that the text inside the quote must be entered either exactly as shown or in the appropriate abbreviated form.

To administer international-based parameters, follow these procedures:

1. At the *command:* prompt, enter **change *system-parameters country-options***.
2. Verify that the screen displays the *System Parameters Country-Options Form*.
3. Use TAB or RETURN to advance to the fields you want to change. (See *System Parameters Country-Options Form* for a description of the fields and values.)
4. Submit the form.
5. To verify the form, type **display *system-parameters country-options***.

System-Parameters Country-Options Form

The System-Parameters Country-Options form appears as follows:

```
change system-parameters country-options              Page  1 of  7
                    SYSTEM-PARAMETERS COUNTRY-OPTIONS


     Companding Mode: Mu-Law           Base Tone Generator Set: 1
 440Hz MCU-dial Tone? n              440Hz Secondary-dial Tone? n
   Digital Loss Plan: 1           Version of Digital Loss Plan: B


TONE DETECTION PARAMETERS
  Tone Detection Mode: 1      Dial Tone Validation Time (msec): 600
     Interdigit Pause: short
```

Figure 0-13. System-Parameters Country-Options Form (Page 1)

```
change system-parameters country-options                Page  2 of  7
                    SYSTEM-PARAMETERS COUNTRY-OPTIONS

      Tone Name      Cadence        Tone
                      Step     (Frequency/Level)
_______________         1:     _______________
                        2:     _______________
                        3:     _______________
                        4:     _______________
                        5:     _______________
                        6:     _______________
                        7:     _______________
                        8:     _______________
                        9:     _______________
                       10:     _______________
                       11:     _______________
                       12:     _______________
                       13:     _______________
                       14:     _______________
                       15:     _______________
```

Figure 0-14. System Parameters Country Options Form (Page 2)

NOTE:
Pages 2 through 7 of the Country-Options form are virtually identical.

The following list identifies the fields on the form and the possible values:

- *Companding Mode*

  Enter **A-law** (UK or Australia) or **Mu-law** (US or Canada) to identify the companding algorithm to be used by system hardware. Default is **Mu-law**.

- *Base Tone Generation Set*

  Enter a country code (either **1** for the US, **2** for Australia, **10** for the UK, or **13** for Germany) to identify the base tone generator set to be used. Default is **1**.

- *440Hz MCU-Dial Tone*

  Enter **y** or **n** to specify whether or not the MCU-dial tone will be changed to a continuous 440Hz/-17 tone. The default is **n**. This value implies that the tone will be administered either on a later page of the form or (if no individual definition is administered) as defined in the Base Tone Generator Set.

- *440Hz Secondary-Dial Tone*

  Enter **y** or **n** to specify whether or not the Secondary (CO) dial tone will be changed to a continuous 440Hz/-17 tone. The default is **n**. This value implies that the tone will be administered either on a later page of the form or (if no individual definition is administered) as defined in the Base Tone Generator Set.

- *Digital Loss Plan*

  Enter a country code (either **1** for the US, **2** for Australia,**10** for the UK, or **13** for Germany) to identify the digital loss plan to be used by the system. Default is **1**.

- *Version of Digital Loss Plan*

  This field is not displayed unless *Digital Loss Plan* has entry **10**. If the field is displayed, enter either **A** or **B.** Default is **B.**

The next two fields control tone detection on the system:

- *Tone Detection Mode*

  Enter a code [either **1** for Italy (precise), **2** for Australia (precise), **3** for the UK (precise), **4** for normal broadband (imprecise), **5** for wide broadband (imprecise), or **6** for the US (precise)] to specify the type of tone detection used on a Tone Detector circuit pack. Default is **1**.

- *Dial Tone Validation Timer*

  This pop up field appears only if *Tone Detection Mode* is set to **4** or **5**. In such a case, a value in ms between **0** and **6375** in an increment of 25 appears. Default is **600**.

- *Interdigit Pause*

  Enter short (5 to 30 ms) or long (20 to 20 ms) to specify the maximum length of the interdigit pause. Breaks lasting a shorter time than indicated in these ranges are bridged or ignored. Default is **short.**

- *Tone Name*

  Enter one of the keywords in the following dash list to indicate which of the individually administrable tones the form modifies. If this field is blank, all entries in the corresponding Tone Frequency/Level fields are ignored. Default is blank.

Acceptable keywords include the following:

— 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
— MCU-dial
— recall-dial
— recall-dont-ans
— redirect
— reorder
— rep-confirmation
— ringback
— secondary-dial
— whisper-page
— zip

- *Cadence Step*

  This display-only field identifies the number (1 through 15) of each tone cadence step.

- *Tone (Frequency/Level)*

  Enter **silence**, **goto**, or a set of values to specify the frequency and level of the tone. The entry **silence** indicates no tone. The entry **goto** indicates to repeat all or part of the sequence, beginning at the specified cadence step. A final step of **silence** with an infinite duration is added internally to any tone sequence that does not end with **goto**. Default is blank.

The following list identifies acceptable numeric value sets for the tone field.

- 330/-8.0
- 330/-5+440/-9
- 330/-8+440/-11
- 350/-17.25
- 350+425/-4.0
- 350+440/-13.0
- 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/-5.0
- 425/-8.0
- 425/-11.0
- 425/-17.25
- 440/-11.0
- 440/-13.0
- 440/-17.25
- 440+480/-13.0
- 440+480/-19.0
- 480/-17.25
- 480/-30.0
- 480+620/-13.0
- 480+620/-24.0
- 525/-11.0
- 620/-17.25
- 697/-8.5
- 770/-8.5
- 852/-8.5
- 941/-8.5
- 950/-5.0
- 950/-10.0
- 1000/0.0
- 1000/+3.0
- 1004/0.0
- 1004/-16.0
- 1209/-7.5
- 1336/-7.5
- 1400/-5.0
- 1400/-10.0
- 1400/-11.0
- 1477/-7.5
- 1633/-7.5
- 1700/-16.0
- 1800/-5.0
- 1800/-10.0
- 2025/-12.1
- 2100/-12.1
- 2225/-12.1
- 2804/-16.0
- goto
- silence

Administering Network Facilities

All video calls connecting to the MCU are transported on Nx56k or Nx64k digital circuits. DS1 or UDS1 trunks must be used to connect the MCU to either a PBX or the public network (interexchange carrier or local exchange company). Provisioning the trunk parameters must be done in conjunction with ordering the network service, provisioning the network access facility, and provisioning call routing.

Defining New ISDN Services

The Network Facilities form allows you to define new ISDN services. The following command can be used with this form:

Table 0-13. ISDN-PRI Network Facilities Commands

Action	Object	Qualifier
change	isdn network-facilities	
display	isdn network-facilities	

Use the following procedure to change the Network-Facilities form when ISDN services are being used:

CAUTION:
Do not change this form unless instructed to do so by QPPCN or Lucent Technologies Bell Laboratories.

1. At the *command:* prompt, type **change isdn network-facilities** and then press RETURN.
2. Verify that the screen displays the Network-Facilities form.
3. From the *Name* field, use TAB and RETURN to advance to the fields you want to change.
4. When the form is completed, verify that the MT displays: *Command successfully completed* followed by *command:*
5. To verify the form, type **display isdn network-facilities** and then press RETURN.

Network Facilities Form

The following list shows the fields on the Network Facilities form and their related values.

- *Name*

  Specify the name of the indicated service/feature (up to 15 characters). Upper and lowercase characters are important; enter the name of the service or feature exactly as given.

- *Type*

  Value **0** indicates the associated entry is a feature. Value **1** indicates the associated entry is a service. Value **2** indicates an incoming call, and value **3** indicates an outgoing call.

- *Coding*

  Specify the encoding value of the associated feature or service.

Setting Network Service and Access Parameters

The trunk signaling used to access the network is a determining factor when subscribing to the network service and when planning the transfer rates to be supported. The MCU supports ISDN-PRI, Channel Associated Signaling (CAS), and Robbed-Bit Signaling (RBS) trunks. Based on the number of video ports configured, the MCU may have multiple T1 or E1 trunks. Based on the number of ISDN-PRI trunks configured, ISDN-PRI NFAS is supported for T1 trunks but not for E1 trunks, and D-channel backup features are supported domestically.

DS1 signaling is commonly categorized as RBS signaling (also known as out-of-band and common channel signaling). CAS signaling can be administered for the UDS1 TN2207 circuit pack with bit rate 2.048 and is the common channel signaling for E1 transmission facilities, which are used to facilitate international capabilities on the MCU.

NOTE:
CAS signaling can be used in a specific country only if it is approved there.

The TN2207 circuit pack does not support RBS. The TN767 circuit pack does not support ISDN over PACCON.

In addition to this signaling type, other parameter settings on the DS1 or UDS1 circuit pack help determine what transmission speed is supported on the DS1 or UDS1.

The DS1 circuit pack provides a DS1 format, digital, multichannel interface between the MCU and T1 transmission facilities. Alternatively, the circuit pack can be directly connected to a local DS1 interface (behind a PBX or on a private network). The same is true for the UDS1 circuit pack, except that it can be used with both T1 and E1 transmission facilities. DS1 or UDS1 Trunk Service provides a set of digital trunks using the DS1 (TN767E) or UDS1 (TN2207) circuit pack that support MCU communications. Digital interfaces are supported for the following types of trunks:

- **ISDN-PRI**

  When the DS1 (UDS1) interface provides ISDN-PRI trunk group service, a TN767E DS1 (TN2207 UDS1) circuit pack can be used in conjunction with a TN778 Packet Control (PACCON) circuit pack. In such a case, the TN765 Processor Interface circuit pack is not required. ISDN-PRI trunks provide end-to-end digital connectivity within the network and can provide national and international digital connectivity to other ISDNs that support the PRI standard. This mode of signaling supports 23 or 30 56k or 64k trunks for video transmission and one trunk for signaling.

- **DS1 tie trunks**

  The TN767E DS1 circuit pack supports DS1 tie trunks in RBS mode. The RBS mode supports 24 trunks for transmission on the circuit pack because the least significant bit (robbed) in every sixth frame of data transmission is replaced by a signaling bit. This limits a trunk to support 56k bandwidth for each B-channel. The TN2207 DS1 circuit pack supports DS1 tie trunks in CAS mode.

- **ISDN-BRI Trunk Interface**

  This interface supports ISDN-BRI connections to the public network. It requires the TN2185 circuit pack, which supports point-to-point calls at the 1B and 2B rates.

Administering DS1 or UDS1 Circuit Packs

DS1 circuit pack administration is required for all DS1 circuit packs that are installed in the MCU.

Two different types of DS1 circuit packs can be installed into the Lucent Technologies MCU, as follows:

- **TN767**

  This is a 24-channel DS1 that is used for data and voice in both non-ISDN and ISDN applications. This circuit pack can be used with both RBS and ISDN-PRI facilities. This circuit pack cannot be used whenever the ISDN-PRI over PACCON feature is administered.

- **TN2207**

  This is a 24- or 32-channel DS1 used for data and voice in non-ISDN and ISDN applications. This circuit pack is required whenever the ISDN-PRI over PACCON feature is administered. (Refer to the "Administering ISDN-PRI Over the Packet Control Circuit Pack" section for details.) The TN2207 circuit pack is also required for ESM installation. (Refer to the "Administering the ESM from the MSM" section for details.)

The following table provides the commands that relate to the DS1 or UDS1 circuit pack.

Table 0-14. DS1 (UDS1) Circuit Pack Administration Commands

Action	Object	Qualifier
add	ds1	location*
change	ds1	location*
display	ds1	location*
list	measurements ds1[-log]	location*
remove	ds1	location*

* Where ***location*** is the DS1 (UDS1) circuit pack location obtained by entering the **list configuration all** command.

To administer parameters for the TN767E DS1 or TN2207 UDS1 circuit pack, follow these procedures:

1. At the *command:* prompt, enter **add ds1 *location*** where ***location*** is the location of the TN767E DS1 or TN2207 UDS1 circuit pack in the carrier.
2. Verify that the screen displays the *DS1 Circuit Pack Form*.
3. Use TAB or RETURN to advance to the fields you want to change (see *DS1 Circuit Pack Form* for a description of the fields and values).
4. Submit the form.
5. To verify the form, type **display ds1 *location***.

DS1 (UDS1) Circuit Pack Form

The DS1 (UDS1) Circuit Pack form is used to administer parameters on the TN767E DS1 or TN2207 UDS1 circuit pack. The circuit pack must be installed before it can be administered.

Also, you must first administer a DS1 circuit pack before you can administer an access-endpoint that uses a DS1 circuit pack for connectivity. DS1 access endpoints connect to the MCU via non-signaled T1 or E1 facilities. Non-signaled endpoints connected to the MCU may be Multiplexers (MUXs), Digital Crosspoint Access Systems (DACSs), and/or H.320 direct T1/E1 codecs.

NOTE:
For more information on access endpoint administration, refer to the "Administering Access Endpoints" section in this chapter.

When you are administering DS1s, ensure that both ends of the T1/E1 circuit are administered correctly. *If you have to make changes to the DS1 form, be sure to remove first all of the related administration forms, including the following: Signaling Group form, Interface Links form, Synchronization Plan form, and Processor Channel Assignments form. Also, be sure to remove the trunk members that are associated with the DS1*.

NOTE:
Before you administer all of the translations that are associated with the DS1 circuit pack, it is recommended that you administer minimum translations and then test to see if the D-channel comes up at both ends.

```
add ds1 1a08                                                Page 1 of 1

                                DS1 CIRCUIT PACK

                Location: 01A08                                 Name: __________
                Bit Rate: 2.048                       Line Coding: hdb3

              Signaling Mode: isdn-pri
                     Connect: network
                                                  Country Protocol: 1
                                                  Protocol Version: a
        Interface Companding: mulaw                            CRC? n
                   Idle Code: 11111111

                                        DCP/Analog Bearer Capability: 3.1kHz
           MMI Cabling Board: ____                  MMI Interface: network

                              MAINTENANCE PARAMETERS
              Slip Detection? n
```

Figure 0-15. DS1 Circuit Pack Form (E1 Signaling)

```
add ds1 1a08                                                Page 1 of 1

                                DS1 CIRCUIT PACK

              Location: 01A08                              Name: _____________
              Bit Rate: 1.544                       Line Coding: b8zs
     Line Compensation: 2                           Framing Mode: esf
        Signaling Mode: isdn-pri
               Connect: network
                                                Country Protocol: 1
                                                Protocol Version: a
                                                             CRC? n
            Idle Code: 11111111
                                  DCP/Analog Bearer Capability: 3.1kHz
     MMI Cabling Board: ____                      MMI Interface: network
                              MAINTENANCE PARAMETERS
        Slip Detection? n
```

Figure 0-16. DS1 Circuit Pack Form (64-Channel DS1)

```
add ds1 1a08                                              Page 1 of 1

                                DS1 CIRCUIT PACK

            Location: 01A08                              Name: _______________
            Bit Rate: 2.048                      Line Coding: b8zs

       Signaling Mode: isdn-ext_____


 Interface Companding: mulaw
           Idle Code: 11111111


    MMI Cabling Board: ______                   MMI Interface: network


                            MAINTENANCE PARAMETERS

       Slip Detection? n
```

Figure 0-17. DS1 Circuit Pack Form (T1 Signaling and External D-Channel)

NOTE:
The *Slip Detection?* field should contain **y** for this variation of the form. Also, *Line Coding* contains **b8zs**. Whenever *Bit Rate* contains **2.048**, *Framing Mode* does not appear.

```
change ds1 a07                                          Page 1 of 1

                              DS1 CIRCUIT PACK

            Location: 01A07                              Name: __________
            Bit Rate: 1.544                      Line Coding: ami-zcs
   Line Compensation: 1                         Framing Mode: esf
      Signaling Mode: robbed-bit




           Idle Code: 11111111

   MMI Cabling Board:                          MMI Interface: network


                           MAINTENANCE PARAMETERS
      Slip Detection? n
```

Figure 0-18. DS1 Circuit Pack Form (T1-RBS)

NOTE:
The *Slip Detection?* field must should contain **y** for this variation of the form.

```
change ds1 a07                                          Page  1 of  1

                                DS1 CIRCUIT PACK

            Location: 01A07                              Name:
            Bit Rate: 2.048                    Line Coding: hdb3
   Line Compensation: 1                       Framing Mode: esf
      Signaling Mode: isdn-pri
             Connect: pbx                          Interface: network
                                          Country Protocol: 1
                                          Protocol Version: a
Interface Companding: mulaw                             CRC? y
           Idle Code: 11111111
                                  DCP/Analog Bearer Capability: 3.1kHz
   MMI Cabling Board:                        MMI Interface: ESM


                            MAINTENANCE PARAMETERS

      Slip Detection? y
```

Figure 0-19. DS1 Circuit Pack Form (ESM)

NOTE:
The *CRC?* field must contain **y** for this variation of the form. Also, it is recommended that the *Slip Detection?* field contain **y**. Finally, for international systems, *Country Protocol* must contain **1**.

The following list identifies the fields on the DS1 Circuit Pack form and the possible values.

NOTE:
Be sure to check the notes following the previous four figures regarding required or suggested values for some of the fields.

- *Location*

  This display-only field shows the location of the DS1 or UDS1 interface circuit pack. The first character identifies the network; the second character identifies the carrier; the third and fourth characters identify the slot number in the carrier; and the last two characters identify the circuit number.

- *Name*

  Enter the name of the DS1 or UDS1 link (up to 15 characters). This is usually the destination of the link (for example, **Access-Ept #2**).

- *Bit Rate*

  Enter **1.544** for the 24-channel DS1 or UDS1 rate. Enter **2.048** for the 30-channel DS1 rate.

  NOTE:
  If the field contains **2.048**, the *Framing Mode* field does not appear in the form.

- *Line Coding*

  This entry indicates which line coding format is used to ensure that the data meets T1- or E1-carrier requirements. For all bandwidths of 64k increments, enter **b8zs** for bipolar eight zero suppression. For a robbed-bit DS1 circuit pack, enter **ami-zcs** (also known as AMI). For 2.048 bit rates, enter **hdb3** or **ami-basic**. The selection must match the method used on the other end of the link. Value **b8zs**, **ami-basic**, or **ami-zcs** is allowed when *Bit Rate* is set to **1.544**.

  When the DS1 circuit pack is used for ISDN, the ISDN D-channel data is inverted when **ami-basic** or **ami-zcs** is entered, and it is not inverted when **b8zs** or **hdb3** is entered.

- *Line Compensation*

  Enter the appropriate number in the range of **1** through **5** as appropriate for the cable length. The cable length is one half the distance from the TN767E DS1 or TN2207 UDS1 circuit pack to the CSU.

  — **1** = 000 to 133 feet

  — **2** = 133 to 266 feet

  — **3** = 266 to 399 feet

  — **4** = 399 to 533 feet

  — **5** = 533 to 655 feet

  NOTE:
  This field appears only whenever *Bit Rate* contains **1.544**.

- *Framing Mode*

  The network type should indicate which choice is correct for the particular DS1/T1 span. Enter **d4** or **esf** to match the method used on the other end of the link. For bandwidths of 64k or greater, set this field to **esf**. For RBS, set this field to **d4.**

  NOTE:
  This field appears only whenever *Bit Rate* contains **1.544**.

- *Signaling Mode*

  For 56k increments, enter **robbed-bit**. For 56k or 64k increments, enter **isdn-ext.** This must match the other end of the circuit. Value **robbed-bit** supports in-band signaling, where 24 channels may be used for voice and the least significant bit of the eight-bit channel codes is “robbed” in every sixth frame to transmit signaling status. This value supports only 56k increments. For ISDN signaling, use **isdn-pri** if the DS1 trunk contains a D-channel. A total of 23 (T1) or 30 (E1) B-channels are provided on each circuit pack. Enter **isdn-ext** if NFAS is being used. ISDN extension signaling slaves off of another DS1 circuit pack for D-channel messages, and it allows all 24 DS0’s (T1) to be used as B-channels. NFAS using E1 (2.048k) can be supported between the MCU and a PBX for increased channel capacity if the PBX supports both E1 and the NFAS feature and if the MCU and PBX are connected back to back.

  NOTE:
  If NFAS is used with E1 (2.048) signaling, the 16th DS0 normally used for a D-Channel should never be used as a DS0. Skip over the 16th channel when administering the DS1.

  CAS signaling is the signaling mode for the international E1 (2.048) bit rate.

  NOTE:
  CAS can be used in a specific country only if it is available there.

- *MMI Cabling Board*

  This field displays the code for the MMI cabling board (for example, **01A10**). One MMI is required for each access endpoint using the 672k rate or higher.

- *Connect*

  This field appears only when **isdn-pri** is selected, and it specifies what is on the far end of the MCU in order to control Layer 3 protocol. Allowable entries are **network** and **pbx**. If the MCU is connected directly to the network, enter **network**. If the MCU is connected behind a PBX, enter **pbx**. Enter **line-side** to enable the switch to function as the network side of the ISDN-PRI.

- *Interface*

  This field appears only when the *Connect* field value is **pbx**. Values include **network** and **user**. Enter **network** to set the MCU to use the network end of the protocol. Enter **user** to set the MCU to use the protocol of the opposite end.

  NOTE:
  One end must be administered using **pbx/user**, the other end using **pbx/network**.

- *Country Protocol*

  This field appears if **isdn-pri** or **CAS** appears for *Signaling Mode*. The table at the end of this bullet list identifies the appropriate country code values, the associated country or region, and the protocol supported for each code.

  NOTE:
  If the field contains value **1**, **10**. **12**, **13**, or **99**, refer to the description for the *Protocol Version* field later in this list for additional administration.

- *MMI Interface*

  Possible values include **pi_esm, pac_esm** and **network** (default). Values **pi_esm** and **network** indicate an ESM connection over the processor interface (PI) link. Also, **pi_esm** indicates that the DS1 circuit pack is cabled to the ESM-MMI circuit pack, while **network** indicates that the DS1 circuit pack is cabled to the H.221-MMI circuit pack. For access-endpoint DS1 connections, leave set to **network**. Value **pac_esm** indicates an ESM connection over the PACCON link.

  If you enter **pi_esm**, the following applies:

  — "ESM PARAMETERS" section (which includes *Sanity Timeout, Startup Timeout*, and *Run State*) appears at the bottom of the form.

  — *Bit Rate* must contain **2.048**.

  — *Line Coding* must contain **hdb3**.

  — *Signaling Mode* must contain **isdn-pri**.

  — *Connect* must contain **pbx**.

  — *Interface* must contain **network**.

  — MMI Cabling Board must be administered.

- *Protocol Version*

  Enter **a**, **b**, or **c**, as appropriate. Refer to the table at the end of this bullet list. This field appears whenever the *Country* field on the Trunk Group form contains **1**, **10**, **12**, **13**, or **99**. Enter **a** for NI-1 and AT&T custom; enter **b** for NI-2. Enter **c** only when the country protocol is 1 (USA).

- *Interface Companding*

  Indicates the companding mode. Enter **mulaw** (default) for a domestic MCU; enter **alaw** whenever this is the required protocol.

  NOTE:
  This field appears only whenever a TN2207 circuit pack is being used.

- *CRC*

  Enter **y** to indicate that a cyclic redundancy check is to be performed on transmissions that the circuit pack receives.

- *Idle Code*

  Enter the 8-digit string (for example, **11111111**) to be sent out on idle DS0 channels. The string to be entered depends on the country where the trunk terminates. Default is **11111111** (US domestic setting).

- *DCP/Analog Bearer Capability*

  Displays when Signaling Mode is set to **isdn-pri**. This field is used to determine bearer capability encoding. Enter **3.1kHz** (default) or **speech**.

- *Slip Detection*

  Enter **y** to maintain a slip count record for each DS1 or UDS1 interface. The slip count determines if the T1- or E1-span is experiencing errors and, if so, the severity of the errors (type alarm).

- *Near-End CSU Type*

  Appears only when the DS1 circuit pack is a TN767E or later suffix model, *Bit Rate* is set to **1.544,** and *Country Protocol* is set to **1** (U.S.). Enter value **other** or **integrated**. Value **other** indicates that no CSU is attached to the DS1 facility or that the CSU is an external unit. No options are available for administering the CSU. Value **integrated** indicates that a 120A CSU module is attached to the DS1 circuit pack. This integrated CSU (ICSU) can accept software-administrable option downlinks and perform the enhanced ICSU functionality if the correct suffix of DS1 circuit pack (TN767E or later) is either administered via a Circuit Pack form or installed in the MCU. When you enter value **integrated**, a second page of the DS1 Circuit Pack form is available to administer CSU module options.

Table 2-15. Country Protocols

Country Code	Country or Region	Protocol(s) Supported
1 - a	US and Canada	AT&T TR 41449/41459 (tested with AT&T network, Canadian network, and MCI network
1- b	US	Bellcore TR1268, NIUF.302, ANSI T1.607
1 - c	US	NORTEL DMS-250 BCS36/IEC01
2	Australia	AUSTEL TS014.1, Telecom Australia TPH 1856
3	Japan	NTT INS 1500
4	Italy	ETS 300 102
5	Netherlands	ETS 300 102
6	Singapore	ETS 300 102
7	Mexico	ETS 300 102
8	Belgium	ETS 300 102
9	Saudi Arabia	ETS 300 102
10 - a	United Kingdom	ETS 300 102 (for connection to DASS II/DPNSS through external converter)
10 - b	United Kingdom	ETS 300 102 (Mercury), British Telecom ISDN 30
11	Spain	Telefonica ISDN Specification
12 - a	France	VN4 (French National PRI)
12 - b	France	ETS 300 102 modified according to P10-20, called Euronumeris
13 - a	Germany	FTZ 1 TR 6 (German National PRI)
13 - b	Germany	ETS 300 102
14	Czech Republic, Slovakia	ETS 300 102
15	Russia (CIS)	ETS 300 102
16	Argentina	ETS 300 102
17	Greece	ETS 300 102
18	China	ETS 300 102
19	Hong Kong	ETS 300 102
20	Thailand	ETS 300 102

Table 2-15. Country Protocols — *Continued*

Country Code	Country or Region	Protocol(s) Supported
21	Macedonia	ETS 300 102
22	Poland	ETS 300 102
23	Brazil	ETS 300 102
98	National ISDN	ETS 300 102
ETSI - a	Europe, New Zealand, etc.	ETS 300 102
ETSI - b		ETS 300 102

Providing a Synchronization Plan

The Synchronization Plan form is used to assign primary and secondary external synchronization sources. It provides synchronization between the MCU and the digital facilities it uses. For details, refer to the "Administering Synchronization" section later in this chapter.

Administering Data Communications

For configurations that include an MCU-ST, various terminals, system printer, or the PC-based CRCS, use the procedures in the following sections.

Establishing Network Control Channel Data Modules

Network control (netcon) channel data modules provide asynchronous circuit-switched interfaces for the following: maintenance and administration terminals, system printer, CRCS, Op-Center, and CDR. These data modules are integrated into the TN777 Network Control circuit pack. You should assign four netcon data modules. Use the Data Module form to establish network control channel data modules.

NOTE:
Once you administer the netcon data modules, you must administer a hunt group with the four netcon data module extensions to provide a common number for access to the netcon channels.

The following commands relate to the data modules:

Table 2-16. Netcon Channel Data Module Administration Commands

Action	Object	Qualifier
add	data-module	xxxx (extension or next)
change	data-module	xxxx (extension)
display	data-module	xxxx (extension) print or schedule
list	data-module	starting extension number or count or print or schedule
remove	data-module	starting extension number or count or print or schedule

Use the following procedure to establish network control channels:

1. At the *command:* prompt, type **change data-module xxxx** and then press RETURN.
2. Verify that the screen displays the Data Module form.
3. From the *Type* field, use TAB and RETURN to advance to the fields you want to change.
4. When the form is completed, verify that the MT displays: *Command successfully completed* followed by *command:*
5. To verify the form, type **display data-module xxxx** and then press RETURN.

Data Module Form (Netcon Channels)

The Data Module form for establishing network control channels appears as follows:

NOTE:
Be sure to save translations on a regular basis to prevent loss of translations if system either loses power or resets with level 3 or higher.

```
add data-module 6882                                    Page  1 of  1

                                DATA MODULE

     Data Extension: 6882            Name: _____________________________ BCC: 2
               Type: netcon________   COS: 1_      Maintenance Extension: 6885
   Physical Channel: 01              COR: 1_




ASSIGNED MEMBER ( Station with a data extension button for this data module )

    Ext     Name
 1: ____    _______
```

Figure 2-20. Data Module Form (Netcon Channels)

The following list identifies the fields on the Data Module form and the possible values.

- *Data Extension*

  This display-only field provides the extension assigned to the data module.

- *BCC*

  This display-only field provides the Bearer Capability Class (BCC) whenever ISDN-PRI is enabled on the Customer-Options form.

- *Type*

  Enter **netcon**.

- *Physical Channel*

  Enter the two-digit circuit number of the netcon port (**01** through **04**).

- *Ext*

  N/A for the MCU. Leave the default value **restricted**.

- *Name*

  Enter a string containing a maximum of 27 characters to identify the group (for example, **NETCON #2**).

- *COS*

  Enter the desired Class of Service. Leave at default value **1**.

- *COR*

  Enter the desired Class of Restriction. Leave at default value **1**.

- *Maintenance Extension*

  The maintenance extension is not used by the MCU; however, it must be administered when you are adding netcon channels.

  Enter an extension number that accommodates a number sequence vis-a-vis the value in the *Data Extension* field. For example, if the *Data Extension* field contains a value ranging from “6881” through “6884” for netcon ports “01” through “04,” enter a value ranging from **6885** through **6888** into the *Maintenance Extension* field for the four netcon ports.

- *Ext*

  N/A for the MCU. Leave blank.

- *Name*

  N/A for the MCU. Leave blank.

Assigning Hunt Groups

Once you administer the netcon data modules, you must administer a hunt group. This hunt group allows you to dial one number to answer multiple calls. By including the four netcon channels that you previously administered, any of the netcon data modules that are not in use can be used by remote G3-MT terminals, Standalone OpCenter, or CRCS to access the MCU

Whenever it is answering calls, the system checks for the busy or idle status of extension numbers in a hunt group. For the Uniform Call Distribution (UCD) type, the system selects the “most idle” extension in the group when it is answering a new call. For the Direct Department Calling (DDC) type, the system selects the first available extension (in the administered sequence) when answering a new call.

NOTE:
This is the only hunt group that should be administered on the MCU.

Table 2-17. Hunt Group Administration Commands

Action	Object	Qualifier
add	hunt-group	1-32 or next
change	hunt-group	1-32
display	hunt-group	1-32, print or schedule
list	hunt-group	print or schedule
remove	hunt-group	1-32

Use the following procedure to add the hunt groups:

1. At the *command:* prompt, type **add hunt-group x** and then press RETURN.
2. Verify that the screen displays the Trunk Group form.
3. From the *Group Extension* field, use TAB and RETURN to advance to the fields you want to change.
4. When the form is completed, verify that the MT displays: *Command successfully completed* followed by *command:*
5. To verify the form, type **display hunt-group x** and then press RETURN.

Hunt Group Form

An example of page 1 of the Hunt Group form appears as follows.

```
add hunt-group 1                                          Page 1 of 9

                                  HUNT GROUP


           Group Number: 1
             Group Name: 123456789012345678901234567
        Group Extension:
             Group Type: ucd-mia

                    COR: 1
          Security Code:
     ISDN Caller Display:
```

Figure 2-21. Hunt Group Form (Page 1)

The following list identifies the fields on the Hunt Group form and their related values:

Hunt Group Form, Page 1 of 6

- *Group Number*

  Displays the hunt group number (**1** through **75**).

- *Group Extension*

  Enter an unused extension number (including one to five digits) to be assigned to the associated hunt group.

- *Group Type*

  Values include **ddc** and **ucd-mia**.

- *Group Name*

  Enter a 27-character string to identify the group (for example, **remote admin**).

- *Security Code*

  N/A for the MCU. Leave blank.

- *COR*

  Indicates the Class of Restriction. Leave at the default value **1**.

- *ISDN Caller Disp*

  Leave this field blank.

Hunt Group Form, Pages 2 through 9

- *Ext* (1: through 4:)

  Enter the extension for each of the four netcon channel data modules that you previously administered in the Data Module form.

  NOTE:
  Do not use the maintenance extension numbers as members of the hunt group.

- *Name*

  This field is automatically populated with the names administered for the respective data-module extensions identified in the Data Module form after form is submitted.

Establishing Administered Connections

For the MCU-ST and CRCS, use the Administered-Connection form to establish a connection that is automatically reestablished whenever the MCU is restarted. An example of the form appears as follows.

```
add administered-connection 1                          Page 1 of 1

                        ADMINISTERED CONNECTION

      Connection Number: 1                                Enable? y
             Originator:
            Destination:
                   Name:

AUTHORIZED TIME OF DAY
              Continuous? y



MISCELLANEOUS PARAMETERS

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

Figure 2-22. Administered Connection Form

The following commands can be used to administer connections:

Table 2-18. Administered-Connection Administration Commands

Action	Object	Qualifier
add	administered-connection	1-600 or next
change	administered-connection	1-600 or next
display	administered-connection	1-600, next, print or schedule
duplicate	administered-connection	1-600 or next
list	administered-connection	1-600 or next
remove	administered-connection	1-600, count, print or schedule

Use the following procedure to establish administered connections:

1. At the *command:* prompt, type **add administered-connection x** and then press RETURN.
2. Verify that the screen displays the Administered-Connection form.
3. From the *Enable* field, use TAB and RETURN to advance to the fields you want to change.
4. When the form is completed, verify that the MT displays: *Command successfully completed* followed by *command:*
5. To verify the form, type **display administered-connection x** and then press RETURN.

Administered-Connection Form

The following list identifies the fields on the Administered-Connection form and their related values:

- *Connection Number*

  This display-only field shows the Administered-Connection number.

- *Enable*

  Enter **y** to indicate that an attempt will be made to establish the administered connection whenever the MCU is restarted or the connection is due to be active. The default is **y**.

- *Originator*

  Enter the assigned extension of the originating access endpoint or data endpoint. The endpoint is a 7400B data module, the MCU-MT, the MCU-ST or CRCS. This endpoint must be local to the MCU. The default is blank.

- *Destination*

  Enter the address of the destination access or data endpoint (**0** through **9)**. This endpoint does not have to be local to the MCU. If a local endpoint is entered, it must be assigned to either an access or data endpoint. Up to 36 characters can be entered. The default is blank.

  NOTE:
  A FAC is restricted to AAR, ARS, and CDR account code dialing.

- *Name*

  Enter an optional name (up to 15 characters) for the connection, such as **MCU-ST.**

- *Continuous*

  Enter **y** if the administered connection is enabled whenever the MCU is restarted. Enter **n** if you want to schedule the connection to be active only during certain times. If you enter **n**, the following three fields are displayed.

- *Start Days (Sun through Sat)*

  Displays only when **n** is entered in the *Continuous* field. Enter a **y** in each day you want to establish the administered connection; otherwise, enter n. If you enter **y**, the administered connection begins on the day specified and continues for the duration set in the *Duration* field.

- *Start TIme*

  Displays only when **n** is entered in the *Continuous* field. Enter the time of day when an attempt to make the connection should occur. Valid entries are **00:00** to **23:59**. The default is **00:00**.

- *Duration*

  Displays only when **n** is entered in the *Continuous* field. Enter the period of time that the scheduled administered connection should remain active. The maximum duration is one minute less than a week. Valid entries are **000** through **167** for hours, **00** through **59** for minutes. The default is **000:00**.

- *Alarm Type*

  Enter the type of alarm to be generated if the administered connection cannot be established or fails and cannot be reestablished and the number of failures equals the alarm threshold. The valid entries are **major**, **minor**, **warning**, or **none**. The default is **warning**.

- *Alarm Threshold*

  Enter the number of times an attempt to establish or reestablish a connection should be tried before an alarm is raised. Valid entries are **1** through **10.** The default is **5**.

- *Retry Interval*

  Enter the number of minutes between attempts to establish or reestablish the connection. Valid entries are **1** through **60**. The default is **2**.

- *Auto Restoration*

  Enter value **y** to indicate that an attempt will 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. Value **y** (default) is ignored in all other situations.

Administering ISDN D-Channels

ISDN-PRI forms provide end-to-end digital connectivity instructions and support a wide range of services, including voice and non-voice services, to which users have access by a limited set of ITU-T-defined, standard multipurpose user-network interfaces.

The MCU supports two types of ISDN-PRI D-channel messaging. The type of messaging discussed in the following sections uses the TN765 Processor Interface circuit pack as an interface between the network and the MCU processor. The other type of messaging is the ISDN-PRI over the Packet Control (PACCON) feature, which uses the TN778 circuit pack. ISDN-PRI over PACCON increases the number of ISDN-PRI D-Channel links that the MCU can support without using NFAS feature. On the other hand, using the TN765 restricts the MCU to four D-channel connections unless the NFAS feature is used.

NOTE:
If ISDN-PRI over PACCON is not administered, the IDSN-PRI D-channels must be associated with the TN765 Processor Interface circuit pack.

For details on ISDN-PRI over PACCON, refer to the "Administering ISDN-PRI Over the Packet Control Circuit Pack" section.

The forms discussed in this section are required to administer ISDN-PRI D-channels that use a TN765 circuit pack.

Processor Interface Data Modules

The processor interface data modules are integrated into the TN765 Processor Interface circuit pack to provide an ISDN-PRI or ISDN-BRI connection to the MCU processor. This connection is provided via a DS1 interface. One direct EIA connection is available. When used, the physical channel assignment is **01**. *You must assign four processor interface data modules for the MCU.*

NOTE:
If a TN765 circuit pack is not being installed into the MCU, processor interface data module administration is not required.

An SCC-MSM supports only one TN765 Processor Interface circuit pack and physical channels or links 01 through 04 only. The physical channel number is referenced via the extension number that is assigned to the four processor interface data modules for links 1 through 4 in the associated Interface Links form (discussed later).

NOTE:
Besides providing an ISDN-PRI D-channel interface, physical channels 01 through 04 can be used to provide an ESM interface. For details on ESM administration, refer to the "Administering the ESM from the MSM" section.

The following table gives the commands that relate to the data modules.

Table 2-19. Processor Interface Data Module Administration Commands

Action	Object	Qualifier
add	data-module	xxxx (extension or next)
change	data-module	xxxx (extension)
display	data-module	xxxx (extension), print, or schedule
list	data-module	starting extension number, count, print, or schedule
remove	data-module	xxxx (extension or next)

To add a data module to provide an ISDN-PRI interface, follow these procedures:

1. At the *command:* prompt, enter **add data-module *xxxx***.
2. Verify that the screen displays the Data Module form (see *Processor Interface Data Module Form* for a list of the fields and their values).
3. Use TAB or RETURN to advance to the fields you want to change.
4. Submit the form.
5. To verify the form, enter **display data-module *xxxx***.

Data Module Form (Processor Interface)

The Data Module form for establishing processor interface data modules appears as follows:

```
add data-module 6882                                    Page  1 of  1

                                  DATA MODULE

  Data Extension: 6882             Name: _____________________________
            Name: procr-intf___     COS: 1_   Maintenance Extension: 6885
Physical Channel: __                COR: 1_









 ASSIGNED MEMBER ( Station with a data extension button for this data module )

     Ext     Name
  1: ____    _______
```

Figure 2-23. Data Module Form (Processor Interface)

The following list identifies the fields on the Data Module form and the possible values.

- *Data Extension*

  This display-only field provides the extension assigned to the data module.

- *BCC*

  This display-only field provides the Bearer Capability Class (BCC) whenever ISDN-PRI is enabled on the Customer-Options form. Mode **2** defines a rate adaptation scheme for the transport of synchronous and asynchronous data at rate of up to 19.2 kb/s.

- *Type*

  Enter **procr-intf** to provide the MCU with processor interface data channels for ISDN-PRI and ESM D-channels.

- *Physical Channel*

  Enter the two-digit circuit number of the processor interface port (**01** through **04**). **You must administer a total of four processor interface data modules at this time**.

- *Name*

  Enter a string containing a maximum of 27 characters to identify the group (for example, **Proc-infc 01**).

- *COS*

  For the Class of Service, enter or leave the default value of **1**.

- *COR*

  For the Class of Restriction, enter or leave the default value of **1**.

- *Maintenance Extension*

  Enter an extension number that accommodates a number sequence vis-a-vis the value in the *Data Extension* field. For example, if the *Data Extension* field contains a value ranging from “6881” through “6884” for processor-interface ports “01” through “04,” enter a value ranging from **6885** through **6888** into the *Maintenance Extension* field for the four processor-interface ports.

- *Ext*

  N/A for the MCU. Leave blank.

- *Name*

  N/A for the MCU. Leave blank.

- *XID*

  Used to identify layer 2 XID testing capability. Default is **y**.

- *Fixed TEI*

  Used to indicate whether or not 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. For terminals with automatic TEI capability, the associated TEI is assigned by the system. Default is **n**.

- *TEI*

  Displays if *Fixed TEI* contains **y**. Enter a 1- to 2-digit number in the range of **0** through **63**.

- *MIM Support*

  (Management Information Message Support). Used to support two types of capabilities: MIM endpoint initialization capability (SPID support) and other maintenance and management capability. Default is **y**.

  NOTE:
  Value **n** is recommended.

- *Endpt Init*

  Displays only if *MIM Support* is **y**. Used to indicate the terminal's endpoint initialization capability. Endpoint initialization is a procedure, required for multipoint operation, by which the 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 an 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 reused terminals must be programmed to match the administered SPID value. Default is **y**.

- *SPID*

  Displays only if *Endpt Init* contains **y**. The Service Profile Identifier (SPID) is a variable parameter containing a maximum of ten 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 are numeric strings in the range of **0** through **9999999999**. Default is the station's extension number.

- *MIM Mtce/Mgt*

  Displays only if *MIM Support* contains **y** (default). Used to indicate if the terminal supports MIM maintenance and management capabilities other than endpoint initialization.

Establishing Processor Interface Links

The Interface Links form identifies, describes, and, when enabled, activates the associated processor interface link. Interface links provide a physical interface between the MCU and ISDN-PRI D-channels. There is one Interface Link form per MCU.

Whenever you are adding an ISDN-PRI trunk, you must first change the *Enable* field to **n** to disable the link until you complete the administration of the trunk. You must then administer the processor channels and signaling group before coming back to this form to enable the links (use **change communication-interface links** and then enter **y** in the *Enable* field).

The following table provides the commands that relate to the processor interface link.

Table 2-20. Processor Interface Links Administration Commands

Action	Object	Qualifier
change	communication-interface	links
display	communication-interface	links, print, or schedule

To change the Processor Interface Links form, follow these procedures:

1. At the *command:* prompt, enter **change communication-interface links**.
2. Verify that the screen displays the *Interface Links* form (see *Interface Links Form* for a list of the fields and their values).
3. Use TAB or RETURN to advance to the fields you want to change.
4. Submit the form.
5. To verify the form, enter **display communication-interface links**.

Interface Links Form

The Interface Links form appears as follows:

```
change communication-interface links                          Page 1 of 1

                                INTERFACE LINKS

                 Est    PI                 Destination       DTE/
 Link   Enable   Conn   Ext     Prot   Digits       Brd      DCE     Identification
  1:      y       y    6881     ISDN   1A15______  1A05      ___     IN HOUSE D CHAN
  2:      y       y    6882     ISDN   DTE_______  1B08      ___     ESM LINK _______
  3:      y       y    6883     ISDN   DTE_______  1A01      ___     ________________
  4:      n       n    6884     BX25   DTE_______  ____      DTE     ________________
  5:      n       n    ____     BX25   DTE_______  ____      DTE     ________________
  6:      n       n    ____     BX25   DTE_______  ____      DTE     ________________
  7:      n       n    ____     BX25   DTE_______  ____      DTE     ________________
  8:      n       n    ____     BX25   DTE_______  ____      DTE     ________________
```

Figure 2-24. Interface Links Form

The following list provides the fields on the Interface Links form and their related values.

- *Link*

  This display-only field indicates the physical interface link number (**1** through **4**) that connects to ISDN-PRI or an ESM.

- *Enable*

  Initially, enter or leave **n** to enable the link. All other fields on this form may be administered at this point. Once the required fields on the Processor Channel Assignment form (discussed later) are administered, you can enter **y** into this field.

  NOTE:
  When this field entry is changed from **n** to **y**, the *Prot* field is checked. If **ESM** is entered for the *Appl* field in the Processor Channels form, the *Prot* field must be set to **ISDN** before the appropriate link can be established.

- *Est Conn*

  Enter **y** in this field for ISDN signaling.

- *PI Ext*

  This display-only field shows the data extension assigned on the Processor Interface Data Module form.

- *Prot*

  Enter **ISDN** for the protocol type to be established on the link.

- *Digits*

  Leave this field blank.

- *Destination Brd*

  Enter the TN767E DS1 or TN2207 UDS1 circuit pack address that has been administered for ISDN D-channel signaling. The address is in the form ***1xyy***, where ***x*** is the carrier and ***yy*** is the circuit pack slot (for example, **1205**).

  NOTE:
  An ESM DS1 circuit pack (if present) must not be administered as the board destination unless the *Appl* field of the Processor Channels form contains **ESM**.

- *DTE/DCE*

  This field is not used, and it is cleared whenever value **ISDN** is entered into the *Prot* field.

- *Identification*

  Enter a name with a maximum of 15 characters that identifies the purpose of the link (for example, **Lec DS1 span #1**).

Making Processor Channel Assignments

The Processor Channel Assignments form associates the TN765 Processor Interface Circuit Pack with the D-channel of the assigned DS1 circuit pack. Both DS1 D-channels and ESM UDS1 D-channels are assigned on this form. (For information on ESM administration, refer to the "Administering the ESM from the MSM" section.)

NOTE:
If ISDN-PRI over PACCON is being used on the MCU, only the ESM is assigned on this form. IDSN-PRI D-Channels are assigned on this form only if ISDN-PRI over PACCON is not being implemented.

The following table lists the commands that relate to the processor channels.

Table 2-21. Processor Channels Administration Commands

Action	Object	Qualifier
change	communication-interface	processor-channels
display	communication-interface	processor-channels, print, or schedule

To change the Processor Channels form, follow these procedures:

1. At the *command:* prompt, enter **change communication-interface processor-channels**.
2. Verify that the screen displays the *Processor Channels* form (see *Processor Channels Form* for a list of the fields and their values).
3. Use TAB or RETURN to advance to the fields you want to change.
4. Submit the form.
5. To verify the form, enter **display communication-interface-processor channels**.

Processor Channel Assignment Form

The Processor Channel Assignment form appears as follows:

```
change communication-interface processor-channels          Page 1 of 4

                        PROCESSOR CHANNEL ASSIGNMENT

     Proc                    Interface                    Remote
     Chan     Appl.          Link  Chan     Priority      Proc Chan Machine-ID
      1:      isdn            1     _          h             __        __
      2:      esm_            2     _          h             __        __
      3:      isdn            3     _          h             __        __
      4:      ____            _     _          _             __        __
      5:      ____            _     _          _             __        __
      6:      ____            _     _          _             __        __
      7:      ____            _     _          _             __        __
      8:      ____            _     _          _             __        __
      9:      ____            _     _          _             __        __
     10:      ____            _     _          _             __        __
     11:      ____            _     _          _             __        __
     12:      ____            _     _          _             __        __
     13:      ____            _     _          _             __        __
     14:      ____            _     _          _             __        __
     15:      ____            _     _          _             __        __
     16:      ____            _     _          _             __        __
```

Figure 2-25. Processor Channel Assignment Form

The following list provides the fields and values on the Processor Channel Assignment form:

- *Proc Chan*

  This display-only field indicates the 64 channels that are available. For the MCU, use only channels 1 through 4; associate the appropriate channel with the communication-interface link being administered to keep the administration consistent (for example, associate communication-interface link 1 with processor channel 1).

- *Appl*

  Enter **ESM** or **ISDN** to specify the type of application that connects to this processor channel.

  The following points pertain to an **ESM** entry:

  — Only one entry per form is permitted.

  — Entry cannot be administered if there are no T.120 ports administered on the Customer Options form.

  — *Channel* and *Remote Proc Chan* fields must be blank.

  — *Priority* field must be set to **high**.

  — DS1 circuit pack administered in the *Destination Brd* field of the Interface Links form should be an ESM DS1 circuit pack.

- *Link*

  Enter a number (**1** through **4**) that corresponds to the assignment in the Interface Links form. This number identifies the interface link/channel pair used to establish a connection.

- *Chan*

  N/A for the MCU. Leave blank.

- *Priority*

  Enter **h** (high) for all applications that are available on the MCU.

- *Remote Proc Chan*

  Leave this field blank.

- *Machine ID*

  Leave this field blank.

Enabling the Processor Interface Link

At this point, be sure to access the Interface Links form (discussed earlier) and change the *Enable* field to value **y** to enable the link. It may take several minutes for the Processor Interface circuit pack to become active and establish the D-channel.

NOTE:
Whenever a link on the TN765 circuit pack is enabled, all links go out of service and are automatically re-established.

Administering Signaling Groups

The MCU requires an interface between the ISDN-PRI D-channel and the MCU processor. To accomplish this, the MCU supports two types of signaling groups.

The Facility-Associated Signaling (FAS) interface allows a D-channel to convey signaling information for B-channels that are located on the same DS1 facilities as the one containing the D-channel. The Non-Facility Associated Signaling (NFAS) interface allows a D-channel to convey signaling information for B-channels that are located on DS1 facilities other than the one containing the D-channel. With NFAS, the signaling group does not necessarily correspond to a single DS1 interface. (NFAS is currently not supported for E1 trunks.)

NFAS provides a backup D-channel on a separate PRI facility for the NFAS D-channel group. With NFAS, failure of the primary D-channel causes automatic transfer of call-control signaling to the backup D-channel. In such a case, the backup D-channel becomes the primary D-channel. When the failed channel returns to service, it becomes the backup D-channel.

NOTE:
D-Channel backup is not supported internationally for the E1 2.048 bit rate.

The Signaling Group form identifies groups of ISDN DS1 interface B-channels for which a given D-channel or D-channel pair will carry associated signaling information. A maximum of 26 signaling groups at a time can be administered.

The following table lists the commands that can be used with the Signaling Group form.

Table 2-22. Signaling Group Commands

Action	Object	Qualifier
add	signaling-group	1-30 or next
change	signaling-group	1-30
display	signaling-group	1-30, print, or schedule
list	signaling-group	1-30, print, or schedule
remove	signaling-group	1-30

To specify a signaling group, follow these procedures:

1. At the *command:* prompt, enter **add signaling-group *x*** where ***x*** is the signaling group number (**1** through **30**) or **next**.
2. Verify that the screen displays the *Signaling Group* form (see *Signaling Group Form* for a list of fields and values).
3. Use TAB or RETURN to advance to the fields you want to change.

4. Submit the form.
5. To verify the form, enter **display signaling-group *x***.

Signaling Group Form

Two variations of the Signaling Group form appear as follows:

```
add signaling-group next                                    Page 1 of 5

                                SIGNALING GROUP

 Group Number: 1
                      Associated Signaling? y        Max number of NCA TSC: 0
                        Primary D-Channel: 01A1524    Max number of CA TSC: 0
                                                   Trunk Group for NCA TSC: _



Trunk Group for Channel Selection: 1
   Supplementary Service Protocol: a
```

Figure 2-26. Signaling Group Form (Option without NFAS)

```
add signaling-group next                                    Page 1 of 5

                                SIGNALING GROUP

 Group Number: 1      Associated Signaling? n     Max number of NCA TSC: 0
                         Primary D-Channel: _____  Max number of CA TSC: 0
                       Secondary D-Channel:     Trunk Group for NCA TSC: _
          Trunk Group for Channel Selection: 99
             Supplementary Service Protocol: a


           Trunk Brd    Interface ID       Trunk Brd    Interface ID
           1: _____          ___           11: _____         ___
           2: _____          ___           12: _____         ___
           3: _____          ___           13: _____         ___
           4: _____          ___           14: _____         ___
           5: _____          ___           15: _____         ___
           6: _____          ___           16: _____         ___
           7: _____          ___           17: _____         ___
           8: _____          ___           18: _____         ___
           9: _____          ___           19: _____         ___
          10: _____          ___           20: _____         ___
```

Figure 2-27. Signaling Group Form (Option with NFAS)

The following list identifies the fields on the Signaling Group form and their related values.

- *Group Number*

  This is a display-only field that provides the signaling group number used as a qualifier in the commands listed in the previous table.

- *Associated Signaling*

  Enter **n** to indicate NFAS; enter **y** to select FAS. (With FAS, the D-channel can carry signaling information only for B-channels on the same DS1 circuit pack. FAS is used for all E1 trunks.)

- *Max Number of NCA TSC*

  N/A for the MCU. Leave this field set to **0**.

- *Primary D-Channel*

  Enter up to six characters for the port number associated with the DS1 or UDS1 interface circuit pack port. For the DS1 circuit pack, port number 24 is used to assign the primary D-channel in the signaling group; for the UDS1 circuit pack, port 16 is used for the same purpose. Both FAS and NFAS are always the 24th port on the DS1 interface circuit pack.

  NOTE:
  An ESM DS1 port (if present) should not be administered as the primary D-channel.

- *Max Number of CA TSC*

  Leave this field set to **0**.

- *Secondary D-channel*

  When NFAS is selected, this field is displayed. Enter up to six characters for the port number associated with the DS1 interface circuit pack port used for secondary D-channel signaling.

  NOTE:
  An ESM DS1 port (if present) should not be administered as the secondary D-Channel.

- *Trunk Group for NCA TSC*

  This field is not used.

- *Trunk Group for Channel Selection*

  Enter the assigned trunk group number. (An ISDN-BRI trunk group may not be used.) Values include **1** through **99** and blank (default).

- *Supplementary Service Protocol*

  Enter **a** if AT&T Non-Call Associated Temporary Signaling Connections (NCA-TSCs) are being sent over the interface. Value **b** is not used for the MCU.

- *Trunk Brd*

  When using NFAS, enter a four-character DS1 interface circuit pack number that has trunk members belonging to this Signaling Group. The four characters are the port network number, carrier letter, and two-digit slot number (**01** through **18**). The default is blank.

  NOTE:
  An ESM DS1 circuit pack (if present) should not be administered as the primary D-Channel.

- *Interface ID*

  When using NFAS, enter an interface ID (**0** through **31**) for the corresponding DS1 interface circuit pack. The interface ID is referenced by both ends to determine which B-channel is to be used to establish a call; therefore, the interface ID numbers on both ends must be the same. The 4ESS™ switch must start with *Trunk Brd* 1 (which should be the primary D-channel) ID 0. If a secondary D-channel is used, it should be *Trunk Brd* 2 ID 1.

  NOTE:
  Pages 2 through 5 are not used.

  Once you administer the processor channels and signaling groups, enter **y** into the *Enable* field of the Interface Links form.

Administering ISDN-PRI Over the Packet Control Circuit Pack

The MCU supports ISDN-PRI over the Packet Control (PACCON) circuit pack (TN778). This increases the number of PRI links that are supported on the MCU. The TN2207 UDS1 circuit pack is required for this feature.

To enable this capability, enter **y** into the *ISDN-PRI over PACCON?* field in the Customer-Options form. Also, enter **y** into the *Packet Bus Activated?* field in the Maintenance-Related System Parameters form.

NOTE:
Whenever you enable ISDN-PRI over PACCON, it is recommended that all DS1 circuit packs function over the PACCON interface.

If you administer ISDN-PRI over PACCON for an MSM, you *need not administer* the following forms: Data Module form, Interface Links form, and Processor Channel Assignment form.

Maintenance-Related System Parameters Form

An example of the Maintenance-Related System Parameters form appears as follows:

```
change system parameters maintenance                          Page  1 of  3


                    MAINTENANCE-RELATED SYSTEM PARAMETERS


OPERATIONS SUPPORT PARAMETERS
            Product Identification: 1000000000
         First OSS Telephone Number:                   Abbrev Alarm Report? y
        Second OSS Telephone Number:                   Abbrev Alarm Report? n
   Alarm Origination to OSS Number: neither
        Cleared Alarm Notification? n                Suspension Threshold: 5
              Restart Notification? n
           Test Remote Access Port? n
        CPE Alarm Activation Level: none
    Sch Adj Alarm Activation Level: none
              Packet Bus Activated? n
     Customer Access to INADS Port? n
       Repeat Dial Interval (mins): 7

SCHEDULED MAINTENANCE
                  Start Time: 01 : 00                 Stop Time: 06 : 00
          Daily Maintenance: daily           Save Translation: daily
```

Figure 2-28. Maintenance-Related System Parameters Form (Page 1)

```
change system-parameters maintenance                    Page 2 of 3

                    MAINTENANCE-RELATED SYSTEM PARAMETERS

MINIMUM MAINTENANCE THRESHOLDS ( Before Notification )
        TTRs: 4_       CPTRs: 1        Call Classifier Ports: _




ISDN MAINTENANCE
   ISDN_PRI Test Call Extension: ____  ISDN BRI Service SPID: ____


DS1 MAINTENANCE
   DS0 Loop-Around Test Call Extension: ____
```

Figure 2-29. Maintenance-Related System Parameters Form (Page 2)

```
change system-parameters maintenance                    Page  3 of  3

                    MAINTENANCE-RELATED SYSTEM PARAMETERS

          Modem Connection: internal
                 Data Bits: 8
                    Parity: none
```

Figure 2-30. Maintenance-Related System Parameters Form (Page 3)

The fields on the Maintenance-Related System Parameters form are defined in the following paragraphs. Specific entries can be obtained from the Lucent Technologies VTC, technical consultant (if available), and/or the customer.

Page 1: **Operations Support Parameters Area**

- *Product Identification*

  Identifies the MCU to an Operations Support System (OSS), such as INADS or System Network Management Protocol (SNMP). This field must be populated if you want to make changes to this form. The number must start with “1” and contain ten digits.

- *First OSS Telephone Number*

  The first number that the MCU uses for alarm origination. This number can also be used to dial a telephone number when an alarm is raised on the MCU. The number must be obtained from the National Customer Support Center (NCSC) or the Technical Support Center (TSC). Characters # and * are not allowed in the telephone number.

- *Abbrev Alarm Report*

  Enables the Abbreviated Alarm Report feature for the first OSS. Default is **y**.

- *Second OSS Telephone Number*

  The second number that the MCU uses for alarm origination. This number can also be used to dial a telephone number when an alarm is raised on the MCU. The number must be obtained from the National Customer Support Center (NCSC) or the Technical Support Center (TSC). Characters # and * are not allowed in the telephone number.

- *Abbrev Alarm Report*

  Enables the Abbreviated Alarm Report feature for the second OSS. Default is **y**.

- *Alarm Origination to OSS Numbers*

  Indicates one of four options for alarm origination. If this field is set to any value other than **neither** (default), the craft login has a 30-minute inactivity timeout.

  — For value **both**, all major and minor alarms result in an automatic call to both OSS telephone numbers. Both OSS telephone numbers must be administered.

  — For value **first-only**, all major and minor alarms result in an automatic call to the first OSS number only. The switch does not call the second OSS telephone number even if the number is administered. The first OSS telephone number must be administered.

  — For value **neither**, alarm origination does not take place. Warning alarms are not reported to either numbers.

  — For value **second-as-backup**, all major and minor alarms result in an automatic call to the first OSS telephone number. If calling the first OSS telephone number fails four attempts, the switch starts to call the second OSS telephone number as a backup until calling the first OSS telephone number becomes successful. Both OSS telephone numbers must be administered.

- *Cleared Alarm Notification*

  Enables the MCU to originate a call to the alarm receiver and sends an alarm resolution message whenever all previously reported major and minor alarms are resolved. Alarm origination must be activated for *Cleared Alarm Notification* to work. Values include **y** and **n**.

- *Restart Notification?*

  Generates an automatic call to the OSS number if the MCU has a restart. Values include **y** and **n**

- *Test Remote Access Port*

  Indicates whether to test the remote access port on the Processor circuit pack.

- *CPE Alarm Activation Level*

  Indicates the level at which the customer premises equipment (CPE) alarm is activated. If the level is set to **major**, the CPE alarm is activated only on major alarms. If the level is set to **minor**, the CPE alarm is activated for both major and minor alarms. If the level is **warning**, the CPE alarm is activated for all alarms. If the level is set to **none**, the CPE does not activate for any alarm.

- *Sched Adj Alarm Activation Level*

  SCH-ADJ allows the MCU to originate an alarm to the alarm center whenever CRCS fails to communicate with the MCU for a given time period. This halt in communication may occur if the user powers down the CRCS, or if a problem occurs within the CRCS PC. If the user has a CRCS connected to the MCU, the CRCS logs into the MCU once every hour to display the time (this represents an attempt to synchronize the data and time for the MCU and CRCS). Once the CRCS logs into the MCU, the MCU knows that the CRCS is active. If, within the hour, the CRCS does not log into the MCU, an error is logged against SCH-ADJ. If the CRCS does not log into the MCU for three consecutive hours, an alarm is raised.

  Values for this field include **major**, **minor**, **warning**, and **none** (default). Only major or minor alarms automatically notify the alarm center.

- *Packet Bus Activated?*

  Indicates whether the TN778 PACCON circuit pack is inserted and the packet bus is activated. Values include **y** and **n**.

- *Customer Access to INADS Port*

  Allows the user to prohibit access to system administration and maintenance interface via the INADS port when using customer login ID's. This field can be activated only by Lucent Technologies services via system-parameters maintenance administration. Values include **y** and **n** (default).

Page 1: Scheduled Maintenance Area

- *Start Time*

  Indicates the hour and minute the daily scheduled maintenance actions start. Enter a number from **0** to **23** for hours and **0** through **59** for minutes. The scheduled maintenance actions are: daily maintenance and save translation

- *Stop Time*

  Indicates the hour and minute when scheduled daily maintenance will stop running. If any daily maintenance operations are not completed by this time, the system notes where in the sequence it stopped, and it performs those operations during the next scheduled daily maintenance.

- *Daily Maintenance*

  Displays the status **daily** to indicate the maintenance tests are automatically run each day. This value cannot be changed.

- *Save Translation*

  Indicates if the MCU translation data should be automatically backed up on a specified day of the week (**sunday**, **monday**, etc.) all week (**daily**), or not at all (**no**).

Page 2: **Minimum Thresholds Area**

- *TTRs*

  Shows the minimum number of touch-tone receivers (**1** through **200**) administered in the MCU. If fewer than this number are in service, a warning alarm is raised. Enter value **8** for each TN744 Call Classifier circuit pack in the MCU.

- *CPTRs*

  Shows the minimum number of call progress tone receivers (**1** through **200**) administered in the MCU. If fewer than this number are in service, a warning alarm is raised. Enter value **8** for each TN744 Call Classifier circuit pack in the MCU.

- *Call Classifier Ports*

  Shows the minimum number of call classifiers (**1** through **200**) administered in the MCU. If fewer than this number are in service, a warning alarm is raised. Enter value **8** for each TN744 Call Classifier circuit pack in the MCU.

Page 2: ISDN Maintenance Area

- *ISDN-PRI Test CALL extension*

  Defines the extension number for an ISDN test call. This test allows a far-end ISDN node to test the ISDN-PRI trunks between the far end and the MCU. Enter an unassigned DNIS extension number.

- *ISDN-BRI Service SPID*

  N/A for the MCU. Leave blank.

- *DSO Loop-Around Test Call Extension*

  Indicates the extension used by the network service provider to set up a DSO loop-around connection for testing non-ISDN DS1 trunks.

  This feature is used primarily for testing DSO channels associated with non-ISDN-PRI trunks. The loop-around is activated by dialing the test extension number. Multiple DSO Loop-Around connections can be set up by placing multiple calls to the loop-around extension.

  For more information, see the information about Facility Test Calls in the maintenance document for your system.

Page 3:

- *Modem Connection*

  Indicates the type of modem connection. Values are **internal**, **external**, and **none**. Value cannot be **none** if *Alarm Origination to OSS Numbers* is enabled.

- *Data Bits*

  Enter the number of data bits. Values include **7** and **8**.

- *Parity*

  Display-only field. Value is **none**.

Administering Direct Connect Endpoints

The BRI/DCP Direct Connect feature allows endpoints to connect directly to the MCU via ISDN-BRI and/or Digital Communications Protocol (DCP). A maximum of 12 Direct Connect endpoints can be supported simultaneously on the MCU. This total can include up to 12 BRI Direct Connect endpoints but no more than four DCP Direct Connect endpoints.

To administer Direct Connect endpoints, complete the fields in the forms identified in the following table. Appropriate page references to the forms are provided. Detailed instructions for completing the forms are provided after the table

Table 2-23. Required Forms and Fields for Direct Connect

Form	Field	Page Reference
Customer-Options	■ *Max. BRI Direct Connect Port Capacity* ■ *Max. DCP Direct Connect Port Capacity*	page 2-22
Data Module	■ (See the following “Data Module Form” section.)	page 2-97

Customer-Options Form

Access the Customer-Option form by entering **change system-parameters customer-options**. For BRI connectivity, enter the maximum number of BRI ports that can be allocated in the system at the same time into the *Max. BRI Direct Connect Port Capacity* field (for this release, enter **0** or **36**). For DCP connectivity, enter the maximum number of DCP ports that can be allocated in the system at the same time into the *Max. DCP Direct Connect Port Capacity* field (for this release, enter **0** or **4**). Then submit the form.

Data Module Form (7500 Data Module)

The 7500 data module provides support for ISDN-BRI Direct Connect endpoints via one or more TN556 ISDN-BRI circuit packs. Each ISDN-BRI terminal that is supported by the system has an associated Data Module form administered.

Each TN556 circuit pack supports a maximum of 12 Direct Connect ISDN-BRI endpoints. The MCU can support a maximum of three of these circuit packs and, therefore, a maximum of 36 BRI Direct Connect endpoints. The 7500 data module supports SPID initialization capabilities.

The following table gives the commands that relate to the 7500 data module.

Table 2-24. 7500 Data Module Administration Commands

Action	Object	Qualifier
add	data-module	xxxx (extension or next)
change	data-module	xxxx (extension)
display	data-module	xxxx (extension), print, or schedule
list	data-module	starting extension number, count, print, or schedule
remove	data-module	xxxx (extension)

To add a 7500 data module to provide a Direct Connect interface, follow these procedures:

1. At the *command:* prompt, enter **add data-module *xxxx***.
2. Verify that the screen displays the *Data Module* form (see the following list for a list of the fields and their values).
3. Use TAB or RETURN to advance to the fields you want to change.
4. Submit the form.
5. To verify the form, enter **display data-module *xxxx***.

The following list identifies the fields on the Data Module form and the possible values.

```
add data-module next                                    Page  1 of  2

                                  DATA MODULE

 Data Extension: 1009                Name:  _____________________________
           Type: 7500_____________    COS: 1
           Port: _________            COR: 1









CIRCUIT SWITCHED DATA ATTRIBUTES
  Default Duplex: full     Default Mode: async    Default Speed: 1200


DATA MODULE CAPABILITIES
  Default ITC: restricted                  Default Data Application: M2_A
```

Figure 2-31. Data Module Form (7500), Page 1

Page 1:

- *Data Extension*

  This display-only field provides the extension assigned to the data module.

- *Type*

  Enter value **7500** for BRI Direct Connect endpoints.

- *Port*

  Enter the seven character port location for this endpoint.

- *Name*

  Enter a string containing a maximum of 27 characters to identify the group (for example, **BRI-EPT#1**).

- *COS*

  For the Class of Service, enter or leave the default value of **1**.

- *COR*

  For the Class of Restriction, enter or leave the default value of **1**.

- *Default Duplex*

  N/A for the MCU. Leave at default value **full**.

- *Default Mode*

  N/A for the MCU. Leave at default value **async**.

- *Default Speed*

  N/A for the MCU. Leave at default value **1200**.

- *Default ITC*

  N/A for the MCU. Leave at default value **restricted**.

- *Default Data Application*

  N/A for the MCU. Leave at default value **M2_A**.

```
add data-module next                                    Page  2 of  2

                                  DATA MODULE


BRI LINK/MAINTENANCE PARAMETERS

                   XID? y      Fixed TEI? n
           MIM Support? y     Endpt Init? y   SPID: 1009_____  MIM Mtce/Mgt? y
```

Figure 2-32. Data Module Form (7500), Page 2

Page 2:

- *XID*

  N/A for the MCU. Leave at default value **y**.

- *Fixed TEI*

  N/A for the MCU. Leave at default value **n**.

- *MIM Support*

  (Management Information Message Support). Used to support two types of capabilities: MIM endpoint initialization capability (SPID support) and other maintenance and management capability. Default is **y**.

  For BRI Direct Connect endpoints, set this field to value **n**. Setting this field to value **y** activates the next three fields, which are not needed for endpoint operation.

- *Endpt Init*

  Displays only if *MIM Support* is set to **y**. This field is used to indicate the terminal's endpoint initialization capability. Endpoint initialization is a procedure, required for multipoint operation, by which the 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 an 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 reused terminals must be programmed to match the administered SPID value. Default is **y**.

- *SPID*

  Displays only if the *Endpt Init* field is set to **y**. The Service Profile Identifier (SPID) is a variable parameter containing a maximum of ten digits. If administered, the SPID must be different for all data modules. The SPID should always be assigned. Values include numeric strings in the range of **0** through **9999999999**. Default is the station's extension number.

- *MIM Mtce/Mgt*

  Displays only if *MIM Support* is set to **y** (default). Used to indicate if the terminal supports MIM maintenance and management capabilities other than endpoint initialization.

Supporting the System Printer, CDR, CRCS, and OpCenter

Processor data modules or 7400B Data Modules provide a Data Communications Equipment (DCE) interface for MCU connection to equipment such as the system printer, CRCS, OpCenter, and CDR output devices. These modules also provide a Digital Communications Protocol (DCP) interface to the MCU for a maximum of four Direct Connect DCP endpoints. Each such endpoint uses two ports on a TN754 circuit pack. Use the Data Module form to identify the extension number and related information about each remote adjunct that you administer.

The following commands relate to the data modules.

Table 2-25. Processor Data Module Administration Commands

Action	Object	Qualifier
add	data-module	xxxx (extension or next)
change	data-module	xxxx (extension)
display	data-module	xxxx (extension) print or schedule
list	data-module	starting extension number or count or print or schedule

Use the following procedure to add the processor data module:

1. At the *command:* prompt, type **add data-module xxxx** and then press RETURN.
2. Verify that the screen displays the Data Module screen.
3. From the *Type* field, use TAB and RETURN to advance to the fields you want to change.
4. When the form is completed, verify that the MT displays: *Command successfully completed* followed by *command:*
5. To verify the form, type **display data-module xxxx** and then press RETURN.

Data Module Form (Processor Data Module)

The Data Module form for processor data modules appears as follows:

```
add data-module 2001                                    Page  1 of  1

                                  DATA MODULE

 Data Extension: 2001                Name:                          BCC: 2
           Type: pdm                  COS: 1     Remote Loop-Around Test? n
           Port:                      COR: 1       Secondary data module? n
                                                            Connected to: dte




SPECIAL DIALING OPTION:


ASSIGNED MEMBER ( Station with a data extension button for this data module )

     Ext     Name
  1: ____    _______
```

Figure 2-33. Data Module Form (Processor Data Modules)

The following list identifies the fields on the Processor Interface Data Module form and the possible values.

- *Data Extension*

  This display-only field provides the extension assigned to the data module.

- *BCC*

  This display-only field provides the Bearer Capability Class (BCC) code, which identifies the type of call (including voice call or type of data call). BCC is based on the caller's characteristics for non-ISDN endpoints and on the Bearer Capability and Low-Layer Compatibility Information Elements of an ISDN endpoint. The following list identifies and defines the current BCC values:

  — **0** — voice-grade data and voice

  — **1** — DMI mode 1, 56k data transmission

  — **2** — DMI mode 2, (a)synchronous data transmission up to 19.2k

  — **3** — DMI mode 3, 64k circuit/packet data transmission

  — **4** — DMI mode 0, 64 kbps synchronous data

  Enter value **2**.

- *Type*

  Enter **pdm**.

- *Port*

  Enter the seven character port location on a TN754B circuit pack.

- *Name*

  Enter a string containing a maximum of 27 characters to identify the group (for example, **HSL ext 12345**).

- *COS*

  For the Class of Service, enter or leave the default value of **1**.

- *COR*

  For the Class of Restriction, enter or leave the default value of **1**.

- *Connected to*

  Leave this field set to the default value **dte**. Value **isn** (information network system) is not used on the MCU.

- *Remote Loop-Around Test*

  Enter value **y** if the data module supports a loop-back test at the EIA interface. In general, Lucent Technologies equipment supports this test; however, it is not required by the Level 2 Digital Communications Protocol.

- *Secondary Data Module*

  N/A for the MCU. Leave the default value **n**.

- *COR*

  Enter the desired Class of Restriction. The default is **1**.

- *Maintenance Extension*

  Use the next consecutive extension number that appears in the *Data Extension* field.

- *Ext*

  No entry required. This extension number is assigned automatically.

- *Name*

  Make no entry. This name is automatically assigned.

Administering MCU-Extensions, Meet-Me Numbers, and Connections

An MCU-extension is an extension on the MCU that corresponds to the number that either a conferee dials to join a multipoint video conference or the MCU dials to access conference endpoints. The Lucent Technologies MCU supports three types of MCU-extensions: Meet-Me (Dial-In), Dial-Out, and Audio Add-On/UCC.

A Meet-Me number is a network number that is assigned to the MCU and that corresponds to the MCU's location in the network. A conferee dials a Meet-Me number to access an MCU conference. Meet-Me (Dial-In) calls gain access to the MCU via DS1 trunks or specified data interfaces. Once the network delivers a call across these trunks or interfaces to the MCU, the MCU processes the incoming call digits and routes them to the proper MCU-extension. The MCU-extension associates the call to the proper conference, and the MCU selects the necessary resources from its resource pool to process the call. MCU ports are virtual entities internal to the MCU and are composed of different resources (unlike terminating calls at a voice terminal connected to a PBX).

Dial-Out calls are dialed out by the MCU to the appropriate endpoints. A Dial-Out call can use two different telephone numbers per endpoint.

The number of MCU ports is determined by the software administration of the 1x56k, 1x64k, 2x56k, 2x64k, 112k, 128k, 168k, 192k, 224k, 256k, 280k, 320k, 336k, 384k, 448k, 512k, 672k, 768k, 1008k, 1152k, 1288k, 1344k, 1472k, 1536k, 1680k, 1920k, BONDed 112/128k, 168/192k, 224/256k, 280/320k, 336/384k, 448/512k, and 672/768k, UCC, Continuous Presence *Plus*, and T.120 MCU ports and the hardware resources in the MCU. The MCU is shipped with a configuration of hardware resources appropriate for the model ordered.

As calls arrive at the MCU trunks, the network sends the dialed digits across the MCU access trunk. The Meet-Me number dialed is mapped to an MCU-extension, which in turn enables the MCU to allocate available resources for the call. If the incoming call is from a 2B-channel 56k/64k video endpoint, the MCU actually receives two incoming network calls for the same MCU-extension. On the other hand, if the incoming call is from a BONDed 336 (6 x 56k) video endpoint, the MCU receives six incoming calls for the same MCU-extension. Whatever the number of calls, the calls are joined by the MCU once they are placed, and the MCU assumes that they come from the same video endpoint.

An important function of call routing is the ability to manipulate the digits. This capability provides great flexibility when assigning numbers to users and connecting users to conferences. Digit manipulation can be done through the network, on the PBX (if the MCU is behind a PBX), on the MCU trunk group or in the MCU call routing table. This optional capability allows the MCU to handle any digit string matching the North American dial plan, a private network dial plan, or any multinational dial plan.

The Dedicated Access feature enables endpoints to participate in multipoint audio and video conferences via non-signaled T1 facilities or EIA-530, RS449, or V.35 data interfaces via the DI circuit pack. For this feature, the MCU can connect to a multiplexer (MUX), a Digital Access Crosspoint System (DACS), or to H.320 endpoints via DS1 facilities. A maximum of 20 DS1 connections to the MCU are supported. Also, a maximum of 32 data interface (DI) connections to the MCU are supported.

The MCU can also connect to an H.320, H.CTX, or SG4 codec via an EIA-530, RS449, or V.35 data interface by using RS366 Dialing. These data interfaces allow endpoints either connected directly to the MCU or behind an encryption device or proprietary inverse multiplexor to participate in multipoint conferences at a rate in the range 112k to 1920k. RS366 Dialing can also use any of these data interfaces to enable H.323 endpoints connected behind H.323 gateways to participate in MCU conferences.

An Nx56k MCU-extension can be assigned to an Nx64k conference via Speed Match. Also, an MCU-extension greater than 56k can be assigned to a 56k conference. If a video endpoint with a bandwidth greater than 56k dials into a 56k conference, the MCU will notify the endpoint to switch to 56k bandwidth and will make it an audio-only source. If the endpoint responds that it has switched to 56k bandwidth, it will become a video source.

Finally, Low-Speed/Higher-Speed Interworking allows 56k and 64k endpoints to join a conference of a higher bandwidth as Audio Add-On endpoints.

Network Services

The MCU supports only digital trunks for the incoming and outgoing services used by MCU-extensions; therefore, only ISDN-PRI, robbed-bit DS1, or ISDN-BRI Trunk Interface facilities may be used to connect the MCU to the network.

ISDN-PRI facilities support 56k or 64k calls to and from the MCU. The MCU ISDN-PRI interface can support multiple DS1 facilities as well as non-facility associated signaling (NFAS) and D-channel backup. Robbed-bit DS1 facilities support only 56k calls to and from the MCU. ISDN-BRI Trunk Interface provides ISDN-BRI connections to the public network.

The MCU is viewed as an endpoint by the public network. If the MCU resides behind a PBX, the PBX serves as the public network endpoint. Endpoint users must subscribe to a network service that provides a digital transmission service.

The following table provides an overview of the most commonly used digital services.

Table 2-26. Network Services

Network Service	Number Dialed to Join Conference	Bandwidth	Comments
Switched Digital International (SDI)	700-56x-xxxx 700-737-xxxx 700-730-xxxx	56k 1B*, 2B^, 1-channel multirate+, BONDed# 1B*, 2B^, 1-channel multirate+, BONDed#	RBS network access ISDN-PRI% ISDN-PRI%
ACCUNET Switched Digital Service (SDS) SW56	700-56x-xxxx	56k	RBS network access
ACCUNET SDS SW56/64	700-737-xxxx	1B*, 2B^, 1-channel multirate+, BONDed#	ISDN-PRI%
Software Defined Data Network (SDDN)	NNN-NNNN Nxx-xxxx N0/1x-Nxx-xxxx	56k 1B*, 2B^, 1-channel multirate+, BONDed# 1B*, 2B^, 1-channel multirate+, BONDed#	RBS network access ISDN-PRI% ISDN-PRI%
SDDN-Virtual Off-Net	700-56x-xxxx 700-737-xxxx	56k 1B*, 2B^, 1-channel multirate+, BONDed#	RBS network access ISDN-PRI%
Private Network (PBX)	1 to 7 digits		Based on PBX dial plan connecting to the MCU

x = any number from 0 through 9
N = any number from 2 through 9
* 56k, 64k
^112k, 128k
+128k, 192k, 256k, 280k, 320k, 336k, 384k, 448k, 512k, 672k, 768k, 1472k, 1536k, or 1920k
112/128k, 168/192k, 224/256k, 280/320k, 336/384k, 448/512k, or 672/768k
% NonBONDed

Administering the Dial Plan

Through administration of the dial plan, the MCU identifies calls from the network and routes them to available video ports. This is done through the Dial Plan Record form.

The commands in the following table can be used to access the Dial Plan Record form.

Table 2-27. Dial Plan Administration Commands

Action	Object	Qualifier
change	dialplan	
display	dialplan	print or schedule

To change the dial plan, follow these procedures:

1. At the *command:* prompt, enter **change dialplan**.
2. Verify that the screen displays the *Dial Plan Record* form (see “Dial Plan Form” for a list of fields and values).
3. Use TAB or RETURN to advance to the fields you want to change.
4. Submit the form.
5. To verify the form, enter **display dialplan**.

Dial Plan Form

Page 1 of the Dial Plan form appears as follows:

```
change dialplan                                          Page  1 of  1


                                DIAL PLAN RECORD

   North American Area Code: 732                    Local Node Number: 1_
      ARS Prefix 1 Required? y
       Uniform Dialing Plan? 4-digit
 UDP Extension Search Order: local-extensions-first

FIRST DIGIT TABLE
First                              Length
Digit   -1-        -2-        -3-        -4-        -5-        -6-
  1: _________  _________  dac______  _________  _________  _________
  2: fac______  _________  _________  _________  _________  _________
  3: _________  _________  _________  _________  _________  _________
  4: _________  _________  _________  extension  _________  _________
  5: _________  _________  _________  extension  _________  _________
  6: _________  _________  _________  extension  extension  _________
  7: _________  _________  _________  extension  _________  _________
  8: _________  _________  _________  extension  _________  _________
  9: fac______  _________  _________  _________  _________  _________
  0: fac______  _________  _________  _________  _________  _________
  *: _________  _________  fac______  _________  _________  _________
  #: _________  _________  fac______  _________  _________  _________
```

Figure 2-34. Dial Plan Form (Page 1)

The following list shows the fields on the Dial Plan Record and related values.

Dial Plan Form, Page 1 of 1

- *North American Area Code*

  Required only in North America. Enter the area code number where the MCU is located. Value is any area code in the form ***NXX***, where ***N*** is any digit 2 through 9, and ***X*** is any digit 0 through 9. Value must contain only digits (for example, **215**). Leave the field blank if the MCU is not in North America.

- *Local Node Number*

  N/A for the MCU. Leave blank.

- *ARS Prefix 1 Required*

  Displayed only if *North American Area Code* is not blank. Enter **y** if prefix “1” is required to call an area code, or if the office codes in the Home Numbering Plan Area (HNPA) are interchangeable with Numbering Plan Areas (NPAs). Prefix “1” is usually required within those areas where local central office (CO) codes resemble area codes (that is, the middle digit is a “0” or a “1”). For example, “201” in New Jersey or “212” in New York.

- *Uniform Dialing Plan*

  Enter **4-digit** or **5-digit** to signify the use of a 4- or 5-digit dial plan, or none (default) if no Uniform Dialing Plan (UDP) is to be administered. Default is **none**. If **4-digit** or **5-digit** is entered, the *UDP Extension Search Order* field is displayed.

  NOTE:
  The Uniform Dialing Plan field must be enabled on the Customer-Options form before any entry other than none can be entered in this field.

  The Uniform Dialing Plan is a separate form that must be administered if **4-digit** or **5-digit** is entered into this field.

  CAUTION:
  If the entry in the Uniform Dialing Plan field is changed from its previously administered value, all UDP extension codes will be lost.

  The UDP provides a common 4- or 5-digit dial plan that can be shared among a group of switches. Interswitch dialing and intraswitch dialing both require 4- or 5-digit dialing. The UDP is used with Electronic Tandem Network (ETN), Main/Satellite/Tributary, and DCS configurations. Additionally, UDP can be used alone to provide uniform 4- or 5-digit dialing between two or more private switching systems without ETN, DCS, or Main/Satellite/Tributary configurations.

- *UDP Extension Search Order*

  Displays whenever *Uniform Dialing Plan* is set to **4-digit** or **5-digit**. Indicates where the system should look for a dialed extension first. Values include **local-extensions-first** (default) and **udp-table-first**. For the MCU, leave the default value **local-extensions-first**.

  For value **local-extensions-first**, the dialed extension is searched for in local translations first. If the extension is found, the call terminates to the local station. If the extension is not found locally, the UDP tables are searched for an off-switch (UDP) conversion. For value **udp-table-first**, the UDP tables are searched first.

- *First Digit Table*

  This table contains fields that establish each possible first digit and dialed number length for various extensions and codes. Values include **attd**, **pextension**, **extension**, **misc**, **fac**, and **dac**. Administration for values **extension**, **fac**, and **dac** is required on this form before you can administer the following: MCU-extensions, data extensions, hunt groups, access endpoints, trunk groups, and feature access codes. The following list discusses each value for this field, and it indicates where each one can appear in the form.

— **pextension** — should not be administered on the MCU.

— **extension** — used for administering MCU-extensions, data-modules, hunt groups, etc. The extension can have a first digit ranging from 0 through 9, and it can have a length of one to five digits. The rows in the form headed by *0*, *, and # row cannot contain this value. For example, if MCU-extensions 400 through 499 are required, enter value **extension** at the intersection of the fourth row and third column. Another example: if MCU-extensions 6000 through 6999 are required, enter value **extension** at the intersection of the sixth row and fourth column.

NOTE:
The extension cannot have the same first digit as the ARS or AAR facility access code (FAC).

— **fac** (feature access code) - Required before administering the ARS access code on the Feature Access Codes form discussed later in this chapter. The FAC can have a first digit ranging from 0 through 9 plus the * (star) and # (pound) symbols. A FAC can have a length of one to four digits in rows 0 through 9, and it can have a length of one to four digits in the rows headed by * and #. For FACs using the rows headed by * and #, * and # each count as a digit and must be the first digit. For example, *9 could be used to activate a feature when the FAC is administered as two digits in the row headed by *.

— **dac** (trunk access code) - Required before administering trunk groups (discussed later in this chapter). A TAC can have a first digit ranging from 0 through 9, and it can have a length of one to four digits.

— **attd** (attendant) - Defines how users call an attendant. An attendant access numbers can start with any number from 0 through 9, and it contain one or more digits.

NOTE:
An attendant group number can also be defined as a two-digit number. The first digit is defined as "misc" on the First Digit Table, and the second digit (0 through 9) is defined on the Second Digit Table screen.

— **misc** (miscellaneous code) - This code is used if more than one type of code should start with the same digit and be the same length. The dialed length can be one digit, and the first digit can be any digit from 0 through 9. Using a misc code requires that you also define a second digit table.

Second Digit Table

You must complete the Second Digit Table each time you enter value **misc** into the first column (representing digit length of “1”) of the “First Digit Table” in the Dial Plan form. The Second Digit Table is named for the row where **misc** appears. An example of the table appears as follows:

```
add second-digit 5                                          Page  1 of  1

                          SECOND DIGIT TABLE FOR DIGIT 5

SECOND DIGIT TABLE

Digit   Identification     Number of     Digit   Identification     Number of
                            Digits                                  Digits

    0: ___________             0             5: ___________             0
    1: ___________             0             6: ___________             0
    2: ___________             0             7: ___________             0
    3: ___________             0             8: ___________             0
    4: ___________             0             9: ___________             0


attd          dac              extension      fac
```

Figure 2-35. Second Digit Table

A Second Digit Table can exist for every first digit value.

Value **misc** can have a dialed length of 1, and it can have a first digit ranging from 0 through 9. Value **misc** specifies that the call types for this digit are actually determined in the Second Digit Table.

Administering the Uniform Dialing Plan

Use the Uniform Dialing Plan form to administer UDP.

Uniform Dialing Plan Form

Use the following commands to administer the Uniform Dial Plan form.

Table 2-28. Uniform Dialing Plan Administration Commands

Action	Object	Qualifier
change	udp	Enter 1-MAX digits between 0-9, ‘x’ or ‘X’
display	udp	Enter 1-MAX digits between 0-9, ‘x’ or ‘X’ [’print’ or ‘schedule’]
list	udp	Enter [’ext’ x][’to-ext’ x][’type’ x [[’dest’ x][to-dest’ x]]] [’print’ or ’schedule’]

Note: Brackets [ ] indicate the qualifier is optional. Single quotes (‘ ’) indicate the text inside the quote must be entered exactly as shown or an abbreviated form of the word may be entered. MAX is the maximum number available in your system configuration.

An example of the Uniform Dialing Plan form appears as follows:

```
change udp 2                                              Page  1 of  2


                              UNIFORM DIALING PLAN
                                 Ext Codes: 2ddx


                              Ext Code: 2xxx  Type:

dd    Type      dd    Type      dd    Type      dd    Type      dd    Type

0x:             1x:             2x:             3x:             4x:

00:             10:             20:             30:             40:
01:             11:             21:             31:             41:
02:             12:             22:             32:             42:
03:             13:             23:             33:             43:
04:             14:             24:             34:             44:
05:             15:             25:             35:             45:
06:             16:             26:             36:             46:
07:             17:             27:             37:             47:
08:             18:             28:             38:             48:
09:             19:             29:             39:             49:
```

Figure 2-36. Uniform Dialing Plan Form (Page 1)

```
change udp 2                                                Page  2 of  2


                              UNIFORM DIALING PLAN
                                 Ext Codes: 2ddx


                              Ext Code: 2xxx  Type:

dd    Type      dd    Type      dd    Type      dd    Type      dd    Type

5x:             6x:             7x:             8x:             9x:

50:             60:             70:             80:             90:
51:             61:             71:             81:             91:
52:             62:             72:             82:             92:
53:             63:             73:             83:             93:
54:             64:             74:             84:             94:
55:             65:             75:             85:             95:
56:             66:             76:             86:             96:
57:             67:             77:             87:             97:
58:             68:             78:             88:             98:
59:             69:             79:             89:             99:
```

Figure 2-37. Uniform Dialing Plan Form (Page 2)

Make assignments as required for the following fields on each page of the form.

- *Ext Codes*

  Displays a string of digits, "x" and "dd." The position of the “dd” shows which digit positions of the Ext Codes are being administered. For example, value **512dd** means that the last two digits of the Ext Codes, for the leading digits 512, are being administered on the form. The string “5ddxx” shows that the second and third digits of the Ext Codes, for the leading digit 5, are being administered on the form (these entries would all be wildcard entries).

- *Ext Code, Type* (and associated data)

  Displays what rules apply if the values on the form are not administered. For example, shows a form for Ext Codes starting with 512. If these entries are not administered, the entry for Ext Code 51xxx applies to these entries, and if Ext Code 51xxx is not administered, the entry for Ext Code 5xxxx applies to these entries. In a 5-digit UDP, these read-only fields display only if at least three digits are specified in the command-line key, although one set (the set on the left) of these fields would still display if two digits are specified in the command-line key. If only one digit is specified, or if ’x’ or ’X’ is specified in the command-line key, then neither of these sets of fields display, since there is no Ext Code more general than those listed on the form.

- *dd*

  Displays the two digits (or x's) representing the extension or group of extensions to be administered. For this field, "xx:" is repeated on both pages since it is the default for all other entries on both pages. Changing the type associated with this field on one page changes it on both pages.

- *Type*

  Specify the type of treatment for the extension (**AARCode**, **ENPNode**, **Local**, **TempOOS**, **UDPCode**, or blank). That is, specify whether the extension should receive intercept treatment, be converted to AAR, or be converted via ENP. The difference between AAR Location Codes and UDP Location Codes is that 7-digit AAR Numbers created from AAR Location Codes may be changed during AAR analysis by AAR digit conversion before being routed. Seven-digit AAR Numbers created from UDP Location Codes are not subject to digit conversion before being routed with AAR digit analysis. When an extension code that is displayed on the UDP form is not administrable because it conflicts with the translations in the first and/or second digit tables, then the type field for that extension code displays "NotInDP."

  The entry TempOOS indicates an extension is temporarily out-of-service. Calls to these extensions receive reorder. A blank entry indicates this Ext code is not administered. If no matching code can be found, then the call receives intercept treatment.

- *Location Code*

  Specify the 3-digit location code for Ext Codes administered with a type of AARCode or UDPCode. This field displays if the *Type* field is set to **AARCode** or **UDPCode**.

- *Node Number*

  Specify the node number for Ext Codes administered with a type of ENPNode. This field appears if the *Type* field is set to **ENPNode**.

  NOTE:
  This node number must not be the same as the local node number on the Dial Plan form.

Assigning MCU-Extensions and Meet-Me Numbers

The MCU is given a network number that corresponds to its location in the network. This number is the Meet-Me number that conferees dial to join a multipoint conference. Network numbers can be obtained from Network Services or, if the MCU is an SDDN or private network location, the Meet-Me numbers can be selected and assigned according to the existing dial plan. After the Meet-Me numbers are obtained, each number needs to be associated with an MCU-extension on the MCU.

NOTE:
Whenever possible, the last four digits of the MCU-extension should be the same as the last four digits of the corresponding Meet-Me number. This helps with troubleshooting so you can quickly relate a dialed Meet-Me number with the matching MCU-extension on the MCU. It is important to note, however, that an MCU-extension cannot begin with a **0**.

The following table provides the commands that can be used to administer MCU-extensions.

Table 2-29. MCU-Extension Administration Commands

Action	Object	Qualifier
add	mcu-extension	extension or next
change	mcu-extension	extension
display	mcu-extension	extension, count, print, or schedule
duplicate	mcu-extension	extension
list	mcu-extension	extension, count, print, or schedule
remove	mcu-extension	extension

NOTE:
An MCU-extension cannot be removed if it is assigned to a conference reserved on the MCU.

To add an MCU-extension, follow these procedures:

1. At the *command:* prompt, enter **add mcu *xxxx*** where ***xxxx*** is an extension number or **next**.
2. Verify that the screen displays the *MCU-Extension* form (see "MCU-Extension Form" for a list of fields and values).
3. Use TAB or RETURN to advance to the fields you want to change.
4. Submit the form.
5. To verify the form, enter **display mcu *xxxx***.

   NOTE:
   Be sure to make the appropriate changes to the MCU-Extensions Form described earlier in "Recording Meet-Me Numbers and MCU-Extensions."

Considerations For Meet-Me Dial-In Numbers

You should obtain network (Meet-Me) numbers from the network provider. After you obtain these numbers, you need to associate each number with an MCU-extension on the MCU. Whenever possible, the last four digits of the MCU-extension should be the same as the last four digits of the corresponding Meet-Me number. This helps with troubleshooting in that you can quickly relate a dialed Meet-Me number with the matching MCU-extension on the MCU.

NOTE:
An MCU-extension cannot begin with “0.”

The MCU customer can assign either of two types of Meet-Me Numbers, as follows:

- **Per-Port Meet-Me Numbers**

  In this case, the customer assigns as many Meet-Me Numbers as there are bridge ports. For example, an MCU with a 2B capacity of 24 ports would require 24 Meet-Me Numbers from the provider for video calls.

- **Per-DS0 Meet-Me Numbers**

  In this case, the customer assigns a Meet-Me Number to each DS0. For example, an MCU with two ISDN-PRI DS1 spans would need 46 Meet-Me Numbers from the provider for video calls.

Whoever is ordering service for the MCU should request from the provider the required number of incoming numbers that they would like to administer on the MCU.

Considerations For Dial-Out Numbers

Dial-Out MCU-extensions are used to originate outgoing video and audio calls. Dial-out video calls originate in three different ways, as follows:

- For all speeds of multirate calls, the MCU dials only one number.
- For 2B calls, the MCU makes two calls; it calls either one number twice or two separate numbers, depending on the far-end video numbers.
- For BONDed calls, the MCU makes a separate call for each DS0 required for the bandwidth of the call. The MCU is programmed to dial one number for a BONDed call and then let the far end answer and tell the MCU regarding what other numbers need to be dialed to complete the BONDed connection.

You should administer at least one Dial-Out MCU-extension for each Meet-Me video MCU-extension that is administered on the MCU.

NOTE:
Audio Add-On Dial-Out calls use the same pool of extensions that is used for incoming Audio Add-On/UCC calls.

Considerations For Audio Add-On/UCC MCU-Extensions

Audio Add-On/UCC MCU-extensions can be used for both Audio Add-On and UCC endpoints. An incoming Audio Add-On or UCC party joins a multimedia conference in the same manner as a Dial-In Meet-Me endpoint. The Audio Add-On party is identified on the Conference Record form, and it is assigned a Meet-Me number that has been reserved for Audio Add-On use. The Audio Add-On participant in an MCU conference can also be part of an audio conference (for example, ALLIANCE, PBX, etc.). UCC endpoints are restricted to incoming only.

Be sure to order from the network provider, and then assign, one MCU-extension for each Audio Add-On and UCC port based on the corresponding port capacity that is administered in the Customer-Options form. By using information received from the network provider, the customer can usually specify the format of the called party numbers for audio calls that are delivered to the MCU. Typically, the network delivers the last four digits of the dialed number for both ISDN-PRI and DS1 robbed-bit service.

MCU-Extension Form

The MCU-Extension form appears as follows:

```
add mcu-extension 75000

                    MCU EXTENSION

                     Extension: 75000
                 Meet-Me Number: _________________
               Audio Add-On/UCC? n
          Bandwidth per Channel: Nx56k
```

Figure 2-38. MCU-Extension Form

The Lucent Technologies MCU supports three types of MCU-extensions: Meet-Me (Dial-In), Dial-Out, and Audio Add-On/UCC. The following list shows the fields on the MCU-Extension form and their related values.

- *Extension*

  A display-only field that shows the number assigned to the MCU-extension.

  NOTE:
  If this is a Meet-Me Audio Add-On/UCC or video number, the extension number being administered should match the digits being received from the service provider, if possible. This would eliminate digit manipulation on the MCU and make it easier to administer the system.

- *Meet-Me Number*

  Enter the network number *only* if this is a Meet-Me number or an Audio Add-On/UCC number. For video dial-out mcu-extensions, leave this field blank. This number should reflect the number dialed at the far end to reach this extension.

  NOTE:
  CRCS uses this field when the MCU-extension utility feature is used.

- *Audio Add-On/UCC*

  Enter **yes** or **no**, as appropriate**.** The default is **no**.

- *Bandwidth per Channel*

  If the Meet-Me Number supports the 56k- and 64k-interval bandwidths, enter value **all**. If the Meet-Me Number supports only the 56k-interval bandwidths, leave the default value **Nx56k**.

  NOTE:
  This field appears only if the *Audio Add-On/UCC* field is set to **no**.

Duplicate MCU-Extension Form

This form is used to administer MCU-extensions in groups of a maximum of 32 at a time, as opposed to administering one such extension per form. Meet-Me, Dial-Out and Audio Add-On MCU-extensions can be “duplicated.” Adding MCU-extensions via this form requires that all the numbers be administered with the same parameters (for example, all the extensions must be dial-out extensions).

To “duplicate” an extension, enter the following command:

duplicate mcu-extension *extension no.* start *[extension no.]* count *[X]*,

where:

- First ***extension no.*** is the MCU-extension to be duplicated.
- Second ***extension no.*** is the first generated MCU-extension desired.
- ***X*** is the count (***1*** through ***32***) of generated MCU-extensions desired.

For example:

duplicate mcu-extension *6835* start *6901* count *32*

The following list identifies and describes the fields in the Duplicate MCU-Extension form.

- *Ext*

  This display-only field provides the MCU-extensions being assigned per the count requested by the appropriate command entry.

- *Meet-Me-Number*

  This field identifies the number that must be dialed to reach the MCU-extension. This field can be left blank for dial-out MCU-extensions.

- *Audio Add-on/UCC?*

  This field allows the administrator to add both video and Audio Add-On MCU-extensions. Values include **y** and **n**. For value **y**, the *Bandwidth per Channel* field does not appear.

- *Bandwidth per Channel*

  This field allows the administrator to set the video capabilities to **56kbps** for 56k-interval video calls or to **all** for both 56k-interval or 64k-interval video calls. For value **all**, the network that is connected must be an ISDN-PRI network. For an RBS network, the MCU-extensions associated with that service must be set to value **Nx56k**.

  NOTE:
  This field appears only if the *Audio Add-on/UCC* field is set to **n**.

Administering Trunk Groups

The DS1 T1/E1 service provides digital trunks that support voice, video and data communications. Trunks are communication links that connect two switching systems. Trunks are grouped together via trunk group administration. Trunk group administration allows the system administrator to assign service characteristics to a group of trunks rather than to each trunk individually. Changes can be made and characteristics can be removed on a group basis. Also, changes can be made without taking trunks out of service.

Trunks and access endpoints consume the same resource. Therefore, the sum of the trunks and access endpoints cannot exceed the total number of trunks allowed for a particular MCU configuration.

The Lucent Technologies MCU supports Integrated Services Digital Network-Primary Rate Interface (ISDN-PRI) trunk groups and tie trunk groups (robbed-bit). Understanding the variety of ways in which trunks differ helps simplify decisions when implementing and administering a network.

ISDN-PRI trunk groups deliver ISDN service to the MCU for high-speed connectivity to the public switched telephone network and to other switches in a private or public network. PRI provides 24 64k channels arranged in the North American ISDN standard of 23B+D. This indicates that 24 channels are divided into 23 bearer B-channels at 64k for information transmission and one signaling D-channel at 64k for control and signaling. The MCU also supports the NFAS feature, which allows all 24 B-channels to be used along with the D-channel from another ISDN-PRI span for control and signaling. Outside of the US (except for MCU-to-MCU/Definity ECS connections), the MCU also supports ISDN-PRI using the international E1 format, which provides the 30B+D standard.

Call-by-Call Service Selection enables a single ISDN-PRI trunk group to carry calls to a variety of services. This feature does not require that each trunk group be dedicated to a specific service; rather, it allows you to set up various voice and data services and features for a particular call.

Tie trunks (robbed-bit) carry communications between switches in a private network. The type of tie trunk installed depends upon the type of network established. Robbed-bit network facilities provide a DS1 T1 interface between the MCU and network provider. The DS1 interface supports tie trunks for incoming and/or outgoing calls. Robbed-bit DS1 trunk groups support all 24 DS0 channels for video, voice and data only at a speed of 56k per DS0.

Digital DS1 trunk administration requires the administration of a number of forms depending on the type of service that is used with the MCU. ISDN-PRI can use T1 and/or E1 service, while robbed-bit facilities supports only T1 service.

NOTE:
For both ISDN-PRI and robbed-bit service (RBS), the administrator needs to know what options the far end is using to administer the MCU. Coordination between the MCU administrator and the network provider is essential to ensure that the administration between both ends of the span is compatible.

Use the following procedures to administer ISDN-PRI and tie trunks for the MCU. The following table shows the commands that are used to administer trunk groups using the Trunk Group form.

Table 2-30. ISDN-PRI Trunk Group Commands

Action	Object	Qualifier
add	trunk-group	1-99 (next)
change	trunk-group	1-99 or TAC X
display	trunk-group	1-99 or TAC X, print, or schedule
list	trunk-group	print or schedule
remove	trunk-group	1-99 or TAC X
status	trunk-group	1-99 or TAC X

To add an ISDN or tie trunk group, follow these procedures:

1. At the *command:* prompt, enter **add trunk group *x*** where ***x*** is a trunk group number (**1** through **99**) or **next**.
2. Verify that the screen displays the Trunk Group form (see “Administering ISDN-PRI Trunk Groups” or “Administering Tie Trunk Groups” sections for a list of fields and values).
3. Use TAB or RETURN to advance to the fields you want to change.
4. Submit the form.
5. To verify the form, enter **display trunk-group *x***.

Administering ISDN-PRI Trunk Groups

The following list identifies the fields on the Trunk Group form and their values entered when administering ISDN-PRI trunk groups:

```
add trunk-group 11                                          Page  1 of 10

                                  TRUNK GROUP
     Group Number: 11               Group Type: isdn        CDR Reports? y
       Group Name: OUTSIDE CALL            COR: 1                  TAC:
        Direction: two-way    Outgoing Display? n
      Dial Access? n
     Service Type:                                          TestCall ITC: rest
                          Far End Test Line No:
TestCall BCC: 4
TRUNK PARAMETERS
         Codeset to Send Display: 6      Codeset to Send TCM,Lookahead: 6
        Max Message Size to Send: 260    Charge Advice: none
  Supplementary Service Protocol: a      Digit Handling (in/out): enbloc/enbloc

            Trunk Hunt: cyclical
     Connected to Toll? n                            DTT to DCO Loss: normal
Calling Number - Delete:    Insert:                  Numbering Format:
                                  Synchronization: async       Duplex: full
 Disconnect Supervision - In? y  Out? n
 Answer Supervision Timeout: 0
```

Figure 2-39. Trunk Group Form (Page 1)

Trunk Group Form, Page 1 of 10

- *Group Number*

  Display-only field that shows the number of the trunk group that is currently being accessed.

- *Group Type*

  Enter **isdn** to identify the trunk group.

- *CDR Reports*

  Enter **y** if tracking calls on this trunk group. Otherwise, leave value **n**.

- *Group Name*

  Enter a name of up to 27 characters to identify this trunk group (for example, **isdn to AT&T**, **circuit # of span**, etc). The default is **OUTSIDE CALL.**

- *COR*

  Leave at the default of **1**.

- *TAC*

  Enter the trunk access code that matches the numbering plan assigned on the dial plan for the TAC. CDR uses the TAC to identify the trunk group on CDR reports. Allowable entries must match the dial plan.

- *Direction*

  Values include **incoming**, **outgoing**, and **two-way** (default), depending on how the associated trunk group is being used. (For example, if the far end is administered for outgoing calls on the trunk group, the field should be set to **incoming** for the MCU trunk group.)

- *Outgoing Display*

  N/A on the MCU. Leave at default value **n**.

- *Dial Access*

  Enter **y** to allow this trunk group to be accessed via a TAC; otherwise, enter **n**.

- *Service Type*

  Indicates the service for which this trunk group will be dedicated. Values include those in the following table and any user-defined or selected service as administered on the Network Facilities form. Value **cbc** is recommended to allow for incoming and outgoing voice and video (data).

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-BRI trunk group.
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.
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).
public-ntwk	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 switch). 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.
tandem	Tandem tie trunks integral to an ETN
tie	Tie trunks — general purpose

- *Test Call ITC*

  Controls the encoding of the ITC codepoint for the Bearer Capability IE in the SETUP message. Since the ISDN Testcall does not go through routing, it is never blocked due to an incompatible ITC. Values include **rest** for restricted (56k) and **unre** for unrestricted (64k).

- *Usage Alloc?*

  Displayed whenever the *Service Type* field is set to **cbc**. Leave this field set to **n**.

- *Far-End Test Line No:*

  Specify the number that is sent to the far-end ISDN test line extension. When test trunk is entered, this 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 here must be what the far end expects.

  This number should be supplied by the network provider for the associated trunk group. The network provider may be charged for using this feature.

- *TestCall BCC*

  Controls the encoding of the Bearer Capability IE (BCIE) in the SETUP message. Values include the following:

  — **0** — voice

  — **1** — DCP Mode 1 56k

  — **2** — Mode 2 async (19.2)

  — **3** — Mode 3 circuit (64k)

  — **4** — DCP Mode 0 (64k)

- *TestCall Service*

  Specifies the call-by-call selection for an ISDN-PRI test call. This field appears only if the Service Type field is set to **cbc**. Values are provided in the following table. "Sddn" or any new Facility Type of Type 0 (feature), Type 1 (service), or Type 3 (outgoing) that is defined by the user on the Network Specific Facility Encoding format are excluded.

accunet	ACCUNET Switched Digital Service — part of ACI (AT&T Communications ISDN) phase 2.
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.
ids	Long-Distance Service — part of ACI (AT&T Communications ISDN) phase 2.
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.
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.
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's location to defined service areas referred to as bands; the widest band is 5.
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 — presubscribed common carrier operator.
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.

- *Codeset to Send Display*

  Defines the codeset for sending the information element for the display. For the MCU, enter value **?**.

- *Codeset to Send TCM, Lookahead*

  Defines the codeset for sending the information element for Traveling Class Marks and Lookahead Interflow. For the MCU, enter value **?**.

  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* is set to **tandem**. It is then used by the distant tandem switch to permit access to facilities consistent with the originating user's privileges.

- *Max Message Size to Send*

  Defines the maximum byte size of the ISDN-PRI messages that the MCU can receive. Values include **260** (default), **256**, **244**, and **128**. Leave at the default value **260**.

- *Charge Advice*

  Allows you to accumulate and access charge information about a call. You must have already set *CDR Report* to "y"or "r" (ring-intvl) before changing this field from its default of **none**. You may specify one of the following options:

  — **automatic** — Advice of Charge is enabled automatically for all outgoing calls over this trunk group.

  — **during-on-request** — MCU will request information during and at the end of the call for each outgoing call over this trunk group.

  — **end-on-request** — MCU will request information at the end of the call for each outgoing call over this trunk group.

  — **none** — No Advice of Charge information is collected by the system.

  For value **during-on-request**, the system performance is affected because this value increases message activity on the signaling channel, and this may reduce the maximum call capacity.

- *Supplementary Service Protocol*

  Indicates which supplementary service protocol should be used for services over this trunk group. *Supplementary Service protocols are mutually exclusive*.

  Enter one of the following Supplementary Service Protocols. Default is **a**.

  — **a** — national public network

  — **b** — ISO/ETSI QSIG Private Network

— **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

NOTE:
Do not use this value if *Service Type* is set to **dmi-mos** or **sddn**.

- *Digit Handling*

  This field defines whether the overlap receiving and overlap sending features are enabled. Set the field to **overlap** when you want overlap receiving or overlap sending. Set the field to **enbloc** when you do not want these features enabled. The first field value indicates digit receiving and the second value indicates digit sending. There are 4 possible combinations: **enbloc/enbloc** (default), **enbloc/overlap**, **overlap/enbloc**, and **overlap/overlap**.

  Without overlap receiving or overlap sending enabled, the digits on incoming and outgoing calls are sent enbloc. If *Digit Handling* is set to **overlap/enbloc** or **overlap/overlap**, the following results:

  — Incoming Call Handling Treatment table does not appear.

  — *Digit Treatment* and *Digits* appear.

  — Warning message indicates that all Incoming Call Handling entries are removed when form is submitted.

  — When form is submitted with these values, all Incoming Call Handling entries are removed.

- *Trunk Hunt*

  Enter the order in which the MCU should find available trunks. Values are **descend**, **ascend,** and **cyclical**. The value should be the opposite from that for the other end of the trunk group.

- *Connected to Toll*

  N/A for the MCU. Leave at default value **n**.

- *DTT to DCO Loss* (Audio Only)

  Specifies the loss plan to be used for this trunk group. Leave this field set at the default value **normal**.

- *Calling Number - Delete: Insert: Numbering Format*:

  These three fields comprise the administrable fields for the Calling Line Identification Prefix feature. They display when *Direction* is set to **incoming** or **two-way**.

— *Delete* — Number of digits to delete from the calling party number for all incoming calls on this trunk group. Values include any number from **1** through **15**, **all** or blank.

— *Insert* — Specific digits prefixed to incoming calls when the calling party is a member of this trunk group. Values include any string of digits containing one through 15 digits or blank.

— *Numbering Format* — TON/NPI encoding applied to CPN information modified by the CLI Prefix feature. This encoding does not apply to calls originating locally. *Numbering Format* applies to calls originated from this switch. If this field is blank, the DEFINITY ECS passes on the encoding received in the incoming setup message. If the incoming setup message did not contain CPN information, and if digits are added, the outgoing message contains these digits. If *Numbering Format* is blank in this case, the value defaults to **pub-unk**. If *Numbering Format* 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 in the following table.

Numbering Format Valid Values	Type of Numbering (TON)	Numbering Plan Identifier (NPI)
blank	incoming TON unmodified	incoming NPI unmodified
intl-pub	international(1)	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)
locl-put	local/subscriber(4)	E.164(1)
natl-pub	national(2)	E.164(1)
pub-unk	unknown(0)	E.164(1)
unk-unk	unknown(0)	unknown(0)

- *Numbering Format*

  This field appears if either *Send Calling Number* or *Send Connected Number* is set to **y** or **r**. This specifies the encoding of the "Numbering Plan Indicator" field for identification purposes in the Calling Number and/or Connected Number IEs and in the QSIG Party Number. Values include **public** (default), **unknown**, and **private**. Value **public** indicates that the number plan according to CCITT Recommendation E.164 is used and that *Type of Number* is set to **national**. Value **unknown** indicates that *Numbering Plan Indicator* is set to **unknown** and that *Type of Number* is set to **unknown**. Value **private** indicates that *Numbering Plan Indicator* is set to **PNP** and that the value for *Type of Number* is determined from the ISDN Private-Numbering form.

- *Synchronization*

  N/A for the MCU. Leave at default value **async**.

- *Duplex*

  N/A for the MCU. Leave at default value **full**.

- *Disconnect Supervision...In?*

  Displays when *Direction* is set to **incoming** or **two-way**. Values include **y** (default) and **n**. Internally, the field is set to **n** if *Direction* is set to **outgoing**. For the MCU, enter **y** to indicate that the far end can release a trunk when the calling party releases from a call and the system is able to recognize the release signal.

- *Disconnect Supervision...Out?*

  Displays when *Direction* is set to **outgoing** or **two-way**. Values include **y** and **n** (default). Internally, the field is set to **n** if *Direction* is set to **incoming**. Set to **n** for the MCU. If the field is set to **y**, a warning message appears to alert the user of possible trunk lockups.

  This field specifies whether trunks in this trunk group provide disconnect supervision when seized outgoing (that is, whether these trunks should be allowed to participate in outgoing trunk-to-outgoing trunk calls). This field is set independently of the trunk-to-trunk transfer system parameter, although both options must be set appropriately if outgoing trunk-to-outgoing trunk transfers are desired.

- *Answer Supervision Timeout*

  Used for all MCU trunk types to indicate the amount of time in seconds the system allows for a call to be answered before beginning a CDR record of a call. This interval begins as soon as the outgoing trunk is seized. Values include **0** through **250**; default is **10**.

  NOTE:
  This timeout does not override answer supervision sent from the network or other firmware (DS1 port circuit timers). DS1 firmware-sent answer supervision is administered in *Outgoing End of Dial (sec)* on the Administrable Timers page of the Trunk Group form.

```
add trunk-group 11                                          Page  2 of 10

                                    TRUNK GROUP
TRUNK FEATURES
          ACA Assignment? n                           Wideband Support? n
                                                      Maintenance Tests? y
                              Data Restriction? n
                                     Send Name: n    Send Calling Number: n
                                                   Send Connected Number: n
   Suppress # Outpulsing? n
                                          UUI IE Treatment: service-provider
```

Figure 2-40. Trunk Group Form (Page 2)

Trunk Group Form, Page 2 of 10

- *ACA Assignment*

  N/A for the MCU. Leave the default value **n**.

- *Wideband Support*

  Indicates whether or not Wideband is supported on this trunk group. For 384k or 1536k bandwidth, set this field to **y**. (When Wideband is supported, page 3 of this form needs to be completed.)

- *Data Restriction*

  Leave this field set to **n**.

- *Send Name*

  N/A for the MCU. Leave at default setting **n**.

- *Send Calling Number*

  Specifies whether the calling party's number is sent on outgoing or tandemed ISDN calls. Values include **y**, **n** (default), and **r** (restricted). For value **y**, the ISDN Numbering - Public/Unknown Format form is accessed to construct the actual number to be sent, or the ISDN Numbering-Private form (based on *Numbering Format*) is used. For value **r**, the calling number is sent "presentation-restricted."

NOTE:
The ISDN Numbering - Public/Unknown Format form can override the value for *Send Calling Number* for any administrable block of extensions.

- *Send Connected Number*

  Specifies whether the calling party's number is sent on incoming or tandemed ISDN calls. Values include **y**, **n** (default), and **r** (restricted). For value **y**, the ISDN Numbering - Public/Unknown Format form is accessed to construct the actual number sent, or the ISDN Numbering-Private form (based on *Numbering Format*) is used. For value **r**, the connected number is sent "presentation-restricted."

  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 form, you should not administer Send Connected Number to **r** (restricted) on the ISDN-PRI Trunk Group form because this causes display problems.*

- *Maintenance Tests*

  Leave this field in its default setting of **y**.

- *Suppress # Outpulsing*

  Indicates end-to-end signaling begins with (and includes) the "#" when **y** is entered into this field. The final "#" is suppressed in cases where the system would normally outpulse it. This field should be set to **y** when the Central Office (for example, rotary) or any other facility treats "#" as an error; default is **n**.

- *UUI IE Treatment*

  Specifies whether the Information Element (IE) is shared. Enter value **shared** if the trunk is connected to a R6.0 or later MCU. Enter value **service-provider** (default) if the trunk is connected to a pre-R6.0 MCU or if service provider functionality is desired.

```
add trunk-group 11                                          Page  3 of 10

                                   TRUNK GROUP

                          INCOMING CALL HANDLING TREATMENT
Service/          Called     Called         Del   Insert
Feature            Len       Number
________________  __  ________________  ___  ____________________
________________  __  ________________  ___  ____________________
________________  __  ________________  ___  ____________________
________________  __  ________________  ___  ____________________
________________  __  ________________  ___  ____________________
________________  __  ________________  ___  ____________________
________________  __  ________________  ___  ____________________
________________  __  ________________  ___  ____________________
________________  __  ________________  ___  ____________________
________________  __  ________________  ___  ____________________
________________  __  ________________  ___  ____________________
________________  __  ________________  ___  ____________________
________________  __  ________________  ___  ____________________
________________  __  ________________  ___  ____________________
```

Figure 2-41. Trunk Group Form (Page 3)

Trunk Group Form, Page 3 of 10

- *Service/Feature*

  Specifies the ISDN Services/Features for an incoming call type. See *Service Type* (on page 1 of the form) for a list of predefined Services/Features that can be received.

- *Called Len*

  Specifies the number of digits received for an incoming call. A blank entry is used as a “wild card” entry and, when used, means that any length of digits associated with the specified Service/Feature can match in this field. Valid entries are **0** to **21**, or leave blank.

- *Called Number*

  Specifies the leading digits received for an incoming call. A blank entry is used as a “wild card” entry and, when used, means that any number associated with the specified Service/Feature can match in this field. Valid entries are **1** to **16** digits, or leave blank.

- *Del*

  Specifies the number of leading digits to be deleted from the incoming Called Party Number. Calls of a particular type may be administered to be routed to a single destination by deleting all incoming digits and then administering the *Insert* field with the desired extension. Valid entries are **1** to **21**, **all**, or leave blank.

- *Insert*

  Specifies the digits to be prepended to the front of the remaining digits after any (optional) digit deletion has been performed. The resultant number formed from digit deletion/insertion is used to route the call. Values include up to 16 characters consisting of a combination of the following: 0 through 9, *, #, or leave blank.

```
add trunk-group 11                                          Page  4 of 10


                                      TRUNK GROUP
                                         Administered Members (min/max):  0/0
GROUP MEMBER ASSIGNMENTS                     Total Administered Members:  0

          Port   Code Sfx Name                      Sig Grp
     1: 1A0501            _____________               1_
     2: 1A0502            _____________               1_
     3: 1A0503            _____________               1_
     4: 1A0504            _____________               1_
     5: 1A0505            _____________               1_
     6: 1A0506            _____________               1_
     7: 1A0507            _____________               1_
     8: 1A0508            _____________               1_
     9: 1A0509            _____________               1_
    10: 1A0510            _____________               1_
    11: 1A0511            _____________               1_
    12: 1A0512            _____________               1_
    13: 1A0513            _____________               1_
    14: 1A0514            _____________               1_
    15: 1A0515            _____________               1_
```

Figure 2-42. Trunk Group Form (Page 4)

Trunk Group Form, Page 4 of 10

- *Administered Members (min/max)*

  Displays on all member pages. Indicates the minimum and maximum number of members administered for this hunt group.

- *Total Administered Members*

  Displays on all member pages. Indicates the total number of members administered for this hunt group.

- *Port*

  Enter the trunk port number (up to six characters).

  NOTE:
  ESM DS1 ports (if present) should not be administered as members of any trunk group. These ports are hard-cabled to the ESM MMI ports.

- *Name*

  Enter the name of the member of the trunk group (up to 10 characters). The name can be the 7-digit telephone number or a 10-digit trunk circuit identification number (provided by the trunk supplier). The name identification is needed only for the first member of each DS1 circuit pack.

- *Sig Grp*

  Enter the signaling group number (**1** through **26**). If a DS1 interface appears in one and only one signaling group, then the number of that signaling group appears as a default in this field for any trunk on that interface.

  NOTE:
  The fields and values on pages 4 through 10 are continued from page 3. If *Wideband Support* is set to **n**, pages 4 through 9 appear. If *Wideband Support* is set to **y**, pages 5 through 10 appear.

```
display trunk-group 11                                Page  5 of 10

                                TRUNK GROUP

                          WIDEBAND SUPPORT OPTIONS

                             H0? y
                            H11? n
                            H12? n
                          NXDS0? y   Contiguous? n
```

Figure 2-43. Trunk Group Form (Page 5)

Trunk Group Form, Page 5 of 10

- *Wideband Support Options*

  For 384k bandwidth and service providers supporting NxDS0, enter **y** for *H0*. For 1472k or 1536k bandwidth, enter **y** for *H11*. For 1920k bandwidth (E1 international), enter **y** for *H12*. To specify the “N by DS-zero” multirate service, set *NXDS0* to **y**.

- *Contiguous*

  This field specifies whether to hunt contiguous NXDS0 channels. Values include **y** and **n**; the field should be set to **y**. This field appears only if *NXDS0* contains **y**.

  NOTE:
  All B-channels that are included in the wideband call must reside on the same ISDN-PRI facility.

Administering Tie Trunk Groups

The following list identifies the fields on the Trunk Group form and their values entered when administering tie trunk groups.

```
add trunk-group next                                    Page   1 of 10

                                TRUNK GROUP

Group Number: 2                    Group Type: tie          CDR Reports: y
  Group Name: OUTSIDE CALL               COR: 1                   TAC:
   Direction: two-way        Outgoing Display? n   Trunk Signaling Type:
 Dial Access? n
                                                  Incoming Destination:
   Comm Type: rbavd


         BCC: 1                           ITC: rest
TRUNK PARAMETERS
   Trunk Type (in/out):                 Incoming Rotary Timeout(sec): 5
    Outgoing Dial Type: tone                      Incoming Dial Type: tone
                                             Disconnect Timing(msec): 500
       Digit Treatment:                                      Digits:
                                                 Sig Bit Inversion: none
                                                   DTT to DCO Loss: normal
     Connected to Toll? n
    Incoming Dial Tone? y


                                  Synchronization: async        Duplex: full
Disconnect Supervision - In? y  Out? n
Answer Supervision Timeout: 0
                                          Receive Answer Supervision? y
```

Figure 2-44. Trunk Group Form (Tie Trunks), Page 1

Trunk Group Form, Page 1 of 10

- *Group Number*

  This is a display-only field that shows the number of the trunk group that is currently being accessed.

- *Group Type*

  Enter **tie** to identify the trunk group.

- *CDR Reports*

  Enter **y** if tracking calls on this trunk group. Leave value **n** if not tracking calls.

- *Group Name*

  Enter a name of up to 27 characters to describe this trunk group (for example, **isdn-pri to AT&T**, **circuit # of span**). The default is **OUTSIDE CALL.**

- *COR*

  Leave this field at the default value **1**.

- *TAC*

  Enter a number containing one to four digits (including #) for the trunk access code that matches the numbering plan assigned on the dial plan for the TAC. (Character # can be used as the first digit.) CDR uses the TAC to identify the trunk group on CDR reports. Allowable entries must match the dial plan.

- *Direction*

  Values include **incoming**, **outgoing**, and **two-way** (default), depending on how the associated trunk group is being used. (For example, if the far end is administered for outgoing calls only on this trunk group, the field should be set to **incoming** for the MCU trunk group.)

- *Outgoing Display*

  N/A on the MCU. Leave at default value **n**.

- *Trunk Signaling Type*

  Specifies the kind of signaling to use for members in the trunk group; if the field is administered to a non-blank value, it severely restricts the ports that can be administered in the trunk group. If *Group Type* is set to “tie,” values include **cont** (continuous), **dis** (discontinuous), **tge**, **tgi**, and **tgu**. Value **cont** or **dis** indicates that only ports from a TN2140 Italian tie trunk may be administered as trunk group members. Value **tge**, **tgi**, or **tgu** indicates that only ports from a TN497 Italian Tie trunk can be administered as trunk group members. If the field is blank, ports from any other tie trunk circuit pack may be added as trunk group members.

- *Dial Access*

  Enter **y** to allow this trunk group to be accessed via a TAC; otherwise, enter **n**.

  NOTE:
  Access via ARS should be used for all calls regardless of this field value.

- *Comm Type*

  Set this field to **rbavd** for DS1 T1 service RBS trunks.

- *BCC*

  This field indicates barrier capabilities. Possible values include **0** (voice), **1** (56k robbed-bit), **2** (up to 19.2 [modem is set up]), **3** (restricted 64k), or **4** (64k clear).

- *ITC*

  Leave this field at the default setting of **rest** for restricted.

- *Trunk Type (in/out):*

  Identifies the physical type of incoming and outgoing trunks. Values include **auto/auto**, **auto/delay**, **auto/immed**, **auto/wink**, **delay/auto**, **delay/delay**, **delay/immed**, **delay/wink**, **immed/auto**, **immed/delay**, **immed/immed**, **immed/wink**, **wink/auto**, **wink/delay**, **wink/immed**, and **wink/wink**. The far end must be compatible with the MCU. (For example, value **wink/wink** can be used at both ends; value **wink/delay** at one end requires value **delay/wink** at the other end.)

- *Incoming Rotary Timeout (sec)*

  Specify the number of seconds before an incoming rotary call is timed out. Leave at default value **5**.

- *Outgoing Dial Type*

  Leave at default value **tone**. Values **mf** and **rotary** are not used on the MCU.

- *Incoming Dial Type*

  Leave at default value **tone**. Values **mf** and **rotary** are not used on the MCU.

- *Disconnect Timing*

  Specify in milliseconds the disconnect time. Leave at default value **500** (milliseconds).

- *Digits*

  If either **insertion** or **absortion** was administered in the *Digit Treatment* field, this field should contain either the digits, *****, or **#** that need to be inserted, or the number of digits (**1** through **5**) that needs to be absorbed. If *Digit Treatment* was left blank, this field should be left blank (default).

- *Digit Treatment*

  If inserting digits at the trunk group level, set this field to **insertion**. If absorbing digits at the trunk group level, enter **absorption**. Otherwise, leave blank (default).

- *Sig Bit Inversion*

  This field allows the user to administer the signaling bit inversion (the A-bit and B-bit inversion) on a per-port basis, and it applies only to TN464B or later circuit packs. The CCITT standards define on-hook to be “1” in the A-bit and off-hook to be “0.” This definition is the reverse of the EIA definition. The default in the TN464 firmware for signaling bit inversion is to use the CCITT standard for 32-channel mode and to use the EIA standard for 24-channel mode. Values include **A**, **B**, **A&B**, and none (default).

- *DTT to DCO Loss*

  Specifies the loss plan to be used for the associated trunk group. This field should be left at the default value **normal**.

- *Connected to Toll*

  N/A for the MCU. Leave this field set to **n**.

- *Incoming Dial Tone*

  Leave this field set to **y**.

- *Synchronization*

  N/A for the MCU. Leave this field set to **async**.

- *Duplex*

  N/A for the MCU. Leave this field set to default value **full**.

- *Disconnect Supervision...In?*

  Displays when *Direction* is set to **incoming** or **two-way**. Values include **y** (default) and **n**. Internally, the field is set to **n** if *Direction* is set to **outgoing**. For the MCU, enter **y** to indicate that the far end can release a trunk when the calling party releases from a call and the system is able to recognize the release signal.

- *Disconnect Supervision...Out?*

  Displays when *Direction* is set to **outgoing** or **two-way**. Values include **y** and **n** (default). Internally, the field is set to **n** if *Direction* is set to **incoming**. Set to **n** for the MCU. If the field is set to **y**, a warning message appears to alert the user of possible trunk lockups.

  This field specifies whether trunks in this trunk group provide disconnect supervision when seized outgoing (that is, whether these trunks should be allowed to participate in outgoing trunk-to-outgoing trunk calls). This field is set independently of the trunk-to-trunk transfer system parameter, although both options must be set appropriately if outgoing trunk-to-outgoing trunk transfers are desired.

- *Answer Supervision Timeout*

  Used for all trunk types to indicate the amount of time in seconds that the system allows for a call to be answered before beginning a CDR record of a call. This interval begins as soon as the outgoing trunk is seized. Values include **0** to **250**; default is **10**.

  NOTE:
  This timeout does not override answer supervision sent from the network or other firmware (DS1 port circuit timers). DS1 firmware sent answer supervision is administered in *Outgoing End of Dial (sec)* on the Administrable Timers page of the Trunk Group form.

- *Receive Answer Supervision?*

  Specifies whether the network provides answer supervision for the trunks. Answer supervision refers to notification from the far end that the called party has gone off-hook to answer the call. When answer supervision is not provided, the originating switch can classify a call as answered using

the *Answer Supervision Timeout* field entry (for domestic circuit packs) or, for circuit packs that take administrable timers, by using *Outgoing End of Dial (sec)*. For trunks over which answer supervision is provided by the network, set this field to **y** (default) and *Answer Supervision Timeout* to **0** to disable the firmware timeout. The firmware timeout is set automatically to **0** on circuit packs that take the administrable *Outgoing End of Dial (sec)* field. *Receive Answer Supervision* must be set to **n** if *Direction* is set to **incoming**.

```
add trunk-group next                                    Page  2 of 10

                                    TRUNK GROUP

TRUNK FEATURES
          ACA Assignment? n
                                                        Maintenance Tests? y
                                 Data Restriction? n




   Suppress # Outpulsing? n
                     Seize When Maintenance Busy: neither-end


Incoming Tone (DTMF) ANI: no
         Connected to CO? n
```

Figure 2-45. Trunk Group Form (Tie Trunks), Page 2

Trunk Group Form, Page 2 of 10

- *ACA Assignment*

  N/A for the MCU. Leave the default value **n**.

- *Data Restriction*

  N/A for the MCU. Leave the default value **n**.

- *Suppress # Outpulsing*

  Leave this field set to **n**.

- *Maintenance Tests*

  Values include **y** and **n**. Value **y** enables hourly maintenance tests to be performed automatically on the associated trunk group.

- *Seize When Maintenance Busy*

  This field is effective only for ports on the TN767 and TN464C (or later release) circuit packs. It indicates whether an outgoing seize is generated when a trunk is maintenance-busied, and whether the far-end switch is administered to do likewise. This supports the Electronic Tandem Network Busyout feature. The intended use of this feature is to prevent the far-end switch from reporting problems concerning a trunk that has been removed from service at the near end. Setting this field has no effect on the behavior of the far-end machine; it controls the behavior of the near-end switch and the expected far-end behavior.

  If the administered value of this field is either **near-end** or **both-ends**, an outgoing seize will be initiated whenever a trunk in this trunk group is maintenance-busied. The seize is maintained until the maintenance busyout is released. This action will likely cause alarms at the far-end switch, perhaps leading to a far-end maintenance busyout, unless the far-end 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 seize (including failure to drop from a completed call) is assumed to be the result of a far-end maintenance busy condition. Note that this assumption may be incorrect, since the abnormally long seize may actually be due to a transient or permanent failure of the trunk circuit. Regardless of the cause of the abnormally long seize, the near-end responds by (1) generating a warning alarm indicating that the trunk is assumed to be maintenance busy at the far-end; (2) removing the trunk from service; and, (3) keeping the trunk out of service until a far-end disconnect is received. The values allowed depend on the value in the *Direction* field. When *Direction* is set to **incoming**, the values are **far-end** and **neither-end**.

  When *Direction* is set to **outgoing**, the values are **near-end** and **neither-end**. When Direction is set to **two-way**, the values are **both-ends**, **far-end**, **near-end**, and **neither-end**.

- *Incoming Tone (DTMF) ANI*

  Displays only when *Incoming Dial Type* on page 1 of the form is set to **tone**. Values include ***ANI*DNIS***, **ANI*DNIS***, and **no** (default). For value ***ANI*DNIS***, the far end must provide a leading " * ." For value **ANI*DNIS***, the far end does not provide a leading " * ." These ANI digits are printed on a CDR record, passed to the AUDIX Intuity and ASAI interface, and displayed on the station set (and on tandem calls if the outgoing trunk requires ANI). Then this received ANI is sent to the outgoing trunk.

- *Connected to CO*

  Displays when *Group Type* is set to **tie**. Permits overlap sending to a Central Office. Values include **y** and **n** (default).

```
add trunk-group next                                    Page  3 of 10

                                  TRUNK GROUP

ADMINISTRABLE TIMERS
  Incoming Disconnect(msec): 500              Outgoing Disconnect(msec): 500
  Incoming Dial Guard(msec): 70               Outgoing Dial Guard(msec): 1600
 Incoming Glare Guard(msec): 1500            Outgoing Glare Guard(msec): 1500




                                        Outgoing Seizure Response(sec): 5
Programmed Dial Pause(msec): 1500         Disconnect Signal Error(sec): 240
                                  Incoming Incomplete Dial Alarm(sec): 255



  END TO END SIGNALING
      Tone(msec): 350    Pause(msec): 150


  OUTPULSING INFORMATION
       PPS: 10    Make(MSC): 40   Break(msec):60
```

Figure 2-46. Trunk Group Form (Tie Trunks), Page 3

Trunk Group Form, Page 3 of 10

- *Incoming Disconnect (msec)*

  A disconnect signal on an incoming call is considered valid only if it continues at least for the time specified by this timer. Valid entries are **50** through **2550** in increments of 10. The field cannot be blank; default is **500**. The field appears only for an incoming or two-way trunk group when *Trunk Signaling Type* is either blank or **cont**.

- *Incoming Dial Guard (msec)*

  Enter the interval between the detection of an incoming seizure and the acceptance of the first digit. The MCU will not accept digits before this timer expires. Valid entries are **10** through **2550** in increments of 10; default is **70**.

- *Incoming Glare Guard (msec)*

  Enter the minimum time (in msec) between the time an incoming trunk is released and the same trunk can be seized for an outgoing call. This guard time ensures that the far-end has released all equipment after the incoming trunk is released. Valid entries are **100** to **25500** in increments of 100. The field cannot be blank. Default is **1500**. This field only appears when the trunk group *Direction* is **two-way**.

- *Outgoing Disconnect (msec)*

  A disconnect signal on an outgoing call is considered valid only if it continues at least for the time specified by this timer. This timer begins timing when a disconnect signal has been detected on an outgoing call over the trunk, and it resets when the signal is no longer detected. If the timer times out, the trunk drops. Valid entries are **50** to **2550** in increments of 10. The field cannot be blank. Default is **500**.

- *Outgoing Dial Guard (msec)*

  This timer specifies the minimum time between seizure acknowledgment of a trunk and the outpulsing of digits. For trunks that do not provide seizure acknowledgment, the timer specifies the minimum time between seizure and the outpulsing of digits. Any digit that is dialed by the station user after the user lifts the receiver, but before the timer times out, is not outpulsed until the timer expires. Valid entries are **100** to **25500** in increments of 100. The field cannot be blank. Default is **1600**.

- *Outgoing Glare Guard (msec)*

  Enter the minimum time (in msec) between the time an outgoing trunk is released and the same trunk can be seized for another outgoing call. This guard time will ensure that the far-end has released all equipment after the outgoing trunk is released. Valid entries are **100** to **25500** in increments of 100. The field cannot be blank. Default is 1500. This field appears only for outgoing and two-way trunk groups.

- *Programmed Dial Pause (msec)*

  Enter the length (in msec) of a pause during abbreviated dialing, ARS outpulsing, and terminal dialing operations. This timer is administrable for all outgoing and two-way trunk groups. Valid entries are **100** to **25500** in increments of 100. Default is **1500**. This timer is sent to the TN2207 and TN767 circuit packs. All CO circuit packs that accept administrable timers accept this timer.

- *Outgoing Seizure Response (sec)*

  Enter a maximum time (in sec) by which the MCU should receive the proceed-to-dial indication from the far-end after sending the seize signal. If the indication is not received in this time, a seizure failure response is uplinked. Valid entries are **1** to **255** in increments of 1. Default is **5**. This timer is sent to the TN2207 and TN767 circuit packs.

- *Disconnect Signal Error (sec)*

  Enter the time (in sec) within which a disconnect signal from the far-end is expected after the internal party (a station or a tie trunk) goes on-hook. If the timer expires, the system can assume a disconnect failure and take an appropriate action (such as an error message). This field appears for ground-start trunk groups. Valid entries are **1** to **255** in increments of 1; default is **240**.

- *Incoming Incomplete Dial Alarm (sec)*

  Enter the maximum time (in sec) allowed between incoming seizure and receipt of all digits. Intervals greater than this generate an inline error. Valid entries are **1** to **255** (default) in increments of 1. Only the TN2207 and TN767 circuit packs accept this timer.

- *Tone (msec)*

  Enter the length of a DTMF tone sent in response to a button depression at a hybrid station set. Valid entries are **20** to **2550** in increments of 10; the default is **350**. This field appears only if *Trunk Signaling Type* is blank. This timer is only sent to the TN2207 and TN767 circuit packs.

- *Pause (msec)*

  Enter the minimum length of a pause between tones being sent in response to button depressions at a hybrid station set. It is administrable only if the *Trunk Signaling Type* is blank. Valid entries are **20** to **2550** in increments of 10; default is **150**. This timer is only sent to the TN2207 and TN767 circuit packs.

- *PPS*

  Enter either **10** or **20** to specify the PPS (pulses per second) assigned to rotary dial pulse trunk groups. Valid entries are **10** (default) and 20.

  NOTE:
  The TN767 circuit pack can do only 10pps.

- *Make (msec)*

  Enter the duration of the make interval (the pause between pulses) while the system is outpulsing digits using dial pulse signaling. If *PPS* is **10**, the sum of *Make (msec)* and *Break (msec)* must equal **100**. Valid entries are **20** to **80** in increments of 5. The field cannot be blank. The default is **40**. If *PPS* is **20**, the sum of *Make (msec)* and *Break (msec)* must equal **50**. Valid entries are **10** to **40** in increments of 5. The field cannot be blank. The default is **20**.

- *Break (msec)*

  Enter the duration of the break interval (the pulse duration) while the system is outpulsing digits using dial pulse signaling. If *PPS* is **10**, the sum of *Make (msec)* and *Break (msec)* must equal **100**. Valid entries are **20** to **80** in increments of 5. The field cannot be blank. The default is **60**. If *PPS* is **20**, the sum of *Make (msec)* and *Break (msec)* must equal **50**. Valid entries are **10** to **40** in increments of 5. The field cannot be blank. The default is **30**.

```
add trunk-group next                                    Page  4 of 10


                                        TRUNK GROUP
                                           Administered Members (min/max):  0/0
GROUP MEMBER ASSIGNMENTS                       Total Administered Members:  0

        Port     Code Sfx Name                       Mode      Type  Ans Delay

     1: _______           ____________               _________ ______  ____
     2: _______           ____________               _________ ______  ____
     3: _______           ____________               _________ ______  ____
     4: _______           ____________               _________ ______  ____
     5: _______           ____________               _________ ______  ____
     6: _______           ____________               _________ ______  ____
     7: _______           ____________               _________ ______  ____
     8: _______           ____________               _________ ______  ____
     9: _______           ____________               _________ ______  ____
    10: _______           ____________               _________ ______  ____
    11: _______           ____________               _________ ______  ____
    12: _______           ____________               _________ ______  ____
    13: _______           ____________               _________ ______  ____
    14: _______           ____________               _________ ______  ____
    15: _______           ____________               _________ ______  ____
```

Figure 2-47. Trunk Group Form (Tie Trunks), Page 4

Trunk Group Form, Page 4 of 10

- *Administered Members (min/max)*

  Displays on all member pages. Indicates the minimum and maximum number of members administered for this hunt group.

- *Total Administered Members*

  Displays on all member pages. Indicates the total number of members administered for this hunt group.

- *Port*

  Enter the trunk port number (up to six characters).

- *Name*

  Enter the name of the member of the trunk group (up to 10 characters). The name can be the seven- or 10-digit telephone number or trunk circuit identification number (provided by the trunk supplier). The name identification is required only on the first member of each DS1 circuit pack.

- *Mode*

  N/A for the MCU. Leave blank.

- *Type*

  N/A for the MCU. Leave blank.

- *Ans Delay*

  N/A for the MCU. Leave blank.

ISDN Numbering — Public/Unknown Format Form

The ISDN Numbering — Public/Unknown Format form is used to construct the actual calling number to be sent for incoming, outgoing, or tandemed ISDN calls. To access the form, enter the **change isdn public-unknown-numbering** command. An example of the form appears as follows:

```
change isdn public-unknown-numbering                          Page  1 of  8

                        ISDN NUMBERING - PUBLIC/UNKNOWN FORMAT

Ext    Ext          CPN          Total CPN      Ext    Ext          CPN          Total CPN
Len    Code         Prefix         Length       Len    Code         Prefix         Length

_    _______  ________________    ___          _    _______  ________________    ___
_    _______  ________________    ___          _    _______  ________________    ___
_    _______  ________________    ___          _    _______  ________________    ___
_    _______  ________________    ___          _    _______  ________________    ___
_    _______  ________________    ___          _    _______  ________________    ___
_    _______  ________________    ___          _    _______  ________________    ___
_    _______  ________________    ___          _    _______  ________________    ___
_    _______  ________________    ___          _    _______  ________________    ___
_    _______  ________________    ___          _    _______  ________________    ___
_    _______  ________________    ___          _    _______  ________________    ___
_    _______  ________________    ___          _    _______  ________________    ___
_    _______  ________________    ___          _    _______  ________________    ___
_    _______  ________________    ___          _    _______  ________________    ___
_    _______  ________________    ___          _    _______  ________________    ___
_    _______  ________________    ___          _    _______  ________________    ___
_    _______  ________________    ___          _    _______  ________________    ___
_    _______  ________________    ___          _    _______  ________________    ___
_    _______  ________________    ___          _    _______  ________________    ___
```

Figure 2-48. ISDN Numbering - Public/Unknown Format Form

The following list discusses the fields in the form.

- *Ext Len*

  Specifies the number of digits the extension can have. Enter a number from **0** through **5** that corresponds to the extension lengths allowed by the dial plan, or leave blank.

- *Ext Code*

  Allows for groups of extensions to be administered. The value can contain a maximum of five digits based on the value in the *Ext Len* field.

- *CPN Prefix*

  Specifies the number that is added to the beginning of the extension to form a 10-digit CPN number.

- *Total CPN Length*

  Enter the total number of digits to send. Values include **0** through **15**, or leave blank.

ISDN Numbering — Private Format Form

This form supports Private Numbering Plans (PNPs). It allows you to specify the digits to be put into the Calling Number IE, the Connected Number Information Element (IE), and the QSIG Party Number for extensions in the Private Numbering Plan. To access the form, enter the **change isdn private-numbering** command. An example of the form appears as follows.

```
change isdn private-numbering                          Page  1 of  1

                    ISDN NUMBERING - PRIVATE FORMAT

          Network Level: _
           Level 2 Code: _____            Deleted Digits: 0
           Level 1 Code: _____
```

Figure 2-49. ISDN Numbering — Private Format Form

The following list discusses the fields in the form.

- *Network Level*

  N/A on MCU. Leave blank.

- *Level 2 Code*

  N/A on MCU. Leave blank.

- *Level 1 Code*

  N/A on MCU. Leave blank.

- *Deleted Digits*

  N/A on MCU. Leave blank.

Administering the ESM from the MSM

To administer the ESM from the MSM, you must first establish the MSM-ESM link and then enable T.120 data transfer within the conference. You can administer the ESM for a connection over either the processor interface (PI) link or the PACCON link.

Procedure For Connection Over the PI LInk

To administer the ESM from the MSM with a connection over the PI link, complete the fields in the forms identified in the following table. Appropriate page references to the forms are provided. Detailed instructions for completing the forms are provided after the table

Table 2-31. Required Forms For ESM Administration Over the PI Link

Form	Field	Page Reference
Set Date and Time	■ All	page 2-9
Customer Options	■ *Max. T.120 Port Capacity*	page 2-22
DS1 Circuit Pack	■ (See the following list.)	page 2-61
Data Module (Processor Interface)	■ (See the following list.)	page 2-78
Interface Links	■ (See the following list.)	page 2-81
Processor Assignments Channel	■ (See the following list.)	page 2-84
Signaling Group	■ (See the following list.)	page 2-87
Interface Links	■ *Enable*	page 2-81
Conference Record	■ *Audio Mode* ■ *T.120 Data Mode* ■ *MLP Rate* ■ *App Comp?*	page 1-3

Establish the MSM-ESM link as follows:

1. Verify that the time setting on the MSM is correct by entering the **display time** command. If you need to correct the time, enter the **set time** command and complete the fields in the Set Date and Time form.
2. Access the Customer-Options form by entering **change system-parameters customer-options**. Enter the maximum number of T.120 ports that can be allocated in the system at the same time into the *Max. T.120 Port Capacity* field. Then submit the form.
3. Access the DS1 (UDS1) Circuit Pack form by entering **change ds1 *location***, where ***location*** is the location of the DS1 circuit pack in the carrier. Ensure that the MMI cabling board is administered, and populate each field in the following list with the value specified:
   - *Bit Rate*: **2.048**
   - *Line Coding*: **hdb3**
   - *Signaling Mode*: **isdn-pri**
   - *Connect*: **pbx**
   - *Interface*: **network**
   - *MMI Cabling Board*: (location of this item; for example, **1A10**)
   - *MMI Interface*: **pi_esm**
   - *CRC*: **y**
   - *Slip Detection*: **y**

   Submit the form.
4. Add a Processor Interface data extension by populating the Data Module form. To access the form, enter **add (change) data-module** and the appropriate data extension (for example, **change data-module *6881***). Populate each field in the following list with the value specified:
   - *Type*: **procr-intf**
   - *Physical Channel*: Select an available channel (for example, **01**)
   - *Name*: **PROCR-INTF #1**
   - *COS*: Select the appropriate COS number, or leave as a default
   - *COR*: Select the appropriate COR number, or leave as a default
   - *Maintenance Extension*: Select a maintenance extension (for example, **6882**)

   Submit the form.

5. Access the Interface Links form by entering **change communication-interface links**. Then populate each field in the following list with the value specified.
   - *Enable*: **n** (for now!)
   - *Est Conn*: **y**
   - *Prot*: **ISDN**
   - *Destination Brd*: (address of the relevant DS1 or UDS1 circuit pack)

   Submit the form.
6. Access the Processor Channel Assignments form by entering **change communication-interface processor-channels.** Populate each field in the following list with the value specified.
   - *Link:* (link number administered in the Interface Links form in the previous step)
   - *Appl:* **ESM**
   - *Channel*: blank
   - *Remote Proc Chan*: blank
   - *Priority*: **high**

   Submit the form.
7. Access the Signaling Group form by entering **add signaling-group *x***, where ***x*** is either the signaling group number (**1** through **8**) or **next**. Ensure that channel 16 is represented in the *Primary D-Channel* field (for example, **A0816**, where A08 is the location of the UDS1 circuit pack). Then complete the form according to the directions provided in the “Signaling Group Form” section in this chapter. Finally, submit the form.
8. Access once again the Interface Links form by entering **change communication-interface links.** Then enable the link by setting the *Enable* field to **y**. Finally, submit the form.

To enable T.120 data transfer within the conference, access the Conference Record by entering **change conference *xxx***, where ***xxx*** is the conference number. Then populate each field in the following list with the value specified.

- *T.120 Data Mode*: **any-mlp**
- *App Comp?*: **y** (for each endpoint)

Submit the form.

Procedure For Connection Over the PACCON Link

To administer the ESM from the MSM with a connection over the PACCON link, complete the fields in the forms identified in the following table. Appropriate page references to the forms are provided. Detailed instructions for completing the forms are provided after the table

Table 2-32. Required Forms For ESM Administration Over the PI Link

Form	Field	Page Reference
Set Date and Time	■ All	page 2-9
Customer Options	■ *Max. T.120 Port Capacity*	page 2-22
DS1 Circuit Pack	■ (See the following list.)	page 2-61
Signaling Group	■ (See the following list.)	page 2-87
Conference Record	■ *Audio Mode* ■ *T.120 Data Mode* ■ *MLP Rate* ■ *App Comp?*	page 1-3

Establish the MSM-ESM link as follows:

1. Verify that the time setting on the MSM is correct by entering the **display time** command. If you need to correct the time, enter the **set time** command and complete the fields in the Set Date and Time form.
2. Access the Customer-Options form by entering **change system-parameters customer-options**. Enter the maximum number of T.120 ports that can be allocated in the system at the same time into the *Max. T.120 Port Capacity* field. Then submit the form.
3. Access the DS1 (UDS1) Circuit Pack form by entering **change ds1 *location***, where ***location*** is the location of the DS1 circuit pack in the carrier. Ensure that the MMI cabling board is administered, and populate each field in the following list with the value specified:
   - *Bit Rate*: **2.048**
   - *Line Coding*: **hdb3**
   - *Signaling Mode*: **isdn-pri**
   - *Connect*: **pbx**
   - *Interface*: **network**
   - *MMI Cabling Board*: (location of this item; for example, **1A10**)

- *MMI Interface*: **pac_esm**
- *CRC*: **y**
- *Slip Detection*: **y**

Submit the form.

4. Access the Signaling Group form by entering **add signaling-group *x***, where ***x*** is either the signaling group number (**1** through **8**) or **next**. Ensure that channel 16 is represented in the *Primary D-Channel* field (for example, **A0816**, where A08 is the location of the UDS1 circuit pack). Then complete the form according to the directions provided in the “Signaling Group Form” section in this chapter. Finally, submit the form.

To enable T.120 data transfer within the conference, access the Conference Record by entering **change conference *xxx***, where ***xxx*** is the conference number. Then populate each field in the following list with the value specified.

- *T.120 Data Mode*: **any-mlp**
- *App Comp?*: **y** (for each endpoint)

Submit the form.

Administering Dial Plans

The MCU internal dial plan is the system’s guide to digit translation. The dial plan is designed to give greater flexibility to digit translations. The entire dial plan is dependent on the first digit dialed. The first digit can be one of 11 possible characters, including 1 through 9, *, and #.

When a digit is dialed, the system must know what to expect, based on that digit. For example, if a data module user dials a “4,” the system must know how many more digits to expect and what code type to expect before it can process the call.

The dial plan is administered at system initialization by the system administrator.

Dial plans are available for Automatic Alternate Routing (AAR), Automatic Route Selection (ARS), and the Uniform Dialing Plan (UDP).

Automatic Alternate Routing (AAR)

Automatic Alternate Routing (AAR) provides alternate routing choices for private on-network calls. AAR also provides digit modification to allow on-network calls to route through the public network when on-network routes are not available. Before the forms necessary to enable AAR can be administered, either the Private Networking or Uniform Dialing Plan option on the Customer-Options form must be enabled.

To implement AAR, complete the fields in the forms identified in the following table. Appropriate page references to the forms are provided.

Table 2-33. Required Forms and Fields for AAR

Form	Field	Page Reference
Dial Plan Record	■ All	page 2-108
Class of Restriction	■ *FRL*	page 2-31
Feature Access Code	■ *AAR Access Code* ■ *ARS Access Code*	page 2-157
AAR and ARS Digit Analysis Table	■ All	page 2-160
AAR and ARS Digit Conversion Table	■ All	page 2-171
Route Pattern	■ All	page 2-185
Feature-Related System Parameters	■ AAR/ARS Dial Tone Required	page 2-43

Automatic Route Selection (ARS)

Automatic Route Selection (ARS) allows calls to be routed over the public network according to the preferred (usually the least expensive) route available at the time you place the call. Before the necessary forms can be administered to enable ARS, the ARS option on the Customer-Options form must be enabled.

To implement ARS, complete the fields in the forms identified in the following table. Appropriate page references to the forms are provided.

Table 2-34. Required Forms and Fields for ARS

Form	Field	Page Reference
Class of Restriction	■ *FRL*	page 2-31
Dial Plan Record	■ *Area Code* ■ *ARS Prefix 1 Required*	page 2-108
Feature Access Code	■ *ARS Access Code* ■ *AAR Access Code*	page 2-157
AAR and ARS Digit Analysis Table	■ All	page 2-160
AAR and ARS Digit Conversion Table	■ All	page 2-171
Route Pattern	■ All	page 2-185
Feature-Related System Parameters	■ *AAR/ARS Dial Tone Required*	page 2-43

Uniform Dial Plan (UDP)

The Uniform Dial Plan (UDP) provides a common 4- or 5-digit dial plan that can be shared among several switches. Interswitch dialing and intraswitch dialing are enabled via 4- or 5-digit dialing. UDP is used with Electronic Tandem Networks (ETNs) and Main/Satellite/Tributary configurations. Also, UDP can be used alone to provide uniform 4- or 5-digit dialing between two or more private switching systems without Main/Satellite/Tributary or DCS configurations.

To implement UDP, complete the fields in the forms identified in the following table. Appropriate page references to the forms are provided.

Table 2-35. Required Forms and Fields for UDP

Form	Fields	Reference
Dial Plan	*Uniform Dial Plan*	page 2-108
AAR Analysis	All	page 2-160
AAR and AAR Digit Conversion	All	page 2-171
AAR and ARS Digit Analysis	All	page 2-160
Route Pattern	All	page 2-185

Administering Feature Access Codes

Feature Access Codes (FACs) are used to activate or deactivate certain system features. The following commands are used to administer the Feature Access Code form.

Table 2-36. System Parameters Features Administration Commands

Action	Object	Qualifier
change	feature-access-codes	
display	feature-access-codes	print or schedule

Use the following procedure to change the FAC to activate or deactivate certain system features:

1. At the *command:* prompt, type **change feature-access-codes** and then press RETURN.
2. Verify that the screen displays the *Feature Access Code* screen.
3. Use TAB or RETURN to advance to the fields you want to change.
4. When the form is completed, verify that the MT displays: *Command successfully completed* followed by *command:*
5. To verify the form, type **display feature-access-codes** and then press RETURN.

Feature Access Code Form

The Feature Access Code form appears as follows:

```
change feature-access-codes                              Page  1 of  2

                        FEATURE ACCESS CODE (FAC)




  Auto Alternate Routing (AAR) Access Code: 2__
Auto Route Selection (ARS) - Access Code 1: 9__     Access Code 2:____



               Data Origination Access Code: ____
                   Data Privacy Access Code: #35_
```

Figure 2-50. Feature Access Code Form (Page 1)

```
change feature-access-codes                              Page  2 of  2

                          FEATURE ACCESS CODE (FAC)

              Facility Test Calls Access Code:


                             ISDN Access Code:
```

Figure 2-51. Feature Access Code Form (Page 2)

The following list provides the fields on the Feature Access Code form and their related values.

NOTE:
Before you can use this form, you must administer the value for the FAC on the Dial Plan form.

Page 1:

- *Auto Alternate Routing (AAR) Access Code*

  Enter a 1- to 4-digit number to be used to access AAR. Character * or # may be used as the first digit.

- *Auto Route Selection (ARS)-Access Code 1*

  Enter the digits required to access this feature. This field is usually set to **9** unless this digit is required by the MCU for incoming extension numbers from the network (DNIS digits).

- *Auto Route Selection (ARS)-Access Code 2*

  Enter the digits required to access this feature. Generally, this field is not used.

  NOTE:
  This field could be used if two different customers are using the MCU and need to use separate trunk facilities for dialing out. However, this would not be supported by CRCS.

- *Data Origination Access Code*

  This field can be used if RBS DS1 service is in effect and if remote access is used with modem pooling on the MCU. A caller using the RBS trunks would have to use the access code to have the MCU insert the modem pool onto the call.

- *Data Privacy Access Code*

  N/A for the MCU. Leave blank.

Page 2:

- *Facility Test Calls Access Code*

  N/A for the MCU. Leave blank.

- *ISDN Access Code*

  Used to place an ISDN call without using ARS, AAR, or UDP. Enter a 1- to 4-digit number; * or # may be used as the first digit.

Administering AAR and ARS Digit Analysis

AAR Digit Analysis and ARS Digit Analysis are used to map customer-dialed strings to route patterns.

ARS Digit Analysis analyzes digits that are dialed for a public network call. The digits are analyzed for routing information. The feature compares the dialed digits to entries in the ARS Digit Analysis table and maps the digits to a specific route pattern. This is done as follows:

- Dialed digits that are entered are matched in digit analysis and mapped to a specific route pattern.
- Flexible dialing permits dialed strings containing a maximum of 18 digits to be analyzed so that the proper route can be determined.
- Wild cards can be used in digit analysis to simplify the administration of complex networks.
- Selected route patterns are used to route the call.

Use the following commands to administer the AAR and ARS Digit Analysis Table form.

Table 2-37. Commands for AAR and ARS Digit Analysis

Action	Object	Qualifier*
change	aar analysis ars analysis	Enter digits between 0 to 9, ‘x’ or ‘X’ (dialed string) [‘partition’ 1-8] [‘min’(1-28)]
display	aar analysis ars analysis	Enter digits between 0 to 9, ‘x’ or ‘X’ (dialed string) [‘partition’ 1-8] [‘min’ (1-28)] [’print’ or ’schedule’]
list	aar analysis ars analysis	[‘start’ string] [‘count’ 1-2000] [‘route’(1-254 or r1-r32)], [‘partition’ (1-8)], [‘node’ (1-999)], [‘to-node’ (1-999), [‘print’ or ’schedule’]
list	aar route-chosen ars route-chosen	Enter dialed number, [‘partition’ (1-8)], [’print’ or ’schedule’]

* Some qualifiers are available only with recent MCU configurations. Brackets [] indicate the qualifier is optional. Single quotes (‘’) indicate the text inside the quote must be entered exactly as shown or an abbreviated form of the word may be entered. MAX is the maximum number available in your system configuration.

AAR and ARS Digit Analysis Tables

The AAR and ARS Digit Analysis forms are organized into two parts. The left side of the form displays values that are already administered for routing calls. The right side of the form is a reproduction of the left side except that the fields are blank. If additional administration is required, you can do this by populating the blank fields on the right side of the form. This checks any new administration with existing ARS analysis translations, and it prevents duplicate entries. By using the information that is provided on the left side of the form, you can check any new administration for duplication.

The forms appear as follows:

```
change aar analysis 1                                    Page  1 of  1

                              AAR DIGIT ANALYSIS TABLE
                            Partitioned Group Number: 1    Percent Full:6

     Dialed    Total Rte Call Nd ANI       Dialed    Total Rte Call Nd ANI
     String    Mn Mx Pat Type Num Rq       String    Mn Mx Pat Type NUM Rq
2____________ 7_ 7_ 254 aar   _  n     ____________ __ __ ___ ___   _   n
3____________ 7_ 7_ 254 aar   _  n     ____________ __ __ ___ ___   _   n
4____________ 7_ 7_ 254 aar   _  n     ____________ __ __ ___ ___   _   n
5____________ 7_ 7_ 254 aar   _  n     ____________ __ __ ___ ___   _   n
6____________ 7_ 7_ 254 aar   _  n     ____________ __ __ ___ ___   _   n
7____________ 7_ 7_ 254 aar   _  n     ____________ __ __ ___ ___   _   n
8____________ 7_ 7_ 254 aar   _  n     ____________ __ __ ___ ___   _   n
9____________ 7_ 7_ 254 aar   _  n     ____________ __ __ ___ ___   _   n
____________ __ __ ___ ___    _  n     ____________ __ __ ___ ___   _   n
____________ __ __ ___ ___    _  n     ____________ __ __ ___ ___   _   n
____________ __ __ ___ ___    _  n     ____________ __ __ ___ ___   _   n
____________ __ __ ___ ___    _  n     ____________ __ __ ___ ___   _   n
```

Figure 2-52. AAR Digit Analysis Table

```
change ars analysis 0                                    Page  1 of  1

                              ARS DIGIT ANALYSIS TABLE
                            Partitioned Group Number: 1    Percent Full: 6

     Dialed    Total Rte Call Nd ANI      Dialed    Total Rte Call Nd ANI
     String    Mn Mn Pat Type Num Rq      String    Mn Mn Pat Type Num Rq
0____________ 1  1  den op    _  n    ____________ _  _  ___ __    _  n
0____________ 8  8  den op    _  n    ____________ _  _  ___ __    _  n
0____________ 11 11 den op    _  n    ____________ _  _  ___ __    _  n
00___________ 2  2  den op    _  n    ____________ _  _  ___ __    _  n
01___________ 9  17 den iop   _  n    ____________ _  _  ___ __    _  n
011__________ 10 18 den intl  _  n    ____________ _  _  ___ __    _  n
101xxxx0_____ 8  8  den op    _  n    ____________ _  _  ___ __    _  n
101xxxx0_____ 18 18 den op    _  n    ____________ _  _  ___ __    _  n
101xxxx01____ 16 24 den iop   _  n    ____________ _  _  ___ __    _  n
101xxxx011___ 17 25 den intl  _  n    ____________ _  _  ___ __    _  n
10xxxx1______ 18 18 den fnpa  _  n    ____________ _  _  ___ __    _  n
10xxx0_______ 6  6  den op    _  n    ____________ _  _  ___ __    _  n
10xxx0_______ 16 16 den op    _  n    ____________ _  _  ___ __    _  n
10xxx01______ 14 22 den iop   _  n    ____________ _  _  ___ __    _  n
10xxx011_____ 15 23 den intl  _  n    ____________ _  _  ___ __    _  n
```

Figure 2-53. ARS Digit Analysis Table

Complete the forms as follows:

- **Partitioned Group Number** — N/A for the MCU since partitioning is not used on the MCU. Keep value **1**.
- **Percent Full** — Display-only field. Displays the percentage of the system's memory resources that has been used. A common pool of memory resources is shared between the ARS and AAR Analysis tables.
- **Dialed String** — Enter the dialed string of digits. A maximum of 18 ARS characters consisting of the digits **0** through **9** or ***** are allowed. Use a wildcard letter (**9x** or **X0**) to substitute for any digit occurring in the dialed string at the character position where used.

  NOTE:
  The user-dialed digits are matched to the `Dialed String` entry that most closely matches the dialed number (referred to as the longest match). For example, if a user dials 297-1234 and the AAR or ARS Analysis Table has dialed string entries of 297-1 and 297-123, the match is on the 297-123 entry. In addition, an exact match is made on dialed string entries of the same number of digits in cases where wildcard characters are used. For example, if 957-1212 is dialed and the table has dialed string entries of 957 and X97, the match is on the 957 entry.

- **Min** — Enter the minimum number of digits needed to validate and route the call. Default is blank.
- **Max** — Enter the maximum number of digits to collect, between `Min` and **28**. Default is blank.

  NOTE:
  Keep the ARS Analysis Table as simple as possible. For example, set digit 1 to min 11 max 11; this would take the place of setting digits 120 to min 11 max 11; 121 to min 11 max 11, etc. Setting 1 with min 11 max 11 would allow all 10-digit calls starting with a prefix of '1' for long distance.

- **Rte Pat** (Route Pattern) — Enter the route pattern number to which to send this call, or leave blank (default). For a blank entry or a route pattern that has not been administered, the associated call is denied and intercept treatment is returned. Once this form is administered, route patterns are set up.

  You should administer some sequence of route patterns on the MCU. For example, you might administer a simple network with either all calls going out over one route pattern or a toll call going out on one route pattern and local calls going out on another route pattern, etc.

- **Call Type** (for AAR only) — Enter **aar** for regular AAR calls. Enter **haar** for home AAR calls. Enter **intl** or **pubu** if the administered route pattern contains preferences that are public network ISDN trunks requiring Type of Number (TON) encodings "international" or unknown, respectively. Enter **lev0**, **lev1**, or **lev2** or leave blank (default) to specify Private Numbering Plan (PNP) number formats.
- **Call Type** (for ARS only) — Valid call types/values include the following:

Call Type/ Value	Description	China #1 Call Type
alrt	Alerts attendant console for emergency	normal
emer	emergency call	normal
fnpa	10-digit North American Numbering Plan (NANP) call **NOTE**: There are 11 digits if prefix digit “1” is used.	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 **Note**: This value ignores the *NPA* and *Prefix Mark* fields in the Route Pattern form; therefore, you may need to insert or delete digits.	normal
npvt	national private	normal
nsvc	national service	normal
op	operator	attendant
pubu	international public network number (for example, E.164)	normal
svcl	national (2)	toll-auto
svct	national (2)	normal
svfl		
svft		

For a domestic MCU, enter value **hnpa** for 7-digit calls, value **fnpa** for 10-digit calls, and value **int** for international calls. The remaining call types are generally not used for a domestic MCU.

- **Nd Num** — Identifies the number of the destination node if node number routing or DCS is used.
- **ANI Rq** — Enter **y** if ANI is required on incoming R2-MFC or Russian MF ANI calls. Default is **n**. This field applies only if the *Request Incoming ANI* (non-AAR/ARS) field on the Multifrequency-Signaling-Related System Parameters form is set to **n**.

AAR and ARS Digit Analysis Default Translations

You can delete the defaults by entering **change ARS analysis 0**. Then blank out all of the Dialed Strings with spaces. For service outside of North America, these defaults should be deleted.

ARS Digit Analysis Table

You can list information pertaining to dialed strings and route patterns by issuing the list ars analysis command. An example of the report produced is provided in the following figures

```
list ars analysis                                          Page  1

                          ARS DIGIT ANALYSIS TABLE
                         Partitioned Group Number: 1

    Dialed            Total       Route        Call         Node
    String          Min   Max      Pat         Type        Number
0                    1     1       den         op
0                    8     8       den         op
0                    11    11      den         op
00                   2     2       den         op
01                   9     17      den         iop
011                  10    18      den         intl
101xxxx0             8     8       den         op
101xxxx0             18    18      den         op
101xxxx01            16    24      den         iop
101xxxx011           17    25      den         intl
10xxxx1              18    18      den         fnpa
10xxx0               6     6       den         op
```

Figure 2-54. ARS Digit Analysis Table (Report) — Page 1

```
list ars analysis                                          Page  2

                          ARS DIGIT ANALYSIS TABLE
                          Partitioned Group Number: 1

    Dialed            Total       Route         Call          Node
    String          Min   Max      Pat          Type         Number
10xxx0               16    16      den           op
10xxx01              14    22      den           iop
10xxx011             15    23      den           intl
10xxx1               16    16      den           fnpa
120                  11    11      den           fnpa
1200                 11    11      den           fnpa
121                  11    11      den           fnpa
122                  11    11      den           fnpa
123                  11    11      den           fnpa
124                  11    11      den           fnpa
125                  11    11      den           fnpa
126                  11    11      den           fnpa
```

Figure 2-55. ARS Digit Analysis Table (Report) — Page 2

```
list ars analysis                                          Page  3

                          ARS DIGIT ANALYSIS TABLE
                          Partitioned Group Number: 1

    Dialed            Total       Route         Call          Node
    String          Min   Max      Pat          Type         Number
127                  11    11      den           fnpa
128                  11    11      den           fnpa
129                  11    11      den           fnpa
130                  11    11      den           fnpa
1300                 11    11      den           fnpa
131                  11    11      den           fnpa
132                  11    11      den           fnpa
133                  11    11      den           fnpa
134                  11    11      den           fnpa
135                  11    11      den           fnpa
136                  11    11      den           fnpa
137                  11    11      den           fnpa
```

Figure 2-56. ARS Digit Analysis Table (Report) — Page 3

```
list ars analysis                                          Page  4

                          ARS DIGIT ANALYSIS TABLE
                          Partitioned Group Number: 1

     Dialed            Total       Route         Call          Node
     String          Min    Max     Pat          Type         Number

 138                  11     11      den          fnpa
 139                  11     11      den          fnpa
 140                  11     11      den          fnpa
 1400                 11     11      den          fnpa
 141                  11     11      den          fnpa
 142                  11     11      den          fnpa
 143                  11     11      den          fnpa
 144                  11     11      den          fnpa
 145                  11     11      den          fnpa
 146                  11     11      den          fnpa
 147                  11     11      den          fnpa
 148                  11     11      den          fnpa
```

Figure 2-57. ARS Digit Analysis Table (Report) — Page 4

```
list ars analysis                                          Page  5

                          ARS DIGIT ANALYSIS TABLE
                          Partitioned Group Number: 1

     Dialed            Total       Route         Call          Node
     String          Min    Max     Pat          Type         Number

 149                  11     11      den          fnpa
 150                  11     11      den          fnpa
 1500                 11     11      den          fnpa
 151                  11     11      den          fnpa
 152                  11     11      den          fnpa
 153                  11     11      den          fnpa
 154                  11     11      den          fnpa
 155                  11     11      den          fnpa
 156                  11     11      den          fnpa
 157                  11     11      den          fnpa
 158                  11     11      den          fnpa
 159                  11     11      den          fnpa
```

Figure 2-58. ARS Digit Analysis Table (Report) — Page 5

```
list ars analysis                                          Page  6

                          ARS DIGIT ANALYSIS TABLE
                          Partitioned Group Number: 1

    Dialed            Total       Route         Call          Node
    String          Min   Max      Pat          Type         Number

160                  11    11      den          fnpa
1600                 11    11      den          fnpa
161                  11    11      den          fnpa
162                  11    11      den          fnpa
163                  11    11      den          fnpa
164                  11    11      den          fnpa
165                  11    11      den          fnpa
166                  11    11      den          fnpa
167                  11    11      den          fnpa
168                  11    11      den          fnpa
169                  11    11      den          fnpa
170                  11    11      den          fnpa
```

Figure 2-59. ARS Digit Analysis Table (Report) — Page 6

```
list ars analysis                                          Page  7

                          ARS DIGIT ANALYSIS TABLE
                          Partitioned Group Number: 1

    Dialed            Total       Route         Call          Node
    String          Min   Max      Pat          Type         Number

1700                 11    11      den          fnpa
171                  11    11      den          fnpa
172                  11    11      den          fnpa
173                  11    11      den          fnpa
174                  11    11      den          fnpa
175                  11    11      den          fnpa
176                  11    11      den          fnpa
177                  11    11      den          fnpa
178                  11    11      den          fnpa
179                  11    11      den          fnpa
180                  11    11      den          fnpa
1800                 11    11      den          fnpa
```

Figure 2-60. ARS Digit Analysis Table (Report) — Page 7

```
list ars analysis                                          Page  8

                          ARS DIGIT ANALYSIS TABLE
                          Partitioned Group Number: 1

     Dialed            Total       Route         Call          Node
     String          Min   Max      Pat          Type         Number

 1800555              11    11      den          fnpa
 1809                 11    11      den          fnpa
 181                  11    11      den          fnpa
 182                  11    11      den          fnpa
 183                  11    11      den          fnpa
 184                  11    11      den          fnpa
 185                  11    11      den          fnpa
 186                  11    11      den          fnpa
 187                  11    11      den          fnpa
 188                  11    11      den          fnpa
 189                  11    11      den          fnpa
 190                  11    11      den          fnpa
```

Figure 2-61. ARS Digit Analysis Table (Report) — Page 8

```
list ars analysis                                          Page  9

                          ARS DIGIT ANALYSIS TABLE
                          Partitioned Group Number: 1

     Dialed            Total       Route         Call          Node
     String          Min   Max      Pat          Type         Number

 1900                 11    11      den          fnpa
 1900555              11    11      den          fnpa
 191                  11    11      den          fnpa
 192                  11    11      den          fnpa
 193                  11    11      den          fnpa
 194                  11    11      den          fnpa
 195                  11    11      den          fnpa
 196                  11    11      den          fnpa
 197                  11    11      den          fnpa
 198                  11    11      den          fnpa
 199                  11    11      den          fnpa
 1xxx555              11    11      den          fnpa
```

Figure 2-62. ARS Digit Analysis Table (Report) — Page 9

```
list ars analysis                                          Page  10

                              ARS DIGIT ANALYSIS TABLE
                             Partitioned Group Number: 1

     Dialed            Total       Route         Call          Node
     String          Min   Max      Pat          Type         Number

 1xxx976              11    11      den          fnpa
 2                    7     7       2            hnpa
 3                    7     7       2            hnpa
 4                    7     7       2            hnpa
 411                  3     3       den          svcl
 5                    7     7       2            hnpa
 555                  7     7       den          hnpa
 6                    7     7       2            hnpa
 611                  3     3       1            svcl
 7                    7     7       2            hnpa
 8                    7     7       2            hnpa
 811                  3     3       1            svcl
```

Figure 2-63. ARS Digit Analysis Table (Report) — Page 10

Administering AAR and ARS Digit Conversions

You can convert private-network numbers to other private-network numbers, to public-network ARS numbers, or to extensions in the network. Similarly, you can convert public-network ARS numbers to other public-network numbers, to private-network numbers, or to extensions in the network.

NOTE:
However, whenever AAR is not activated, digits can only be converted from a public network number to another public network number or to a local extension

Accordingly, by using the AAR and ARS Digit Conversion tables, you can do the following:

- Steer some AAR calls to other switches in the private network
- Change the dialed string to a public-network number and route the call via the ARS feature
- Block or intercept certain private-network calls to the attendant group
- Route unauthorized public-network calls to an attendant or give them intercept tone

The form allows for the replacement of all or part of a dialed string with a modified string and/or change of analysis type. The modified string represents an alternative equivalent address and results in call analysis according to the network type. The table is accessed during call-processing.

Digit conversion enhances the capabilities of the AAR and ARS features so that outgoing public-network calls and private-network calls can be changed to local, private, or public network destinations. The calls are then analyzed by the UDP, AAR, or ARS features and routed according to the associated route patterns.

Digit conversion always takes precedence over digit analysis. The conversion process allows all or parts of digits dialed to be modified. The modified string is then further analyzed and routed via analysis or may be further converted.

When the station user dials a collection of digits, the MCU looks for a match in ARS digit conversion and in ARS digit analysis. If a match is found in ARS digit conversion, the MCU does the following:

1. Verifies the minimum and maximum values of the digits received
2. Looks for the best digit string match (route or convert)
3. Looks for the most specific match
4. Deletes the specified number of digits and, if required, replaces these digits with the specified string of digits
5. Determines the network in which to reanalyze the new digits, ARS, AAR, or extension
6. Checks if further conversion is requested (a maximum of seven conversions are permitted)

Use the following commands to administer the AAR and ARS Digit Conversion Table form.

Table 2-38. Commands for Administering AAR and ARS Digit Conversions

Action	Object	Qualifier*
change	aar digit-conversion ars digit-conversion	Enter digits between 0 - 9 ‘*’ or ‘X’
display	aar digit-conversion ars digit-conversion	Enter digits between 0 - 9 ‘*’ or ‘X’ [’print’ or ’schedule’]
list	aar digit-conversion ars digit-conversion	Enter [‘start’ matching pattern][‘count’ (1-400)] [’print’ or ’schedule’]

* Brackets [ ] indicate the qualifier is optional. Single quotes (‘ ’) indicate the text inside the quote must be entered exactly as shown or an abbreviated form of the word may be entered. MAX is the maximum number available in your system configuration.

AAR and ARS Digit Conversion Tables

The forms appear as follows:

```
change aar digit-conversion 0                                Page  1 of  2


                          AAR DIGIT CONVERSION TABLE
                                                        Percent Full: 2

Matching Pattern     Min   Max   Del    Replacement String     Net  Conv ANI Req

_________________    __    __    __     _________________      _    ____    n_
_________________    __    __    __     _________________      _    ____    n_
_________________    __    __    __     _________________      _    ____    n_
_________________    __    __    __     _________________      _    ____    n_
_________________    __    __    __     _________________      _    ____    n_
_________________    __    __    __     _________________      _    ____    n_
_________________    __    __    __     _________________      _    ____    n_
_________________    __    __    __     _________________      _    ____    n_
_________________    __    __    __     _________________      _    ____    n_
_________________    __    __    __     _________________      _    ____    n_
_________________    __    __    __     _________________      _    ____    n_
_________________    __    __    __     _________________      _    ____    n_
_________________    __    __    __     _________________      _    ____    n_
_________________    __    __    __     _________________      _    ____    n_
_________________    __    __    __     _________________      _    ____    n_
_________________    __    __    __     _________________      _    ____    n_
```

Figure 2-64. AAR Digit Conversion Table

```
change ars digit-conversion 0                                Page  1 of  2

                          ARS DIGIT CONVERSION TABLE
                                                          Percent Full: 2
Matching Pattern     Min   Max   Del    Replacement String     Net  Conv ANI Req

_________________    __    __    __     _________________      _    ___     n_
_________________    __    __    __     _________________      _    ___     n_
_________________    __    __    __     _________________      _    ___     n_
_________________    __    __    __     _________________      _    ___     n_
_________________    __    __    __     _________________      _    ___     n_
_________________    __    __    __     _________________      _    ___     n_
_________________    __    __    __     _________________      _    ___     n_
_________________    __    __    __     _________________      _    ___     n_
_________________    __    __    __     _________________      _    ___     n_
_________________    __    __    __     _________________      _    ___     n_
_________________    __    __    __     _________________      _    ___     n_
_________________    __    __    __     _________________      _    ___     n_
_________________    __    __    __     _________________      _    ___     n_
_________________    __    __    __     _________________      _    ___     n_
_________________    __    __    __     _________________      _    ___     n_
_________________    __    __    __     _________________      _    ___     n_
```

Figure 2-65. ARS Digit Conversion Table

NOTE:
When the form is accessed via a **display** or **change** command, the entries are sorted separately in the order of the matching pattern. Specific digits are sorted before the characters “*,” “x,” or “X.”

Complete the forms as follows:

- **Percent Full** — 3-character display-only field. Displays the percentage of the used system’s memory resources. A common pool of memory resources is shared between the ARS and AAR Digit Conversion tables.

  If this field indicates that the system memory resources are in jeopardy of becoming full (100%), perform any of the following to free-up memory resources:

  — Delete unnecessary entries from any of the shared resources tables

  — Reduce `Dialed String` entries to shortest possible length that accomplishes the desired function

- **Matching Pattern** — Enter a one- to 18-digit string to be matched against a dialed number. Matching Pattern entries must be unique within the table. The string may contain the digits 0-9 or a * and wildcard characters “x” or “X.” These are used to match any digit in the range of 0 through 9 or *. If a Prefix Digit is required for 10-digit DDD numbers, then the Prefix Digit "1" must be present in the Matching Pattern string for the match to succeed.

If the associated `Replacement String` has an entry, `Matching Pattern` must also have an entry.

- **Min** — Enter the minimum number of digits in the dialed string.
- **Max** — Enter the maximum number of digits to collect. Value must be between the value for *Min* and value **28**, inclusive.
- **Del** — Enter the number of digits to delete from the beginning of the dialed string.
- **Replacement String —** Enter a 0- to 18-digit dialed string that replaces the deleted portion of the dialed number. The dialed string may contain the digits 0–9 or *, a "#" character, or be blank. The "#" character, when used, must be at the end of the digit-string and is used to indicate end-of-dialing.
- **Net** — Identifies the network in which to analyze the resulting number. Enter value **ext**, **aar**, or **ars**. The resulting digit string is analyzed as an extension number, an AAR address, or an ARS address, respectively.
- **Conv** — Values include **y** and **n**. To allow further conversion, enter value **y**.
- **ANI Req** — Enter **y** if ANI is required on incoming R2-MFC or Russian MF ANI calls. Default is **n**. This field applies only if the *Request Incoming ANI* (non-AAR/ARS) field on the Multifrequency-Signaling-Related System Parameters form is set to **n**.

Administering AAR for MCUs Behind a PBX

The Automatic Alternate Routing (AAR) Analysis form must be changed when the MCU is being connected behind a PBX. Use the form to specify the home AAR code.

Table 2-39. AAR Analysis Administration Commands

Action	Object	Qualifier
change	aar analysis	0-9, x or X (dialed string)
display	aar analysis	0-9, x or X (dialed string)
list	aar analysis	start dialed string, count, rp, part, print or schedule

Use the following procedure to change AAR when the MCU is connected behind a PBX:

1. At the *command:* prompt, type **change aar analysis #** and then press RETURN.
2. Verify that the screen displays the AAR Analysis screen.

3. Use TAB or RETURN to advance to the fields you want to change.
4. When the form is completed, verify that the MT displays: *Command successfully completed* followed by *command:*
5. To verify the form, type **display aar analysis** # and then press RETURN.

AAR Digit Analysis Form

The AAR Digit Analysis form appears as follows:

```
change aar analysis 1                                    Page  1 of  1


                          AAR DIGIT ANALYSIS TABLE
                        Partitioned Group Number: 1    Percent Full:6

     Dialed    Total Rte Call Nd ANI       Dialed    Total Rte Call Nd ANI
     String    Mn Mx Pat Type Num Rq       String    Mn Mx Pat Type NUM Rq
 2_____________ 7_ 7_ 254 aar   _  n     _____________ __ __ ___ ___   _   n
 3_____________ 7_ 7_ 254 aar   _  n     _____________ __ __ ___ ___   _   n
 4_____________ 7_ 7_ 254 aar   _  n     _____________ __ __ ___ ___   _   n
 5_____________ 7_ 7_ 254 aar   _  n     _____________ __ __ ___ ___   _   n
 6_____________ 7_ 7_ 254 aar   _  n     _____________ __ __ ___ ___   _   n
 7_____________ 7_ 7_ 254 aar   _  n     _____________ __ __ ___ ___   _   n
 8_____________ 7_ 7_ 254 aar   _  n     _____________ __ __ ___ ___   _   n
 9_____________ 7_ 7_ 254 aar   _  n     _____________ __ __ ___ ___   _   n
 _____________ __ __ ___ ___   _  n     _____________ __ __ ___ ___   _   n
 _____________ __ __ ___ ___   _  n     _____________ __ __ ___ ___   _   n
 _____________ __ __ ___ ___   _  n     _____________ __ __ ___ ___   _   n
 _____________ __ __ ___ ___   _  n     _____________ __ __ ___ ___   _   n
```

Figure 2-66. AAR Digit Analysis Form

The following list provides the fields on the AAR Analysis form and their related values.

- *Partitioned Group Number*

  The PGN associated with this table for use with corresponding paper forms. Valid entries are **1** to **8**. The default is **1**.

- *Percent Full*

  Displays the percentage of the system's memory resources that have been utilized.

- *Dialed String*

  Enter the dialed string of significant digits. Up to 18 characters (**0** through **9, ***) are allowed. A wildcard character (**x** or **X**) may be used as a substitute for any digit occurring in the string at the character position where used. The default is blank.

- *Min*

  Enter the minimum number of digits needed to validate and route the call. Valid entries are from **1** to the maximum number allowed. The default is blank.

- *Max*

  Enter the maximum number of digits that can be used to validate and route the call. Valid entries are **1** through **28**. The default is blank.

- *Route Pat*

  Enter the routing pattern number (**1** through **254**) on a pointer to an associated RHNPA form (**r1** through **r32**) to be used to route the call once all identifying digits have been received. The default is blank.

- *Call Type*

  Enter **aar** for AAR address to indicate the call will be terminated locally instead of routing to another node. If the location code is deleted from the dialed number, the remaining digits are used to route the call to a local extension. Enter **intl** or **pubu** if the administered route pattern contains preferences that are public network ISDN trunks requiring Type of Number (TON) encodings "international" or unknown, respectively. Enter **lev0**, **lev1**, or **lev2** or leave blank (default) to specify Private Numbering Plan (PNP) number formats.

- *Nd Num*

  Identifies the number of the destination node if node number routing or DCS is used.

- *ANI Rq*

  Enter **y** if ANI is required on incoming R2-MFC or Russian MF ANI calls. Default is **n**. This field applies only if the *Request Incoming ANI* (non-AAR/ARS) field on the Multifrequency-Signaling-Related System Parameters form is set to **n**.

Administering Terminals and Modules

Use the Station form to assign buttons to various terminals and modules.

The following commands can be used to administer various terminals.

Table 2-40. Station Administration Commands

Action	Object	Qualifier
add	station	extension or next
change	station	extension
display	station	extension
list	station	extension, type, count, port, print, or schedule
remove	station	extension

To add a maintenance alarm terminal, follow these procedures:

1. At the *command:* prompt, enter **add station *x*xxx** where ***xxxx*** is an extension number or **next**.
2. Verify that the screen displays the *Station* form (see *Station Form* for a list of fields and values).
3. Use TAB and RETURN to advance to the fields you want to change.
4. Submit the form.
5. To verify the form, enter **display station *xxxx***.

Station Form

Page 1 of the Station form appears as follows:

```
add station next                                          Page 1 of 4

                                    STATION

Extension: 31000
     Type: 7444D                                            COR: 1
     Port: _____________________                            COS: 1
     Name: _____________________



STATION OPTIONS

     Data Module? n
  Display Module? y
```

Figure 2-67. Station Form (Page 1)

The following list shows the fields on page 1 the Station form and their related values.

- *Extension*

  A display-only field that shows the extension number of the terminal being administered.

- *Type*

  This field shows the model terminal assigned to this extension. .

- *COR*

  The Class of Restriction assigned to this terminal. This should be assigned to the same COR as any other terminal.

- *Port*

  Specify the MCU port that is associated with this terminal.

- *COS*

  The Class of Service assigned to this terminal should be the same as the COS assigned to any other terminal.

- *Name*

  A label that identifies this terminal.

- *Data Module*

  When **y** is entered, a 7400B data module must be administered for the terminal.

- *Display Module*

  Set to **y** if the terminal has a display.

Station Form, Page 2 of 5

```
add station next                                          Page 2 of 4

                                    STATION

NON-SWITCH DATA
     Room: _________                                  Headset? n
     Jack: _____                                     Speaker ?  n
    Cable: _____




BUTTON ASSIGNMENTS
     1: call-appr                                6: _________
     2: call-appr                                7: _________
     3: call-appr                                8: _________
     4: _________                                9: _________
     5: _________                               10: _________
```

Figure 2-68. Station Form (Page 2)

- *Room*

  Specify the room number where this maintenance alarm terminal is located.

- *Headset*

  This field is not used. Leave it set to **n**.

- *Jack*

  Specify the jack number where the terminal is connected.

- *Cable*

  Enter the cable number that connects the jack to the MCU.

- *Button Assignments*

  Assign the desired features on the appropriate buttons. The following table lists the recommended terminal button assignments for buttons 1 through 10.

Table 2-41. Terminal Button Assignments (1 through 10)

Button	Assignment	Button	Assignment
1	call-appr	6	ac-alarm
2	call-appr	7	cdr1-alrm
3	call-appr	8	pr-sys-alm
4	lsvn-call	9	trk-ac-alm
5	aca-call	10	verify

Station Form, Page 3

```
add station next                                          Page 3 of 4

                                    STATION


BUTTON ASSIGNMENTS
  11: _________                      12: _________

  13: _________                      24: _________
  14: _________                      25: _________
  15: _________                      26: _________
  16: _________                      27: _________
  17: _________                      28: _________
  18: _________                      29: _________
  19: _________                      30: _________
  20: _________                      31: _________
  21: _________                      32: _________
  22: _________                      33: _________
  23: _________                      34: _________
```

Figure 2-69. Station Form (Page 3)

- *Button Assignments*

  Use this continuation of screen 2 to assign features to the appropriate buttons. The following table lists the recommended terminal button assignments for buttons 11 through 34.

Table 2-42. Terminal Button Assignments (11 through 34)

Button	Assignment	Button	Assignment
11		23	
12		24	link-alarm Lnk #: 1
13	major-alrm	25	link-alarm Lnk #: 2
14	minor-alrm	26	link-alarm Lnk #: 3
15	warn-alrm	27	link-alarm Lnk #: 4
16	mmi-cp-alm	28	
17	vc-cp-alm	29	
18	ds1-alarm	30	
19		31	
20		32	
21		33	
22		34	

Station Form, Page 4

```
add station next                                        Page 4 of 4

                                    STATION

DISPLAY BUTTON ASSIGNMENTS

1: _________
2: _________
3: _________
4: _________
5: _________
6: _________
7: _________
```

Figure 2-70. Station Form (Page 4)

- *Display Button Assignments*

  Assign the display button appearances to the appropriate buttons as shown in the following table.

Table 2-43. Display Button Assignments

Button	Assignment
1	normal
2	inspect
3	trk-id
4	trunk-name
5	date-time
6	timer
7	

Station Form, Page 5

```
add station next                                          Page 5 of 5

                                    STATION

DATA MODULE
     Data Extension: ____                 BCC: 2        ITC: restricted
               Name: _________________    COR: 1        COS: 1









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

Figure 2-71. Station Form (Page 5)

NOTE:
This screen displays only when **y** is entered in the *Data Module* field on the first screen.

- *Data Extension*

  Enter the extension assigned to the data module.
- *BCC*

  Leave this field set to **2.**
- *ITC*

  Information Transfer Capability used for routing calls according to the endpoint's ITC. Default is **restricted**.
- *Name*

  Optional name for identification.
- *COR*

  The Class of Restriction assigned to this extension.
- *COS*

  The Class of Service assigned to this extension.
- *Ext Name*

  The terminal with a data extension button for this data module.

Administering Routing Patterns

Route patterns direct outgoing calls to a trunk group so that calls can route over a trunk that terminates at switches with different dial plans on a pattern and preference basis. More than one combination of dialed digits can point to the same route pattern. Available trunks can be selected, on a priority basis, as alternate facilities to route the call.

The digit translations performed by the AAR Digit Analysis Table and by the ARS Digit Analysis Table on a public destination number cause a specific route pattern to be selected for routing the call. The route pattern can contain a sequence of trunk groups on which an attempt is made to route the call.

The route patterns specified on the Route Pattern form are used by the following features: AAR, ARS, GRS, Call-by-Call (CBC) Service Selection, and Facility Restriction Level (FRL).

GRS is an enhancement to AAR/ARS. GRS performs routing based on call attributes, such as Bearer Capability Classes (BCCs). CBC allows you to reach multiple services via the same ISDN B-channel. This eliminates the need to dedicate each trunk or channel to a specific service.

Before a call is routed, the MCU checks the following:

- Facility Restriction Level (FRL) to ensure that the calling party's FRL is greater than or equal to the routing preference's FRL
- Calling party's BCC matches the BCC specified for the routing preference (ISDN calls only)
- Facility type is okay for the call type
- There is an unused circuit (trunk port) available in the selected trunk group

Call-by-Call Service Selection uses the same route patterns and route preferences that are used by AAR, ARS, and GRS. The service or facility used on an outgoing CBC Service Selection call is determined by information assigned in the AAR/ARS/GRS route patterns.

You can allow a variety of services to use a single trunk group. The system obtains trunking efficiency by distributing traffic over all the available trunks. You can then assign services that are used on incoming and outgoing CBC Service Selection calls.

NOTE:
Whenever the MCU is connected to a 5ESS-type central office (CO), all of the services that are supported under the Lucent Technologies and NI-2 country options are available.

Each route pattern has six preferences available for each of 254 route patterns on the MCU. ARS and AAR share route patterns.

To track how a call is routed, issue the **list ARS route-chosen** command. This command displays the minimum and maximum digits dialed, route pattern, and the call type that the MCU software will use for a specific dialed number.

When all checks are completed and compatibility is ensured, digits are sent over the network to connect the call.

Use the following commands to administer the Route Pattern form

Table 2-44. Routing Pattern Commands

Action	Object	Qualifier*
change	route-pattern	1-254
display	route-pattern	1-254 [print]
list	route-pattern	Enter [‘trunk’ (1-99)] [‘service’/feature name string] [‘print‘ or ‘schedule’]

* Brackets [ ] indicate the qualifier is optional. Single quotes (‘ ’) indicate the text inside the quote must be entered exactly as shown or an abbreviated form of the word may be entered. MAX is the maximum number available in your system configuration.

Route Pattern Form

The Route Pattern form implements one route pattern. Each route pattern can contain alternate trunk groups.

The Route Pattern form appears as follows:

```
change route-pattern 1                                    Page  1 of  1

                           Pattern Number: 1

   Grp.  FRL NPA Pfx      Toll No. Del Inserted                      IXC
   No.           Mrk      List Digits  Digits
1: 11_    3  732  1        __   _     9_____________________________ user
2: 12_    3  732  1        __   _     9_____________________________ user
3: ___    _  ___  _        __   _     ______________________________ user
4: ___    _  ___  _        __   _     ______________________________ user
5: ___    _  ___  _        __   _     ______________________________ user
6: ___    _  ___  _        __   _     ______________________________ user

    BCC VALUE  TSC         ITC   BCIE Service/Feature        Numbering
   0 1 2 3 4 W                                               Format
1: y y y y y n  n          rest       _________________       _________
2: y y n n n n  n          rest       _________________       _________
3: y y y y y n  n          rest       _________________       _________
4: y y y y y n  n          rest       _________________       _________
5: y y y y y n  n          rest       _________________       _________
6: y y y y y n  n          rest       _________________       _________
```

Figure 2-72. Route Pattern Form (Page 1)

Make assignments as required for the following fields on the form:

- **Pattern Number** — This is a display-only field when the form is accessed using an administration command such as **change** or **display**.
- **Grp No.** — Enter the desired trunk group number to specify which trunk-group will be associated with the entries on this line of the form.
- **FRL** — Enter the FRL (**0** through **7**) to be assigned this routing preference (this line of parameters on the form). Value **0** is the least restrictive and value **7** is the most restrictive. Trunk group FRLs are changeable from pattern to pattern. The calling party's FRL must be greater than or equal to this FRL to access the associated trunk-group.
- **NPA**— This entry is not required for AAR. For ARS, enter the NPA of the distant end (terminating endpoint). For tie trunks, the NPA field is left blank. Values include three digits in the form: first digit (**2**–**9**), second digit (**0**–**9**), third digit (**0**– **9**). Default is blank.

- **Pfx Mrk** — Because some, but not all, calls need to outpulse digit “1,” the route pattern must distinguish between which of these routing preferences will outpulse digit “1” and which will not outpulse this digit. This field is used for this purpose. In general, this field applies only to the hnpa and fnpa call types. Values include **0** through **4** and blank (default).

  NOTE:
  However, prefix marks **2** and **3** *are not used on the MCU* because it is easier to insert digits on 7-digit toll calls then it is to populate the toll tables. Use only prefix marks **0**, **1**, and **4**.

  0 Suppresses a user-dialed digit “1” for 10-digit FNPA calls and sends a user-dialed digit “1” for 7-digit hnpa calls.

  1 Outpulses a “1” for 10-digit calls and does not outpulse a "1" for 7-digits calls. Use prefix mark “1” whenever HNPA calls do not require dialing digit “1” and a user dials a “1” on HNPA calls.

  2 N/A for the MCU. Outpulses a “1” for all toll calls including 10 digits or seven digits, whether or not the user dials a “1.” Typically, 7-digit hnpa toll calls are indicated as toll calls on the ARS toll table. If the digits are toll digits, the digit “1” is sent.

  3 N/A for the MCU. Outpulses a “1” for all toll calls and keeps or inserts the NPA to ensure that all toll calls are ten digits. (Use digit insertion on MCU.)

  4 Always suppresses a user-dialed prefix digit “1.” For example, an SDN/MEGACOM call could go directly to a 4ESS (toll switcher), and the 4ESS would reject this call because it starts with “1.”

  Default value blank provides the same treatment as prefix mark **0**.

- **Toll List** — N/A for the MCU. Leave blank.

- **No. Del. Digits** — Enter the total number of digits to be deleted from the dialed string when selecting this trunk group for call routing. Default is blank.

  Any nonblank value in this field (including **0**) causes any user-dialed IXC code to be deleted in addition to the number of digits specified.

  NOTE:
  Use the *Prefix Mark* field on this form to suppress toll prefixes following user-dialed IXC codes. This capability is useful when the call is routed via ISDN-PRI trunks to a 4ESS office. The 4ESS will reject a call if the IXC code is sent along with the address digits.

- **Inserted Digits** — Enter the actual digits to be inserted. A maximum of 52 digits may be outpulsed. This includes 36 digits that you enter here plus up to 18 digits that were originally dialed. When they are used, special symbols count as two digits each. The special characters used in route patterns are described in the following table.

Special Character	Description
*	CO or PBX feature
#	Signals end of dialing
,	1.5 second pause
+	Wait for dial tone up to the Off Premises Tone Detection Timer and then either outpulse or intercept tone based on `Out Pulse Without Tone y/n` (Feature-Related System Parameters form). Outpulsed DTMF digits are sent at the rate specified by the End-to-End Signaling Tone and Pause parameters (Trunk Group Administrable Timers form).
%	Start End-to-End Signaling. Send digits as DTMF inband tones at the rate specified by the End-to-End Signaling tone and Pause parameters (Trunk Group Administrable Timers form).

- **IXC** — Displays when `ISDN-PRI` is enabled on the Customer-Options form. For ISDN Preference trunks, enter the last two to four digits to represent an equal access inter-exchange carrier (IXC) code number (such as 10XX or 10XXX).

  NOTE:
  This length should match the format established on the IXC Codes form.

  This field is used by all calls that route via an IXC; it is also used for Call Detail Recording (CDR). If an IXC is not specified, a call is transmitted by the presubscribed common carrier.

  NOTE:
  The IXC 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, then enter this IXC value into the `Inserted Digits` field.

- **BCC Value** (required by GRS feature) — Displays when `ISDN-PRI` is enabled on the Customer-Options form. Enter **y** (default) or **n** in the appropriate BCC column (0, 1, 2, 3, 4, or W) to specify whether the BCC is valid for the associated route. The following defines the BCC values:

BCC Value	Description
0	Voice-Grade Data and Voice
1	56k Data (Mode 1)
2	64k Data (Mode 2)
3	64k Data (Mode 3)
4	64k Data (Mode 0)
W	128k to 984k Data (Wideband)

- **TSC** — N/A for the MCU. Leave set to value **n**.
- **ITC** (Information Transfer Capability) — Used to administer the type of traffic (restricted, unrestricted, or both) that is allowed to be carried by this routing preference. For value **rest** (restricted), only calls originating from restricted endpoints may access this route pattern. For value **unre** (unrestricted), only calls originating from unrestricted endpoints may access this route pattern. For value **both**, calls originating from both restricted and unrestricted endpoints may access the route pattern.

  NOTE:
  This field must be set to **unre** or **both** if `BCC W` is set to **y**.

- **BCIE** (Bearer Capability Information Element) — Used to determine how to create the ITC codepoint in the BCIE of the setup message. This field applies only to ISDN trunks and is displayed and administrable only if `ITC` is set to **both**. Values include **ept** (endpoint) and **unr** (unrestricted). Default is **ept**.
- **Service/Feature** (required by the Call-by-Call Service Selection feature) — Displays when `ISDN-PRI` is enabled on the Customer-Options form. Enter up to 15 characters that represent the Service/Feature carried by a call in this route pattern. Default is blank. Values include **accunet**, **i800**, **inwats**, **lds**, **mega800**, **megacom**, **multiquest**, **operator**, **outwats-bnd**, **sdn**, **sub-operator**, and **wats-max-bnd**.
- **Numbering Format** — For a domestic MCU, leave blank.
- **LAR** — N/A for the MCU. Leave value **none**.

Administering Alphanumeric Dialing

This feature allows a system administrator to change a mapped string (the actual digit dialing string) to an alpha name associated with the string. It is used to replace extension numbers with easier to remember alpha characters. For example, you could replace the netcon hunt group extension assigned for remote administration with the alpha name **CRCS**.

The Alphanumeric Dialing Table form consists of 68 paired alpha name and/or mapped string fields. Entries may be made in any order on the form; however, before the form is displayed for changing or reviewing, the entries are sorted alphanumerically by alpha name. All entries are moved to the beginning of the table, leaving any blank entries at the end.

The following administration commands access the Alphanumeric Dialing Table form.

Table 2-45. Station Administration Commands

Action	Object	Qualifier
change	alphanumeric-dial-table	
display	alphanumeric-dial-table	print

Follow the procedure below to use the alphanumeric dialing feature:

1. At the *command:* prompt, type **change alphanumeric-dial-table** and then press RETURN.
2. Verify that the screen displays the Alphanumeric Dialing Table form.
3. From the *Alpha-name* field, enter the first alpha name associated with a mapped string (see below for details). Use TAB or RETURN to advance to the next field.
4. When the form is completed, verify that the MT displays: *Command successfully completed* followed by *command:*
5. To verify the form, type **display alphanumeric-dial-table** and then press RETURN.

Alphanumeric Dialing Table Form

The following list shows the fields on the Alphanumeric Dialing Table form and their related values.

Alphanumeric Dialing Table Form, Page 1 through 6 of 6

- *Alpha-Name*

  This field accepts up to 8 alphanumeric characters that must begin with an alpha character and may not have any blank spaces between characters. 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. The default is blank.

- *Mapped String*

  Enter from 1 to 24 characters containing alphanumeric, readability, delimiters and/or special characters. This entry is used to generate the final dialing string and can include Feature Access Codes. Numbers 0 through 9, letters a through z (uppercase letters are mapped to lowercase), and character delimiters (such as / and -) are permitted. A mapped string may not contain an alpha name whose mapped string also contains an alpha name.

Administering Video Modes

The Lucent Technologies MCU supports the following standard video modes:

- H.261 and its variations, including H.261-down_only, H.261-w/upgrade, H.261-CIF-Floor, H.261-CIF-Now, and H.261-QCIF-Now
- H.263 and its variations (which involve H.263/H.261 transcoding), including H.263/1-down_only and H.263/1-w/upgrade

The MCU also supports the following proprietary video modes:

- H.CTX/H.CTX+
- SG4
- MRV-w/upgrade

The video modes in the previous lists are administered via the *Video Mode* field in the Conference Record form, which is discussed in Chapter 1 of this document. Refer to the “Video Mode” bullet item within the “Conference Record Form” section for details on administering video modes.

As suggested previously, the MCU supports H.263/H.261 transcoding. For complete details on administering this feature, refer to the “Administering H.263/H.261 Transcoding” section in this chapter.

NOTE:
Refer to the appropriate sections in Appendix B of this document for a complete discussion of the following video modes, including feature interactions: H.263 video mode, H.263/H.261 transcoding, H.CTX/H.CTX+, MultiRate Video (MRV), and SG4.

In addition, be sure to refer to the “Interactions” sections in Appendix B for the following features, for which there is a discussion of one or more video mode interactions: Continuous Presence *Plus*, On-Screen Display, Speed Match, T.120 Data Conferencing, and Universal Conference Control.

Administering Cascades

Cascading is the linking of two MCUs to allow both MCUs to participate in the same conference. One MCU can be programmed to set up a link with another MCU by populating the appropriate fields in several forms. The resulting MCU-MCU link is called a cascade link. Once the cascade link is established, it is maintained by one of the MCUs in the cascaded MCU pair for the duration of the conference.

To enable cascading, one MCU must be established as the dial-out MCU and the other MCU as the dial-in MCU. Also, one MCU must be established as the primary MCU and the other MCU as the secondary MCU.

NOTE:
The types attributed to an MCU (dial-out or dial-in and primary or secondary) are not MCU-specific, but conference-specific. In other words, an MCU is assigned the appropriate types according to the requirements of each conference. Therefore, for example, an MCU might be a dial-out MCU in one conference but a dial-in MCU in another conference.

To administer cascading, complete the fields in the forms identified in the following table. Appropriate page references to the forms are provided. Detailed instructions for completing the forms are provided after the table

Table 2-46. Required Forms and Fields for Cascading

Form	Field	Page Reference
Customer-Options	■ *Cascading*	page 2-22
Conference Record	■ *Cascade Mode* ■ *Type* ■ *Dial-out #1* ■ *Dial-out #2* ■ *MCU-Ext*	page 1-3

Customer-Options Form

Access the Customer-Options form and enter **y** into the *Cascading?* field.

Conference Record Form

Access the Conference Record form to populate the required fields for cascading, as discussed in the following sections.

Establishing the Primary and Secondary MCUs

In a cascaded conference, one MCU is administered as the primary MCU and the other as the secondary MCU. This is done to make the MCU feature interactions as transparent as possible to the user. Generally, this administration can be performed to accommodate the requirements at the customer site.

To establish a primary MCU and a secondary MCU, do the following:

1. Access the Conference Record form for the MCU that is to serve as the primary MCU, enter value **primary** into the *Cascade Mode* field on page 1, and submit the form.
2. Access the Conference Record form for the MCU that is to serve as the secondary MCU, enter value **secondary** into the *Cascade Mode* field on page 1, and submit the form.

Procedure for Cascaded Conferences in Broadcast with Auto-Scan or Presentation Mode

Always ensure that an H.320 endpoint connects to the *primary* MCU if the endpoint is one of the following:

- Broadcaster in broadcast with auto-scan mode
- Presenter in presentation mode

With this setup, the secondary MCU must be configured by accessing the Conference Record form for this MCU, typing the MCU-extension of the cascade link in the *Control Ext* field on page 1 of the form, and then submitting the form.

Establishing the Dial-Out and Dial-In MCUs

At the start of a cascaded conference, one MCU launches a call (that is, dials out) to another MCU to establish a cascade link. The MCU that launches the call is called the dial-out MCU. The MCU that receives the call is called the dial-in MCU. Both MCU types can be established by populating the appropriate fields on page 2 of the Conference Record form for each MCU.

To establish the dial-out MCU, do the following:

1. Access the Conference Record form for the MCU that is to serve as the dial-out MCU.
2. On page 2 of the form, enter value **CAS** (cascade) or **BCAS** (BONDed cascade) into the first *Type* field (for Conferee 1).
3. In the *Dial-out # 1* field for Conferee 1, enter a number that reaches the dial-in MCU. If this is a 2B conference, enter the same number into the corresponding *Dial-out # 2* field**.**

The following figure is an example of page 2 of a completed Conference Record form for a dial-out MCU:

```
change conference 123                                    page  2 of 8

                                CONFERENCE RECORD
CONFEREES
 1  Type: CAS_  MCU-Ext: 5000_  Rate: _____   Meet-Me Number: xxxxxxxxxxxxxxx
(Re)dial? _    App Comp? _  Basic/Enh: basic___    Site Name: Site 1_________
     Dial Out #1: 705-555-1234_________  Dial Out #2: 705-555-1234_________
   Password Reqd? y

 2  Type: P64_  MCU-Ext: 5001_  Rate: _____   Meet-Me Number: xxxxxxxxxxxxxxx
(Re)dial? _    App Comp? _  Basic/Enh: basic___    Site Name: Site 2_________
     Dial Out #1: _____________________  Dial Out #2: _____________________
   Password Reqd? y

 3  Type: P64_  MCU-Ext: 5002_  Rate: _____   Meet-Me Number: xxxxxxxxxxxxxxx
(Re)dial? _    App Comp? _  Basic/Enh: basic___    Site Name: Site 3_________
     Dial Out #1: _____________________  Dial Out #2: _____________________
   Password Reqd? y

 4  Type: P64_  MCU-Ext: 5003_  Rate: _____   Meet-Me Number: xxxxxxxxxxxxxxx
(Re)dial? _    App Comp? _  Basic/Enh: basic___    Site Name: Site 4_________
     Dial Out #1: _____________________  Dial Out #2: _____________________
   Password Reqd? y
```

Figure 2-73. Conference Record Form for a Dial-Out MCU (Page 2)

4. Save the Conference Record form.

NOTE:
If you access the Conference Record form by entering the **add conference** command (as is appropriate for administering cascading), the *Meet-Me Number* field is blank. However, if you access the form by entering the **change conference** command, and if the Meet-Me Number(s) corresponding to the appropriate MCU-extension are administered on the MCU-Extension form, the Meet-Me Number(s) appear(s) within the *Meet-Me Number* field.

To establish the dial-in MCU, do the following:

1. Access the Conference Record form for the MCU that is to serve as the dial-in MCU.
2. On Page 2 of the form, enter value **CAS** (cascade) or **BCAS** (BONDed cascade) into the first *Type* field (for Conferee 1).
3. In the *MCU-Ext* field for Conferee 1, enter the extension (usually the last four digits) of one of the CAS- or BCAS-type dial-out numbers assigned to the corresponding dial-out MCU.

The following figure is an example of page 2 of a completed Conference Record form for a dial-in MCU:

```
change conference 123                                    page  2 of 8

                                CONFERENCE RECORD
CONFEREES
 1  Type: CAS_  MCU-Ext: 1234_  Rate: _____   Meet-Me Number: xxxxxxxxxxxxxxx
(Re)dial? _    App Comp? _  Basic/Enh: basic___    Site Name: Site 1_________
     Dial Out #1: _____________________  Dial Out #2: _____________________
   Password Reqd? y

 2  Type: P64_  MCU-Ext: 6001_  Rate: _____   Meet-Me Number: xxxxxxxxxxxxxxx
(Re)dial? _    App Comp? _  Basic/Enh: basic___    Site Name: Site 2_________
     Dial Out #1: _____________________  Dial Out #2: _____________________
   Password Reqd? y

 3  Type: P64_  MCU-Ext: 6002_  Rate: _____   Meet-Me Number: xxxxxxxxxxxxxxx
(Re)dial? _    App Comp? _  Basic/Enh: basic___    Site Name: Site 3_________
     Dial Out #1: _____________________  Dial Out #2: _____________________
   Password Reqd? y

 4  Type: P64_  MCU-Ext: 6003_  Rate: _____   Meet-Me Number: xxxxxxxxxxxxxxx
(Re)dial? _    App Comp? _  Basic/Enh: basic___    Site Name: Site 4_________
     Dial Out #1: _____________________  Dial Out #2: _____________________
   Password Reqd? y
```

Figure 2-74. Conference Record Form for a Dial-In MCU (Page 2)

4. Save the Conference Record form.

Administering Star Cascading

Star Cascading allows a maximum of 25 cascaded endpoints per MCU, as opposed to one such endpoint in previous releases. As such, this feature merely requires the administration of additional cascaded endpoints within the Conference Record form. All other administration is identical to conventional Cascading.

Administering Access Endpoints

The Access Endpoints feature allows endpoints to participate in multipoint conferences via the following:

- Non-signaled DS1 (T1 or E1) facilities (DS1 interface option)
- EIA-530, RS449, and V.35 data interfaces (data interface option)

The port designated for each access endpoint is dedicated to that endpoint.

The MCU can use Non-Signaled T1 or E1 facilities to connect to a multiplexer (MUX), a digital Access Crosspoint System (DACS), or to H.320 endpoints via DS1 facilities. A maximum of 25 DS1 connections of all types to the MCU are supported.

The MCU can also use an EIA-530, RS449, or V.35 data interface along with RS366 dialing to connect to an H.320, H.CTX, or SG4 codec. Each of these data interfaces (DIs) allows endpoints that are either connected directly to the MCU or behind an encryption device or proprietary inverse multiplexor to participate in multipoint conferences at a rate in the range of 112k to 1920k. RS366 dialing can also use any of these data interfaces to enable H.323 endpoints connected behind H.323 gateways to participate in MCU conferences.

The MCU requires administration of access endpoint forms for both DI and DS1 access endpoints. Additionally, DS1-type access endpoints require administration of an associated DS1 form prior to the administration of the access endpoint.

To administer Access Endpoints, complete the fields in the forms identified in the following table. Appropriate page references to the forms are provided. Detailed instructions for completing the forms are provided after the table

Table 2-47. Required Forms and Fields for Access Endpoints

Form	Field	Page Reference
DS1 Circuit Pack (DS1 interface option only)	■ All	page 2-56
Customer-Options (DI interface option only)	■ *DI Direct Connect*	page 2-30
Conference Record	■ *Type*	page 1-3
Access Endpoint	■ (See the following “Access Endpoint Form” section.)	page 2-198 (DI) page 2-203 (DS1)
Synchronization Plan	■ (See the following “Synchronization Plan Form” section.)	page 2-207 (DI) page 2-210 (DS1)

DS1 Circuit Pack Form

To administer the DS1 interface option only, refer to page page 2-56 for instructions on populating the DS1 Circuit Pack form.

Conference Record Form

Access the Conference Record Form. The *Type* field on page 2 of the Conference Record form supports two types of access endpoints: DS1 endpoints and data interface (DI) endpoints. Ensure that the *Type* field contains **DA**, unless the access endpoint supports RS366 dialing. In such a case, valid field values include **P64** and **DA**.

Access Endpoint Form

From the console, bring up the Access Endpoint form by entering either **add access-endpoint *xxxx*** or **change access-endpoint *xxxx***, where ***xxxx*** is the endpoint extension.

Data Interface Option

To administer an EIA-530, RS449, or V.35 interface, complete the steps in this section. A sample form for administering an RS449 data interface appears as follows:

```
change access-endpoint 21010                          Page  1 of  1

                              ACCESS ENDPOINT

        Extension: 21010                          Name: __________
   Interface Type: RS449__                (Starting) Port: ______
   Bandwidth Type: maximum                 Bandwidth Rate: 56k__
              COR: 1_                                 COS: 1_
MMI Cabling Board: ____
      Timing Mode: DCE_________                DSR (DCE): during-conference
        DCD (DCE): during-conference           CTS (DCE): during-conference
        DTR (DTE): during-conference           RTS (DTE): during-conference
Encryption Resync: KIV-7___                  Resync Interval: 8_




   RS-366 Dialing? _

      Send Timing: standard
   Receive Timing: standard
  Terminal Timing: standard                   Line Loop Test: off
```

Figure 2-75. Access Endpoint Form (RS449 Interface)

1. Once the Access Endpoint form appears, complete (where necessary) the following fields:

   - *Extension*

     Display-only field that shows the extension number specified in the command line. This extension number is used to access the access endpoint.

   - *Name*

     Enter up to 11 alphanumeric characters to identify the endpoint. Default is blank.

   - *Interface Type*

     Defines the type of interface to be administered for the current access endpoint. Enter **V.35**, **RS449**, or **EIA-530**.

NOTE:
Value **DS1** (default) is used to administer non-signaled DS1 facilities.

Value **V.35**, **RS449**, or **EIA-530** turns off the following form fields: *Communication Type*, *Width*, and *ITC*. Also, the value turns on the following fields: *Bandwidth Type*, *Bandwidth Rate*, *Timing Mode*, *DSR (DCE)*, *MMI Cabling Board*, and *DCD (DCE)*. Values **EIA-530** and **RS449** turn on the *Encryption Resync* field. For a change operation, the interface type must not already be in use on a conference call, and it must not already be administered on the System Synchronization plan.

- *(Starting) Port*

  Indicates the location of the circuit pack and the circuit to which the endpoint will be connected.

  Enter the appropriate seven values according to the breakdown in the following table.

Character Position	Meaning	Value
1-2	Network number	01
3	Carrier	A through E
4-5	Slot number	01 through 20
6-7	Circuit Number	01

NOTE:
The associated endpoint must not already be in use. Also, the associated circuit pack must be installed or administered, and it must correspond to the interface type.

For a change operation, the port must not already be administered on the System Synchronization plan.

- *Bandwidth Type*

  Identifies how the rate provided in the *Bandwidth Rate* field is to be used. Values include **maximum** (default) and **fixed**.

NOTE:
For a change operation, the associated endpoint must not already be in use on a conference call, and it must not already be administered on the System Synchronization plan.

- *Bandwidth Rate*

  Indicates the maximum bandwidth speed. (Endpoints can operate at a lower bandwidth.) Default is **56k**.

  NOTE:
  For a change operation, the associated endpoint must not already be in use on a conference call, and it must not already be administered on the System Synchronization plan.

- *COR*

  For the Class of Restriction, leave at default value **1**.

- *COS*

  For the Class of Service, leave at default value **1**.

- *MMI Cabling Board*

  This field identifies the Multimedia Interface (MMI) circuit pack to which this board is cabled. Enter the 5-character board number according to the breakdown in the following table.

Character Position	Meaning	Value
1-2	Network number	01
3	Carrier	A through E
4-5	Slot number	01 through 20

  NOTE:
  For rate 672k or greater, the MMI Cabling Board must be installed and administered, and it must not already be in use.

  For a change operation, the DI circuit pack must not already be administered on the System Synchronization plan.

- *Timing Mode*

  Indicates how the MCU will operate [as either Data Communication Equipment (DCE) or Data Terminal Equipment (DTE)] for the given port and also whether the clocking is supplied internally or externally from the MCU. Values include **DCE**, **DCE-Extclock**, **DCE-Extended**, **DCE-TA**, **DTE** (default), and **DTE-Intclock**. Value **DCE-TA** drops the clock rate to 56k whenever the call is not up. For connections to a port on an IMUX, the field contains **DTE**. For connections to a port on a codec, the field contains **DCE** or **DCE-TA**. For connections to a port on an H.323 gateway, the field contains **DCE**.

The DCE side of the interface provides the synchronous transmit and receive clock signals to the DTE.

The MCU system clock is based on the active synchronization source. Whenever the MCU is configured as a DCE, the MCU supplies the clock signals based on timing derived from the MCU's system clock. In DCE external clock mode, the MCU system clock is used to provide the receive timing signal and to clock the received data. Also, the port accepts the transmit clock external (TT) from the connected equipment for received data timing. This setting prevents data loss due to phase shifts between the MCU and the DTE. The MCU should be configured as a DCE whenever video conferencing endpoints are directly connected to the MCU via the V.35 interface.

Whenever the MCU is configured as a DTE, the MCU accepts both the transmit and receive clock signals from the connected DCE. In DTE internal clock mode, the received data is timed on the receive timing signal, but the output data is timed from the MCU system clock.

The MCU is usually configured as a DTE whenever it is connected to a KG encryption device or to a terminal adapter.

NOTE:
For a change operation, the timing mode must not already be in use on a conference call.

- *DSR (DCE)*

  Indicates whether DSR (DCE) is either on during the conference or always on. Values include **during-conference** (default) and **always-on**. For Encryption Resync applications, enter **always-on**.

- *DCD (DCE)*

  Indicates whether DCD (DCE) is either on only during the conference or always on. Values include **during-conference** (default) and **always-on**. For Encryption Resync applications, enter **always-on**.

- *CTS (DCE)*

  Indicates whether CTS (DCE) is either on during the conference or always on. Values include **during-conference** (default) and **always-on**. For Encryption Resync applications, enter **always-on**.

- *DTR (DTE)*

  Indicates whether DTR (DTE) is either on during the conference or always on. Values include **during-conference** (default) and **always-on**. For Encryption Resync applications, enter **always-on**.

- *RTS (DTE)*

  Indicates whether RTS (DTE) is either on during the conference or always on. Values include **during-conference** (default) and **always-on**. For Encryption Resync applications, enter **always-on**.

- *Encryption Resync*

  Indicates whether encryption will be used on the associated port to enable loss of framing controls. Values include **off** (default), **KG-level**, **KG-pulse**, and **KIV-7**.

  NOTE:
  This field is not displayed unless the *Interface Type* field is set to **EIA-530** or **RS449**.

- *Resync Interval*

  Indicates the time in seconds (**0** through **15**) needed by the exterior device to resync with the MCU after the MCU sends a resync pulse. This is intended to prevent sending a second resync pulse before the exterior system has time to resync with the MCU. Default is value **8**.

  NOTE:
  This field appears only if the *Encryption Resync* field contains **KG-Pulse** or **KIV-7**.

- *RS-366 Dialing*

  Enter **y** to enable RS-366 Dialing or **n** to disable it. Default is blank.

- *Send Timing*

  Controls the timing detection protocol. Values include **standard** and **inverted**. (These values indicate detection modes.)

- *Receive Timing*

  Refer to the description for *Send Timing*.

- *Terminal Timing*

  Refer to the description for *Send Timing*.

- *Line Loop Test*

  Enables a loopback test. Value include **off** and **on**. For value **on**, a loopback occurs on both sides of the DI circuit pack, within the internal part of the TDM bus, and back to the endpoint. This results in isolating the DI circuit pack for trouble detection other than on the DI circuit pack.

  NOTE:
  This field is always turned off during a translation load (reset system 3 or greater).

DS1 Interface Option

To administer a DS1 interface, complete the steps in this section. A sample form appears as follows:

```
change access-endpoint 21005                          Page 1 of 1

                              ACCESS ENDPOINT

         Extension: 21005                              Name: just_4show
    Interface Type: DS1                    (Starting) Port: 01A1618
Communication Type: wideband________                  Width: 6
               COR: 1                                   COS: 1
               ITC: unrestricted
```

Figure 2-76. Access Endpoint Form (DS1 Interface)

1. Once the form appears, complete (where necessary) the following fields:

   - *Extension*

     Display-only field that shows the extension number specified in the command line. This extension number is used to access the access endpoint.

   - *Name*

     Enter up to 11 alphanumeric characters to identify the endpoint (for example, **Access-Endpoint #1**). Default is blank.

   - *Interface Type*

     Defines the type of interface to be administered for the current access endpoint. Enter **DS1** (default).

   - *(Starting) Port*

     Indicates the location of the circuit pack and the circuit to which the endpoint will be connected.

Enter the appropriate seven values according to the breakdown in the following table.

Character Position	Meaning	Value
1-2	Network number	01
3	Carrier	A through E
4-5	Slot number	01 through 20
6-7	Circuit Number	01 through 31

NOTE:
The port must not already be in use. Also, the associated circuit pack must be installed or administered, and it must correspond to the interface type.

For a change operation, the port must not already be administered on the System Synchronization plan.

- *Communication Type*

  Indicates the type of DS1 protocol that is to be used. For a DS1 access endpoint, enter **56k-data** (default), **64k-data**, or **voice-grade-data**, as appropriate. Value **64k-data** is not permitted for robbed-bit trunks. For a Wideband Access Endpoint (WAE), enter **wideband**.

  NOTE:
  Value **56k-data**, **64k-data**, or **wideband** turns on the *ITC* field. Value **voice-grade-data** turns off the *ITC* field.

- *Width*

  This field is displayed only if *Communication Type* contains **wideband**. Beginning with the specified starting port, enter the number of adjacent DS0 ports (**2** to **30**) that comprise the WAE. Value **6** (default) defines a 384k WAE. This field cannot be blank.

  NOTE:
  If value **24** (T1) or **30** (E1) is entered, all lower video rates (including 768k, 384k, and 128k) are also supported for this endpoint.

- *COR*

  For the Class of Restriction, leave at the default value **1**.

- *COS*

  For the Class of Service, leave at the default value **1**.

- *ITC* (Information Transfer Capability)

  NOTE:
  This field is displayed whenever *Communication Type* contains “56k-data,” “64k-data,” or “wideband.” This field is not used whenever RBS is administered.

  The field is used to determine the type of transmission facilities to be used for ISDN calls originated from the corresponding endpoint. Valid entries are **restricted** (default for “56k-data”) and **unrestricted** (default for “64k data” and “wideband”).

  The value **unrestricted** indicates that only unrestricted facilities are to be used to complete the call. An unrestricted facility is a transmission facility that does not enforce “1’s” density digital transmission (that is digital information is sent exactly as is). Always enter **unrestricted** for Wideband and 64k data access endpoints.

  The value **restricted** indicates that either restricted or unrestricted transmission facilities will be 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 “0’s” will be converted to a sequence of seven “0’s” and a digital “1”).

  Whenever an access endpoint is added with the ITC administered as “unrestricted,” the endpoint’s associated port must be a channel of a DS1 circuit pack with the *Zero Code Suppression* field administered as “B8Zs.” Otherwise, form submission is rejected with the appropriate error message. On the other hand, whenever an access endpoint is added with the ITC administered as “restricted,” the endpoint’s associated port can be a channel from a DS1 circuit pack with the *Zero Code Suppression* field administered as either “ZCS” or “B8ZS.”

  For an existing access endpoint, the field can be changed from “restricted” to “unrestricted” only if its associated port is a channel of a DS1 circuit pack with its *Zero Code Suppression* field administered as “B8ZS.” Otherwise, form submission is rejected with the appropriate error message.

2. Submit the Access Endpoint form.

Administering Synchronization

The term “synchronization” refers to an arrangement whereby digital facilities operate from a common clock. Whenever digital signals are transmitted over a communications link, the receiving end must be synchronized with the transmitting end to read the digital signal properly.

The Synchronization Plan form is used to assign primary and secondary external synchronization sources. The information in the form provides synchronization between the MCU and the digital facilities it uses. The MCU must be synchronized to the incoming data rate of 8,000 frames per second. This timing information reference may be derived from DS1 or data interface (DI) facilities.

The clock and the software monitor the synchronization signal. If a loss of synchronization signal (frame slip rate) is detected, the signal source switches from the primary synchronization sources to the secondary synchronization sources. If both these synchronization sources are lost, the MCU switches to the TN768 or TN780 Tone Clock circuit pack source. Loss of the reference signal, excessive phase changes, and variation in timing frequency can cause loss of data or dropped data connections.

The following table provides the commands that relate to the synchronization plan.

Table 2-48. Synchronization Plan Administration Commands

Action	Object	Qualifier
change	synchronization	
display	synchronization	print or schedule

Use the following procedure to change the synchronization assignments:

1. At the *command:* prompt, enter **change synchronization**.
2. Verify that the screen displays the *Synchronization Plan* form.
3. Use TAB or RETURN to advance to the fields you want to change (see *Synchronization Plan Form* for a list of fields and values).
4. Submit the form.
5. To verify the form, enter **display synchronization**.

Data Interface Option

Use the information in this section to provide synchronization between the MCU and a data interface. The following figure provides an appropriate example of the Synchronization Plan form.

```
change synchronization                                    Page 1 of 2


                              SYNCHRONIZATION PLAN
       Stratum: 4     Circuit Type: DI_    Primary: 01B0201     Secondary: 01B0501
DS1 SYNCHRONIZATION SOURCES                    DI SYNCHRONIZATION SOURCES
Location     Name       Slip       Port  Rate  Mode    Timing     Ext     Name
  01A15                   n      01B0101 56k   max  DCE-Extclock 20000 eia530 one
  01A16                   n      01B0201 56k   fix  DTE          20001 RS449 test
  01A13  ESM ds1          y      01B0301 1920k max  DTE          20002 V.35 tests
                                 01B0401 56k   max  DTE          20003 rs-449 two
                                 01B0501 56k   fix  DTE          20004 Fifth circ









          NOTE:  DS1 and DI stratum 4 source is type II synchronization
```

Figure 2-77. Synchronization Plan (Source from DI Facilities)

NOTE:
Before a DI port can be assigned to the synchronization plan, it must be administered as an access endpoint with bandwidth type “fixed” and timing mode “DTE.”

- *Stratum*

  Display-only field shows value **4** (default) to define the level or quality of the synchronization clock.

  NOTE:
  Value **4** turns off the *Port Network* field, and it turns on many of the other fields.

- *Circuit Type*

  Enter **DI** to indicate the type of circuit pack that is to be used for the synchronization clock.

  NOTE:
  Default value **DS1** is used to administer synchronization for DS1 facilities.

  For value **DI**, the *Location*, *Name*, and *Slip* fields headed by "DS1 SYNCHRONIZATION SOURCES" are not applicable.

- *Primary*

  Identifies the DI circuit pack providing the primary stratum 4 clock. Enter the appropriate 7-character alphanumeric value according to the breakdown provided in the following table.

Character Position	Meaning	Value
1-2	Network number	01
3	Carrier	A through E
4-5	Slot number	01 through 20
6-7	Circuit Number	01

  NOTE:
  The DI circuit pack must be installed and administered. Also, the port circuit (network number) must be 01. Finally, the associated port must be available, its corresponding *Mode* field must be set to **fix**, and its corresponding *Timing* field must be set to **DTE**.

- *Secondary*

  Identifies the DI circuit pack that will provide the stratum 4 clock if the primary DI circuit pack fails. Enter the appropriate 7-character alphanumeric value according to the breakdown provided in the following table.

Character Position	Meaning	Value
1-2	Network number	01
3	Carrier	A through E
4-5	Slot number	01 through 20
6-7	Circuit Number	01

NOTE:
The secondary DI circuit pack must be installed and administered, and it must not be the same as the primary DI circuit pack. Also, the port circuit (network number) must be 01. Finally, the associated port must be available, its corresponding *Mode* field must be set to **fix**, and its corresponding *Timing* field must be set to **DTE**.

The following fields are headed by “DI SYNCHRONIZATION SOURCES.” and are relevant only whenever a data interface is administered via the Access Endpoint form.

- *Port*

  Display-only field. Identifies the location of DI ports that have been administered as access endpoints. Default is blank.

- *Rate*

  Display-only field. Indicates the bandwidth for the associated DI port as administered on the Access Endpoint form. Values include **56k** through **1920k** (excluding **1288k**) and blank (default).

- *Mode*

  Display-only field. Indicates the bandwidth mode of the bandwidth in the *Rate* field. Values include **fix(ed)**, **max(imum)**, and blank.

- *Timing*

  Display-only field. Indicates the value for the *Timing Mode* field in the Access Endpoint form for DI circuit packs. Values include **DCE**, **DCE-Extclock**, **DTE-Extended**, **DTE**, **DTE-Intclock**, and blank.

- *Ext*

  Display-only field. Identifies the 5-digit extension of the access endpoint for which the associated DI port is administered.

- *Name*

  Display-only field. Identifies the name given to the circuit when it was administered. For DI facilities, this name is taken from the Access Endpoint form. Default is blank.

DS1 Interface Option

Use the information in this section to provide synchronization between the MCU and a DS1 interface. The following figure provides an appropriate example of the Synchronization Plan form.

```
change synchronization                                    Page 1 of 2


                              SYNCHRONIZATION PLAN
       Stratum: 4     Circuit Type: DS1     Primary: 01B12   Secondary: 01B13
DS1 SYNCHRONIZATION SOURCES                    DI SYNCHRONIZATION SOURCES
Location     Name        Slip       Port  Rate  Mode    Timing     Ext     Name
 01A15  DS1 one            n
 01A16  DS2 two            n
 01A13  ESM ds1            y









           NOTE:  DS1 and DI stratum 4 source is type II synchronization
```

Figure 2-78. Synchronization Plan (Source from DS1 Facilities), Page 1

```
change synchronization-plan                              Page 2 of 2


                              SYNCHRONIZATION PLAN

DS1 SYNCHRONIZATION SOURCES                    DI SYNCHRONIZATION SOURCES
Location     Name        Slip         Port  Rate Mode Timing   Ext     Name
   ____  ________________  _
   ____  ________________  _
   ____  ________________  _
   ____  ________________  _
   ____  ________________  _
   ____  ________________  _
   ____  ________________  _
   ____  ________________  _
   ____  ________________  _
   ____  ________________  _
   ____  ________________  _
   ____  ________________  _


          NOTE:  DS1 and DI stratum 4 source is type II synchronization
```

Figure 2-79. Synchronization Plan (Source From DS1 Facilities), Page 2

NOTE:
Before a DS1 port can be assigned to the synchronization plan, a DS1 circuit pack must be administered.

The following list identifies the fields on the form and the related values.

- *Stratum*

  Display-only field shows value **4** to define the level or quality of the synchronization clock..

- *Circuit Type*

  Display-only field containing value **DS1** (default). Indicates the type of circuit pack that is to be used for the synchronization clock.

  NOTE:
  Value **DS1** uses the *Primary* and *Secondary* (board) fields.

- *Primary*

  Identifies the DS1 circuit pack providing the primary stratum 4 clock. Enter the appropriate 5-character alphanumeric value according to the breakdown provided in the following table.

Character Position	Meaning	Value
1-2	Network number	01
3	Carrier	A through E
4-5	Slot number	01 through 20

  NOTE:
  The DS1 circuit pack must be installed, administered, and available.

- *Secondary*

  Identifies the DS1 circuit pack that will provide the secondary stratum 4 clock if the primary DS1 circuit pack fails. Enter the appropriate 5-character alphanumeric value according to the breakdown provided in the following table.

Character Position	Meaning	Value
1-2	Network number	01
3	Carrier	A through E
4-5	Slot number	01 through 20

  NOTE:
  The secondary DS1 circuit pack must be installed, administered, and available. It must also not be the same as the primary DS1 circuit pack.

The following fields are headed by “DS1 SYNCHRONIZATION SOURCES.” and are relevant only whenever a DS1 circuit pack is administered in the DS1 Circuit Pack form.

- *Location*

  Display-only field. Identifies the location of the administered DS1 circuit pack.

- *Name*

  Display-only field. Identifies the name given to the circuit when it was administered. For DS1 facilities, this name is taken from the DS1 form.

- *Slip*

  Display-only field. Indicates if slip detection is enabled. Values include **y** and **n**.

Checking Access Endpoints

This section presents and discusses the report for checking access endpoints.

Access Endpoint Report

The Access Endpoint report is used to gather information on DI and DS1 access endpoints. To access the report, enter the **list access-endpoint [*xxxxx*]** command, where ***xxxxx*** is the access endpoint number and optional.

An example of the report appears as follows.

```
list access-endpoint

                                ACCESS ENDPOINT

                                        ------- DS1 -------  -------- DI -------
                         Interf         Communication        Band Band  Timing
Ext    Port    Name      Type    COR COS Type          Width  Rate Type  Mode

12345 01A0101 DS1 exampl DS1     63  15  wideband       6
56789 01A1001 DI example V.35    1   1                        1920K fix  DCE
11111 01A0301 DS1-type   DS1     9   3   wideband       6
55555 01A1201 DI-type    EIA-530 1   1                        768K  max  DTE
55000 01A0501 DI-brd     RS449   11  11                       1920K fix  DCE-Extc
```

Figure 2-80. Access Endpoint Report (Page 1)

```
status access-endpoint 6776                                  Page 2 of 2

                      DI CIRCUIT PACK STATUS INFORMATION


    DTE LEADS:    DTR: 0        RTS: 0
    DCE LEADS:    DCD: 0        DSR: 0          CTS: 0         RING IND: 0


    DTE Send Timing Input: 0___K             Internal Clock Rate: 384_K
 DTE Receive Timing Input: 0___K              Receive Timing PPM: 0___K
DCE Terminal Timing Input: 0___K                       FIFO Rate: 0 B

                Input    Output   Loop Active: n  Using Terminal Timing: n
 FIFO Overflow: 0        0        Using Cable: n            Timing Mode: DTE
FIFO Underflow: 0        0        Cable Ready: y             Restricted: n
    FIFO State: reset__  reset__
     FIFO Size: empty__

               --------  STATUS BYTES  --------
           00  00  00  00  00  1C  EE  00  00  08  00  00
           00  00  00  00  30  00  00  00  00  00  00  3F
```

Figure 2-81. Access Endpoint Report (Page 2)

The following list shows the data collected for the report.

- *Ext*

  Lists the extension number specified in the command line. This extension number is used to access the trunk endpoint.

- *Port*

  Lists the location of the circuit pack and the circuit to which the endpoint will be connected.

- *Name*

  Lists the alphanumeric characters used to identify the endpoint.

- *Interf(ace) Type*

  Lists the type of interface administered for the current access endpoint.

- *COR*

  Lists the appropriate class of restriction (COR) number.

- *COS*

  Lists the appropriate class of service (COS) number.

- *DS1 Communication Type*

  Lists the data rate and/or transmission medium for the access endpoint.

- *DS1 Width*

  Lists the channel width for the access endpoint.

- *DI Band Rate*

  Lists the data rate for the access endpoint.

- *DI Band Type*

  Identifies whether the transmission rate provided is fixed or the maximum.

- *DI Timing Mode*

  Indicates how the MCU will operate (as either DCE or DTE) for the given port and also whether the clocking is supplied internally or externally from the MCU.

- *DTE LEADS:...DTR:...RTS*:

  Indicates whether the specified DTE lead(s) (DTR and/or RTS) are enabled. (The DTE side is responsible for raising the DTR and RTS.) Value **1** indicates that the lead(s) are enabled, and value **0** indicates that the lead(s) are not enabled.

- *DCE LEADS:...DCD:...DSR...CTS...RING IND*:

  Indicates whether the specified DTE lead(s) (DCD, DSR, CTS, and/or RING IND) are enabled. (The DCE side is responsible for raising the DCD, DSR, and CTS.) Value **1** indicates that the lead(s) are enabled, and value **0** indicates that the lead(s) are not enabled.

- *DTE Send Timing Input*

  Indicates a clock rate detected by the DI circuit pack from the other side. Values include **0k**, **56k**, **64k**, **112k**, **128k**, **336k**, **384k**, **512k**, and **768k**.

- *DTE Receive Timing Input*

  Indicates a clock rate detected by the DI circuit pack from the other side. Values include **0k**, **56k**, **64k**, **112k**, **128k**, **336k**, **384k**, **512k**, and **768k**.

- *DCE Terminal Timing Input*

  Indicates a clock rate detected by the DI circuit pack from the other side. Values include **0k**, **56k**, **64k**, **112k**, **128k**, **336k**, **384k**, **512k**, and **768k**.

- *Internal Clock Rate*

  Indicates the rate that the MCU is generating (that is, the rate that the MCU will put on any output time lead). Values include **0k**, **56k**, **64k**, **112k**, **128k**, **336k**, **384k**, **512k**, and **768k**.

- *Receive Timing PPN*

  Indicates the clock rate and, indirectly, the clock count on the receive timing lead. Values include **56k**, **64k**, **112k**, **128k**, **336k**, **384k**, and **1920k**. To get the actual clock count, divide the appropriate value by .512. This count is used for synchronization whenever the MCU is in DTE mode.

- *FIFO Rate*

  Indicates the number of B-channels that are currently in use. Values include **1B**, **2B**, **3B**, **4B**, **5B**, **6B**, **8B**, and **12B**.

- *FIFO Overflow/FIFO Underflow:...Input...Output*

  Indicates whether the input and output data buffer (FIFO) overflow buffers are slipping. A data buffer slips whenever it becomes full; in such a case, all the data is lost, and the data buffer is reset. These fields are frequently cleared by firmware.

  Slipping is more likely to occur whenever the MCU is in DTE mode and the MCU's synchronization source clock differs from the clock provided by the DCE. If a value greater than **0** (and up to **15**) appears in the *FIFO Overflow...Input* field, the DCE device clock is faster than the MCU synchronization source clock. If a value greater than **0** (and up to **15**) appears in the *FIFO Underflow...Input* field, the MCU synchronization source clock is faster than the DCE device clock. Value **0** in each of these fields indicates that the clocks are synchronized.

  If a value greater than **0** (and up to **15**) appears in the *FIFO Underflow...Output* field, the DCE device clock is faster than the MCU synchronization source clock. If a value greater than **0** (and up to **15**) appears in the *FIFO Overflow...Output* field, the MCU synchronization source clock is faster than the DCE device clock. Value **0** in each of these fields indicates that the clocks are synchronized.

- *FIFO State/FIFO Size...Input...Output*

  Identifies the state of the input and output data buffers (FIFOS). Values include **held reset**, **enabled**, **never enabled**, l**ess than 1/8 full**, **at least 1/8 full**, **at least 3/8 full**, **full**, and **empty**. Where appropriate, a corresponding rate indication is also provided.

- *Loop Active*

  Indicates whether various internal loops are active toward the endpoint. Values include **y** and **n**. Value **y** indicates that the line loop, TDM loop, and FIFO loop are active. The line loop and TDM loop can be activated via the Access Endpoint form.

- *Using Cable*

  Indicates whether the DI/MMI cable is being used. Values include **y** and **n**.

- *Cable Ready*

  Indicates whether the DI/MMI cable is connected and whether the DI circuit pack can communicate with the MMI circuit pack via the cable. Values include **y** and **n**.

- *Using Terminal Timing*

  Indicates whether terminal timing is being used within the system. Values include **y** and **n**.

- *Timing Mode*

  Indicates how the data interface is configured. Values include **DTE** and **DCE**.

- *Restricted*

  Indicates whether or not the call is restricted. Values include **y** and **n**. Value **n** indicates that the call is unrestricted.

NOTE:
The "STATUS BYTES" area at the bottom of page 2 of the form is reserved for technical personnel and is intended for troubleshooting purposes.

Performing System Backups and Saving Translations

The **save translations** command copies the current system translations onto the memory card located on the Network Control circuit pack. One system memory card is included with the MCU. A backup memory card may be purchased optionally.

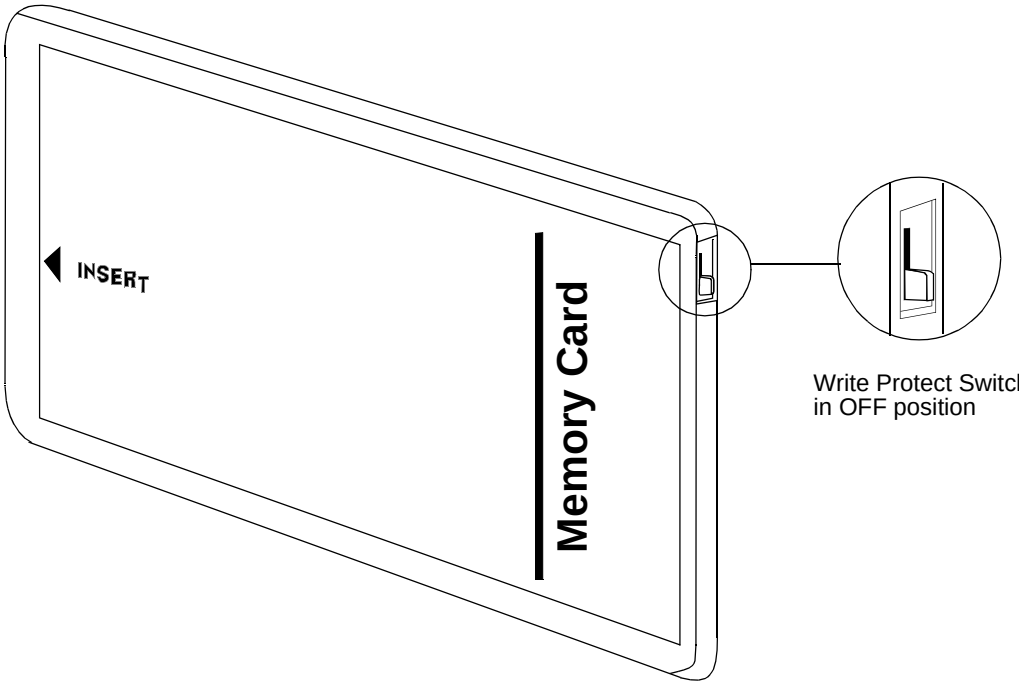

Figure 2-82. Memory Card

To save translations on the original system memory card and to make a backup memory card, follow these procedures:

1. Be sure the system memory card is inserted in the Network Control circuit pack with the write-protect switch in the OFF position.
2. Log in as **cust** on the MCU-MT.
3. At the *command:* prompt, enter **save translation** and press RETURN.
4. Check for a **0** in the *Error Code* column; a **0** indicates the save translation was successfully completed.

   NOTE:
   If a number other than **0** appears, the save translation process did not complete. Record the number in the *Error Code* field along with any *Error Message,* then call either the Lucent Technologies Video Technical Center (VTC)/Customer Care Center at **800 242-2121** or the International Technical Assistance Center (ITAC) at **303 804-3777**.

5. If the save translation procedure is successful, remove the system memory card and insert the backup memory card (if you have purchased one). Repeat Steps 2 and 3 and then swap the memory cards. Place the backup memory card in a safe place.

NOTE:
Translations can also be saved during scheduled maintenance by using the Maintenance-Related System Parameters form.

Checking Status of the MMI Circuit Packs

The MMI Boards form provides information on all the MMI circuit packs within the MCU system. Specifically, the form identifies MMI circuit packs connected to DI circuit packs as well as to DS1 circuit packs.

The following table provides the command used to access the form.

Table 2-49. Processor Channels Administration Commands

Action	Object	Qualifier
list	mmi	-----

To access the form, follow this procedure:

1. At the *command:* prompt, enter **list mmi**.
2. Verify that the screen displays the *MMI Boards* form (see *MMI Boards Form* for a list of the fields and their values).

MMI Boards Form

The MMI Boards form appears as follows:

```
list mmi                                                    Page 1


                                  MMI BOARDS

MMI   MMI       MMI      Ports  Ports                ---Cabled---
Addr  Code  Synch Status In Srv In Use  Cable Status Addr   Code    Interface

1A02 TN787K listening      24     0  in-service     01A10 TN2207   ESM
1A07 TN787K providing      32     0  out-of-service 01A04 TN2207   network
1A10 TN787K listening      32     0  in-service     01A11 TN2207   network
```

Figure 2-83. MMI Boards Form

The following list identifies the fields on the MMI Boards form and the possible values.

- *MMI Addr*

  Indicates the address of the MMI port circuit pack.

- *MMI Code*

  Indicates the “TN” code for the MMI port circuit pack (for example, TN787K).

- *MMI Synch Status*

  Indicates the synchronization status of the MMI circuit pack. Values include **providing**, **listening**, and **out-of-synch**.

- *Ports In Srv*

  Indicates the total number of MMI resources in service per circuit pack. Values range from **0** to **32** for the MSM and **0** to **24** for the ESM.

  NOTE:
  This field applies to MMI ports that are not in a maintenance-busy state and are therefore available for use.

- *Ports In Use*

  Indicates the total number of MMI resources being used per circuit pack. Values range from **0** to **32** for the MSM and **0** to **24** for the ESM.

  NOTE:
  This field applies to MMI ports that are either reserved or being used. Even if the “Appl Compliant” flag is set to “no,” the ports are reserved. This is true even in the case where the ports are not in use due to the flag being changed from “no” to “yes” while the conference is active.

- *Cable Status*

  Indicates the status of the cable linking the DS1 and MMI circuit packs. Values include **in-service**, **out-of-service**, and **maint-busy**.

- *Cabled Addr*

  Identifies the address of the DS1 or DI circuit pack that is cabled to the corresponding MMI board.

- *Cabled Code*

  Identifies the “TN” code of the circuit pack to which the MMI board is connected. Possible values displayed are “TN767,” “TN2207,” and “TN2236.”

- *Interface*

  Indicates whether the MMI circuit pack is cabled to the network-DS1 circuit pack (**network**) or to the ESM-DS1 circuit pack (**ESM**).

Administering Continuous Presence *Plus*

Continuous Presence *Plus* is a customer option that enables a multipoint video conference to compose a picture consisting of the video from a maximum of 16 H.320 sites participating in the conference. CRCS and Universal Conference Control can be used only to control the composition of the picture.

The following are required to enable this feature:

- Inclusion of a single TN2237C, version 16 or higher Video Processor circuit pack
- Administration of the appropriate fields in the Customer-Options form
- Administration of the appropriate fields in the Conference Record form

To administer Continuous Presence *Plus*, complete the fields in the forms identified in the following table. Appropriate page references to the forms are provided. Detailed instructions for completing the forms are provided after the table

Table 2-50. Required Forms and Fields for Continuous Presence *Plus*

Form	Field	Page Reference
Customer-Options	■ *Conferencing Suite* ■ *Continuous Presence/Executive Conf*	page 2-22
Conference Record	■ (See the following “Conference Record Form” section.)	page 1-3

To administer CPP conferences, populate the forms described in the following sections as directed.

NOTE:
Continuous Presence *Plus* is discussed in detail in Appendix B.

Customer-Options Form

The system administrator must set the *Continuous Presence/Executive Conf* and/or *Conferencing Suite* field(s) to a value between **0** and **8** that, when combined with the values within the *Speed Match (SM)* and *Executive Conference* fields, does not exceed a total of eight (corresponding to the maximum number of possible Video Processor circuit packs within the system).

NOTE:
Actually, there are 10 VP circuit packs in the system; however, only eight VP circuit packs can be used. If one of the circuit packs has a problem, one of the two remaining “spare” circuit packs is used.

Conference Record Form

Complete the following steps:

1. Ensure that the *Continuous Presenc*e field contains **y**.
2. Use the *Quad (1—4) Ext* fields to assign specific endpoints to a quadrant in the Continuous Presence *Plus* display. To assign an endpoint to a specific quadrant, enter its MCU-extension into the appropriate field. If any field is left blank, the corresponding quadrant is considered “unassigned.” Endpoints are VAS-switched into the unassigned quadrants corresponding to such fields, as appropriate. The MCU-extension cannot be a Universal Conference Control (UCC), Audio Add-On, or control extension in a Presentation or Chair mode conference.
3. Use the *Starting Image* field to specify the starting video composition for the CPP conference. Values include **full**, **dual-panoramic**, **lower-panoramic**, **upper-panoramic**, and **quad** (default). This field appears only whenever the *Continuous Presence* field contains **y**.
4. Use the *Force Video Now To* field to toggle between 3-image, quadrature, and full-screen video within a conference. Values include **dual-panoramic**, **lower-panoramic**,**upper-panoramic**, **full**, **quad**, and blank (default). This field appears only whenever the *Continuous Presence* field contains **y**. Whenever the Conference Record is submitted, the display change is applied to the conference and remains in effect until it is changed by another change conference command, UCC, or CRCS.
5. Use the *Empty Quadrant Extension* field to provide a fill pattern in the Continuous Presence *Plus* display whenever there are fewer than four parties in the conference. The fill endpoint is an endpoint in the conference. This endpoint’s MCU-extension is entered into this field. If there are only two active endpoints in the conference, the video from the fill endpoint is displayed in the remaining two quadrants. The MCU-extension cannot be a UCC, Audio Add-On, or control extension in a Presentation or Chair mode conference.

Checking the Status of the Video Processor Circuit Packs

The List Video Processors form provides information on all the administered Video Processor circuit packs within the MCU system. To access the form, enter **list video-processors** at the command prompt. The form appears as follows.

```
list video-processors

                                VIDEO PROCESSORS

VP                  Service                          Sync        Alloc In
Addr Application    State                            Status      Conf


01A10 vp-plus       in-service                       listening
01A11 vp-plus       out-of-service                   out-of-synch
01A07 vp-plus       in-service                       listening
```

Figure 2-84. List Video Processors Form

The following list identifies the fields on the forms and the possible values:

- *VP Addr*

  Indicates the address of the Video Processor circuit pack.

- *Application*

  Indicates the Continuous Presence *Plus* application performed by the Video Processor circuit pack. Value **vp-plus** indicates the TN2237 Video Processor Plus circuit pack mixer and adapter application.

- Service State

  Indicates the service state of the Video Processor circuit pack. Values include **in-service** and **out-of-service**.

- *Sync Status*

  Indicates the synchronization status of the circuit pack. Values include **listening** and **out-of-sync**. The circuit pack derives synchronization from the providing MMI circuit pack.

- *Alloc in Conf*

  If the field contains a number, indicates that the Video Processor circuit pack is active in a conference. The number displayed is the ID of the active conference.

Administering Modem Pool Groups

The MCU supports the Combined Modem Pooling solution. The required hardware is called a *combined conversion resource*. It includes a separate Trunk Data Module and modem administered as a unit. The Trunk Data Module component of the conversion resource is a 7400A Data Module. The module connects to a digital port using Digital Communications Protocol (DCP); the modem connects to an analog port. The MCU supports the Combined Modem Pooling Solution for use in remote connectivity between the MCU and the two CRCS modules, ResCenter and OpCenter, whenever all these components are not colocated.

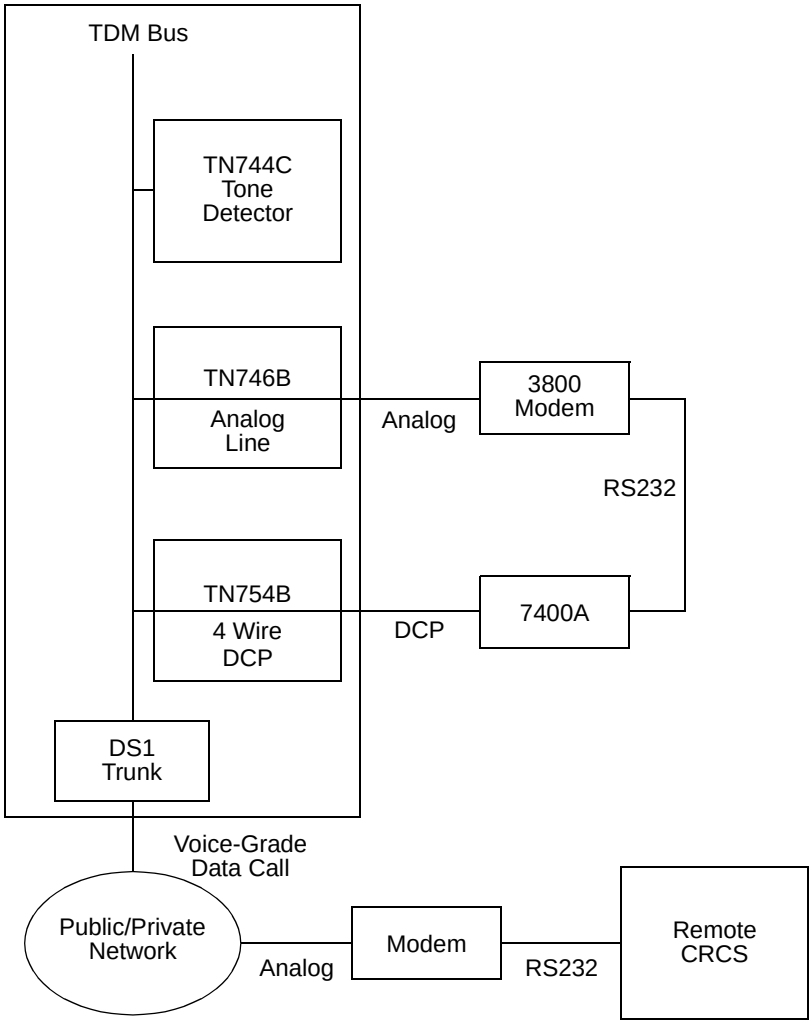

Figure 2-85. MCU/CRCS Combined Modem Pool Configuration

The commands in the following table can be used to access the Modem Pool Groups form.

Table 2-51. Modem Pool Group Commands

Action	Object	Qualifier*
add	modem-pool num	1 - 5
change	modem-pool num	1 - 5
display	modem-pool num	1 - 5, print or schedule
list	modem-pool num	print or remove
remove	modem-pool num	1 - 5

* Qualifier “maximum” is the maximum number available in your system configuration.

To change information associated with modem pool groups, follow these procedures:

1. At the *command:* prompt, enter **change modem-pool num**.
2. Verify that the screen displays the Modem Pool Groups form. (See “Modem Pool Groups Form” for a list of fields and values.)
3. Use TAB or RETURN to advance to the fields you want to change.
4. Submit the form.
5. To verify the form, enter **display modem-pool num.**

Modem Pool Groups Form

The “add” Modem Pool Groups form appears as follows:

```
add modem-pool 2                                          Page  1 of  1

                                  MODEM POOL GROUP
                      Group Number: 2                        Group Type: combined
                        Modem Name: _______                Hold Time(min): 5
                     Time Delay(sec): 0                         Direction: two-way
    Answer Supervision Timeout(sec): 0

        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:
```

Figure 2-86. Modem Pool Groups Form (“Add” Option)

The “list” Modem Pool Groups Form appears as follows:

```
list modem-pool

                                MODEM POOL GROUPS

Group    Group          Group    Modem    Hold     Duplex  Sync    Modem   Group
Number   Type           Size     Name     Time     Mode    Mode    Speed   Direction
------   ------         ------   ------   ------   ------  ------  ------  ----------
1        combined       0                 5        full    async   9600
```

Figure 2-87. Modem Pool Groups Form (“List” Option)

The following list identifies and describes the fields on the Modem Pool Groups form.

- *Group Number*

  Enter a group number whenever you are completing a paper form. This is a display-only field whenever the form is accessed via an **add** or **change** administration command.

- *Group Type*

  Enter **combined**.

- *Modem Name*

  Enter a 1- to 6-digit alphanumeric character string to indicate the name of the modem pool.

- *Hold Time (min)*

  Enter the maximum number of minutes (**1** through **99**) that a conversion resource in the group may be held while a call waits either in a queue or reserved after Data Call Preindication. Default is **5**.

- *Time Delay*

  Enter the time delay in seconds (**0** through **255**) to insert between sending the ringing to the modem and the off-hook alert to the data module. Default is **0**. (This field displays only when *Group Type* is **combined**.)

- *Direction*

  Enter the direction (**incoming**, **outgoing** or **two-way**) of the call for which a modem pool will operate. Default is **two-way**. (This field displays only when *Group Type* is **combined**.)

- *Answer Supervision Timeout*

  Used for all MCU trunk types to indicate the amount of time in seconds that the system allows for a call to be answered before beginning a CDR record of a call. This interval begins as soon as the outgoing trunk is seized. Values include **0** through **250**; default is **10**.

  NOTE:
  This timeout does not override answer supervision sent from the network or other firmware (DS1 port circuit timers). DS1 firmware-sent answer supervision is administered in *Outgoing End of Dial (sec)* on the Administrable Timers page of the Trunk Group form.

The following three fields can be assigned for the Combined Modem Pooling solution:

- *Speed*

  Enter one, two, or three communication speeds in bits per second of the conversion resources in the group. Speeds include **LOW**, **0** to **300** blind sampled, **300**, **1200**, **2400**, **4800**, **9600**, or **19200**. Separate entries with more than one speed with slashes (for example, **300/1200/2400**, which indicates a maximum of three running speeds). For CRCS connectivity, enter **9600**. Default is **LOW/300/1200**.

- *Duplex*

  Enter **full** or **half** to indicate the duplex mode of the conversion resources in the group. For CRCS connectivity, enter **full**. Default is **full**.

- *Synchronization*

  Enter **sync** or **async** to indicate the synchronization mode of the conversion resources in the group. For CRCS connectivity, enter **async**. Default is **async**.

  The Port Pair Assignments area contains the following field:

- *Analog Digital*

  Enter the port numbers of the modem/TDM pair in a conversion resource. Two port entries are required. Include seven characters for each entry. Use the following breakdown:

Character Position	Meaning	Value
1-2	Cabinet number	01 through 03
3	Carrier	A through E
4-5	Slot number	01 through 20
6-7	Circuit Number	01 through 32

Administering Executive Conferencing

Executive Conferencing is a customer option that enables a conference to compose a picture of two or three “other” sites (endpoints) for each conference participant. With this feature, no endpoint ever sees itself.

To administer Executive Conferencing, complete the fields in the forms identified in the following table. Appropriate page references to the forms are provided. Detailed instructions for completing the forms are provided after the table

Table 2-52. Required Forms and Fields for Executive Conferencing

Form	Field	Page Reference
Customer-Options	■ *Conferencing Suite* ■ *Continuous Presence/Executive Conf* ■ *Executive Conference*	page 2-22
Conference Record	■ (See the following “Conference Record Form” section.)	page 1-3

To administer Executive Conferencing conferences, populate the forms described in the following sections as directed.

NOTE:
Executive Conferencing is discussed in detail in Appendix B.

Customer-Options Form

The system administrator must set the *Executive Conference*, *Continuous Presence/Executive Conf*, and/or *Conferencing Suite* field(s) to a value between **0** and **8** that, when combined with the value within the *Speed Match (SM)* field, does not exceed a total of eight (corresponding to the maximum number of possible Video Processor circuit packs within the system).

Conference Record Form

Complete the following steps:

1. Ensure that the *Executive Conference* field contains **y.**
2. Use the *Site (1—3) Ext* fields to assign specific endpoints in the Executive Conferencing display. To assign an endpoint, enter its MCU-extension into the appropriate field.

These fields can be left blank. If any field is left blank, the corresponding screen display is considered “unassigned” and is provided with a fill pattern. This means that the corresponding conference participant is not established as a video source for the broadcast set. For example, if an extension is administered for Site #1 and Site #2 but no extension is provided for Site #3, then, in Executive Conferencing Mode: Site #1 sees a panoramic image of Site #2, along with a blue fill pattern in the other half of the screen; Site #2 sees Site #1 in panoramic view, along with a blue fill pattern in the other half of the screen; Site #3 sees a dual-panoramic image of Sites #1 and #2, and Sites #4 through #25 see a panoramic view of Site #1, along with a quad view of Site #2 and a blue fill pattern in the remaining quadrant. You can recover from this situation by using either the Dynamic Resizing capability to administer a broadcast extension for Site #3 or UCC to go to full screen or to assign a conference participant to be Broadcaster #3.

3. Use the *Starting Image* field to specify the starting video composition for the Executive Conferencing conference. Values include **full** and **executive** (default). This field appears only whenever the *Executive Conference* (or *Continuous Presence*) field contains **y**.

4. Use the *Force Video Now To* field to toggle between 3-image and full-screen video within a conference. Values include **full** and **executive** (default). This field appears only whenever the *Executive Conferenc*e (or *Continuous Presence*) field contains **y**. Whenever the Conference Record is submitted, the display change is applied to the conference and remains in effect until it is changed by another change conference command, UCC, or CRCS.

Administering T.120 Data Conferencing

To administer T.120 Data Conferencing, complete the fields in the forms identified in the following table. Appropriate page references to the forms are provided. Detailed instructions for completing the forms are provided after the table

Table 2-53. Required Forms and Fields for T.120 Data Conferencing

Form	Field	Page Reference
Customer-Options	■ *Max. T.120 Port Capacity*	page 2-22
Conference Record	■ *Audio Mode* ■ *T.120 Data Mode* ■ *MLP Rate* ■ *App Comp?*	page 1-3
Set Date and Time	■ All	page 2-9
DS1 Circuit Pack	■ (See the “Administering the ESM from the MSM” section.)	page 2-61
Data Module	■ (See the “Administering the ESM from the MSM” section.)	page 2-78
Interface Links	■ (See the “Administering the ESM from the MSM” section.)	page 2-81
Processor Assignments Channel	■ (See the “Administering the ESM from the MSM” section.)	page 2-84
Signaling Group	■ (See the “Administering the ESM from the MSM” section.)	page 2-87
Interface Links	■ *Enable*	page 2-81

To administer T.120 Data Conferencing, populate the forms described in the following sections as directed.

Customer-Options Form

The system administrator must set the *Maximum Port Capacity: T.120* field to the appropriate value in the range of **4** to **24** in intervals of four.

Conference Record Form

Populate the form as follows:

1. Set the *T.120 Data Mode* field to **any-mlp**. This value indicates that the conference has data applications. Endpoints that have MLP capabilities are primary endpoints, and they have video capabilities. Endpoints that do not have MLP capabilities are secondary endpoints, and they have no video capabilities.
2. Populate the *MLP Rate* field to indicate the data rate for T.120 Data Conferencing. Values include **var-mlp**, **6.4k, 8k, 14.4k, 16k, 22.4k, 24k, 30.4k, 32k, 38.4k, 40k,** and **46.4k**.

   NOTE:
   MLP rates **40k**, **46.4k**, and **var-MLP** for T.120 Data Conferencing are not allowed in Speed Match conferences.

   For T.120 conferences, if the *Audio Mode* field contains **G.711**, **G.722**, **auto-w/upgrade** or **auto-down_only**, the *MLP Rate* field must contain either **6.4k**.

3. Set the *App Comp* (Application Compliant) field to **y**.

Forms For Administering the MSM-ESM Link

Complete the steps provided in the “Administering the ESM from the MSM” section earlier in this chapter.

Administering OneNumber Access

To administer OneNumber Access, complete the fields in the forms identified in the following table. Appropriate page references to the forms are provided. Detailed instructions for completing the forms are provided after the table

Table 2-54. Required Forms and Fields for OneNumber Access

Form	Field	Page Reference
Customer-Options	■ *OneNumber Access*	page 2-22
Conference Record	■ *OneNumber Access*	page 1-3

Customer-Options Form

Access the Customer-Options form, and enter the appropriate value from the following list into the *OneNumber Access* field to enable the feature appropriately.

- Value **aud** enables the feature only for Audio Add-On endpoints. H.320 and BONDed endpoints are not allowed.
- Value **mrate/aud** enables the feature for H.320 endpoints in multirate conferences and Audio Add-On endpoints in all conference types. H.320 endpoints in 2B conferences and BONDed endpoints are not allowed.
- Value **all** enables the feature for H.320 endpoints in 2B conferences, as well as for Audio Add-On and multirate endpoints in all conference types. BONDed endpoints are not allowed.

NOTE:
If a value to enable the feature is entered, any duplicate extensions administered for the feature must be of the type enabled via the value. Otherwise, such extensions are blocked via administration.

Conference Record Form

Access the Conference Record form and set the *OneNumber Access* field to **y**. This allows multiple endpoints to use the same MCU-extension within the same conference. Setting the field to **n** denies this capability. The field can be modified only from **n** to **y** while Dynamic Conference Resizing is activated.

Administering Lip Synchronization

Lip Synchronization can be administered for CPP, SM, or H.263/H.261 Transcoding conferences. One of two modes, Preserved Lip Sync Mode or Audio Priority, may be enabled.

To administer Lip Synchronization, complete the fields in the forms identified in the following table. Appropriate page references to the forms are provided. Detailed instructions for completing the forms are provided after the table

Table 2-55. Required Forms and Fields for Lip Synchronization

Form	Field	Page Reference
Feature-Related System Parameters	■ *Default Custom Lip Sync <=128K* ■ *Default Custom Lip Sync 224/256K* ■ *Default Custom Lip Sync >=336K* **NOTE**: These fields are required only for customizing Lip Sync delay.	page 2-43
Conference Record	■ *Lip Sync,* or*:* ■ *Lip Sync <=128K* ■ *Lip Sync 224/256K* ■ *Lip Sync >=336K* ■ *Value* (only for customizing Lip Sync delay)	page 1-3

Feature-Related System Parameters Form

To customize the Lip Synchronization delay for conferences at a rate of 128k or lower, populate the *Default Custom Lip Sync <=128K* field with a value of **0** through **1000** in increments of 20. To customize the delay for conferences at a rate of 224/256k, populate the *Default Custom Lip Sync 224/256K* field with a value of **0** through **1000** in increments of 20. To customize the delay for conferences at a rate of 336k or higher, populate the *Default Custom Lip Sync >=336K* field with a value of **0** through **1000** in increments of 20.

Conference Record Form

Use this form to administer one of two options, as explained in the following sections.

Option for Transcoded or Non-SM CPP Conferences

To establish Lip Synchronization mode for either H.263/H.261 transcoded conferences or CPP conferences without SM, populate the *Lip Sync* field, as appropriate. One of two modes, Preserved Lip Sync Mode or Audio Priority, may be enabled.

To enable Preserved Lip Sync Mode, enter one of two values into the field. Value **auto** enables a preselected MCU delay of a length determined by the endpoint type and/or audio standard. Preselected audio delays are defined within and uplinked by the TN2237 circuit pack. Value **custom** enables a delay of **0** to **1000** msecs in 20 msec increments. Entering this value brings up an associated *Value* field, which allows you to enter the appropriate msec value. To enable Audio Priority, enter value **none** into the field.

Option for SM Conferences

To establish Lip Synchronization mode for SM conferences at any of three rate ranges, populate the *Lip Sync <=128K, Lip Sync 224/256K,* or *Lip Sync >=336K* fields, as appropriate. One of two modes, Preserved Lip Sync Mode or Audio Priority, may be enabled.

To enable Preserved Lip Sync Mode, enter one of two values into the field(s). Value **auto** enables a preselected MCU delay of a length determined by the endpoint type and/or audio standard. Preselected audio delays are defined within and uplinked by the TN2237 circuit pack. Value **custom** enables a delay of **0** to **1000** msecs in 20 msec increments. Entering this value brings up an associated *Value* field, which allows you to enter the appropriate msec value.

NOTE:
As a default, each *Value* field is automatically populated with the corresponding value administered in the Feature-Related System Parameters form. For details, refer to "Features-Related System Parameters Form" on page 2-43.

To enable Audio Priority, enter value **none** into the field.

Administering Speed Match Conferences

To administer Speed Match conferences, complete the fields in the forms identified in the following table. Appropriate page references to the forms are provided. Detailed instructions for completing the forms are provided after the table

Table 2-56. Required Forms and Fields for Speed Match

Form	Field	Page Reference
Customer-Options	■ *Speed Match (SM)* ■ *Conferencing Suite*	page 2-22
Conference Record	■ (See the following "Conference Record Form" section.)	page 1-3

To administer Speed Match conferences, populate the forms described in the following sections as directed.

Customer-Options Form

The system administrator must set the *Speed Match (SM)* and/or the *Conferencing Suite* field(s) to a value between **0** and **8** which, when combined with the values within the *Continuous Presence/Executive Conf* and *Executive Conference* fields, does not exceed a total of eight (corresponding to the maximum number of possible Video Processor circuit packs within the system).

The Video Processor circuit packs can accommodate a mixture of full screen-only Speed Match conferences (with or without H.263/H.261 transcoding), CPP Speed Match conferences, CPP-only, and Executive Conferencing conferences. The number of simultaneous conferences using some form of video processing is controlled by the number of each of the three right-to-use permissions as displayed on the Customer-Options form. In particular, the sum of the CPP-only, full screen SM, combination SM/CPP, H.263/H.261 transcoding, and Executive Conferencing right-to-use permissions may not exceed eight since one dedicated Video Processor circuit pack is required to support each of these conference types. CPP/EC permissions are valid only for non-SM, single transfer rate CPP (or CPP-only) conferences or Executive Conferencing conferences. SM permissions are valid only for full screen SM or H.263/H.261 transcoding conferences. However, Conferencing Suite permissions can be used to provide a CPP-only conference, a full screen-only SM conference, an SM conference that also provides CPP video to and from endpoints at various transfer rates, an H.263/H.261 transcoding conference, or an Executive Conferencing conference. Therefore, as long as all of the available Conferencing Suite right-to-use

permissions have not been used, any such available permissions may be applied to administer a CPP-only, full screen-only SM, SM/CPP, H.263/H.261 transcoding, or Executive Conferencing conference. Therefore, for example, if the *SM* field contains **5**, the system supports a combined total of five full-screen conferences that are either speed-matched or transcoded. This means that the system can support, for example, three full-screen speed-matched conferences and two full-screen H.263/H.261 transcoded conferences (for a combined total of five conferences).

The right-to-use option for SM is preset by the Video Technical Center (VTC) upon delivery to the customer. To disable the feature, the port capacity for SM and Conferencing Suite must be set to **0** on the Customer-Options form. To enable the feature, the port capacity must be set to a nonzero number corresponding to the "right-to-use" number for the feature desired by the customer.

A maximum of eight conferences administered via the *Conferencing Suite* field at transfer rate 448/512k is allowed. A maximum of six such conferences administered at transfer rate 672/768k is allowed.

NOTE:
These conference maximums assume a three-party BONDed configuration with two parties administered at the indicated "high" rate and one party administered at the 2B rate.

By default, all the endpoints within an SM conference operate at the primary transfer rate. The transfer rates of the endpoints (except Audio Add-On and UCC endpoints) that will operate at a rate other than the primary transfer rate must be administered.

Conference Record Form

The following list explains how to administer Speed Match via the Conference Record form. The *Speed Match* field appears on page 1 of the form; the *Rate field* appears on pages 2 through 8 of the form.

1. Enter into the *Speed Match* field value **y** to allocate speed matching resources for the conference.

   The field cannot be set to **y** unless the *Speed Match* or *SM/CPP* field in the Customer-Options form is set to a value in the range **1** through **8**. Sufficient speed matching resources must be available for the requested time period.

   Also, the field cannot be set to **y** unless various fields in the Conference Record form are populated correctly. The following list identifies these fields along with the appropriate values for Speed Match, and it provides any necessary commentary:

   — *Status* — **inactive** (The field cannot be set for active or in-use conferences.)

— *Bandwidth* — **1x56k**, **1x64k**, **2x56k**, **2x64k**, **112k**, **128k**, **224k**, **256k**, **336k**, **384k**, **448k**, **512k**, **672k**, or **768k**

— *Video Mode* — Values for H.261-associated video modes, including **H.261-down_only**, **H.261-w/upgrade**, **H.261-CIF-Floor**, **H.261-CIF-Now**, or **H.261-QCIF-Now**

— *CIF Frame Rate* — **7.5 fps**, **10 fps**, **15 fps**, or **auto**

— *QCIF Frame Rate* — **7.5 fps**, **10 fps**, **15 fps**, or **auto**

— *MLP Rate* — If T.120 Data Conferencing is enabled, this field must contain one of the following values: **6.4k**, **8k**, **14.4k**, **16k**, **22.4k**, **24k**, **30.4k**, **32k**, or **38.4k**. SM conferences do not allow the following MLP rates: 40k and 46.4k. Also, variable rates (indicated by value “var_mlp”) are not allowed.

2. Ensure that the conditions in the following list are satisfied:

   — OneNumber Access is supported but only for Audio Add-On endpoints. Multimedia endpoints are not supported.

   — UCC endpoints cannot be speed matched. Therefore, if the *Type* field on the form contains **UCC**, the corresponding *Rate* field must be blank.

   — Ensure that the rate of the Dedicated Access (DA) endpoints is compatible with the administered endpoint rate. Also, ensure that the rate of the specified access endpoints for the dial-out numbers contained in the *Dial-Out #1* and *Dial-Out #2* fields is equal to or greater than the rate at which the endpoint is administered.

   — Ensure that the *Type* field contains **BOND** or **BCAS** if the *Rate* field contains **112k**, **128k**, **224k**, **336k**, **448k**, or **672k**.

   — If the *Rate* field contains **2x56k** or **2x64k,** populate the *Dial-Out #2* field as well as the *Dial-Out #1* field. If the *Rate* field does not contain either of these values, ensure that the *Dial-Out #2* field is blank.

3. Enter into the *Rate* field the rate at which the endpoint is joining the Speed Match conference. Values include **1x56k**, **1x64k**, **2x56k**, **2x64k**, **112k**, **128k**, **224k**, **256k**, **336k**, **384k**, **448k**, **512k**, **672k**, **768k**, and blank.

   The MCU allows a maximum of four rates at which an endpoint can join a Speed Match conference. Any rate entered into this field that exceeds this total is denied.

   NOTE:
   This field must be blank if *Speed Match* is set to **n**. The field must also be blank for Audio Add-On and UCC endpoints. The field may be blank for endpoints at the primary transfer rate as indicated in the *Bandwidth* field.

Administering H.263/H.261 Transcoding

To administer H.263/H.261 Transcoding (H.320 Translator), complete the fields in the forms identified in the following table. Appropriate page references to the forms are provided. Detailed instructions for completing the forms are provided after the table

Table 2-57. Required Forms and Fields for H.263/H.261 Transcoding

Form	Field	Page Reference
Customer-Options	■ *Speed Match (SM)* ■ *Conferencing Suite*	page 2-22
Conference Record	■ *Video Mode* ■ *Continuous Presence* ■ *Speed Match* ■ *H.263/H.261 Transcoding*	page 1-3

To enable H.263/H.261 Transcoding, populate the forms discussed in the following sections as directed.

Customer-Options Form

The system administrator must set either the *Speed Match (SM)* or *Conferencing Suite* field in the form to a nonzero value.

NOTE:
The right-to-use (RTU) for SM applies to either full-screen Speed Match or full-screen H.263/H.261 transcoding. Similarly, the RTU for Conferencing Suite applies to any one of the following: full-screen SM, full screen H.263/H.261 transcoding, CPP, CPP with SM, or Executive Conferencing.

The following table summarizes the RTUs for several options (fields) on the Customer-Options form.

Table 2-58. RTUs for Several Options on the Customer Options Form

Option (Field)	Meaning
Continuous Presence/Executive Conf	--- Continuous Presence *Plus,* or --- Executive Conferencing
Speed Match (SM)	--- full-screen Speed Match, or --- full-screen H.263/H.261 Transcoding at Same Speed
Conferencing Suite	--- CPP, or --- full-screen Speed Match, or --- full-screen H.263/H.261 Transcoding at Same Speed, or --- CPP with Speed Match, or --- Executive Conferencing
Executive Conference	--- Executive Conferencing only

Conference Record Form

Complete the following steps:

1. Ensure that the *Video Mode* field in the form contains **H.263/1-down_only** or **H.263/1-w/upgrade**.
2. Ensure that both the *Continuous Presence* field and the *Speed Match* field in the form contain **n**.
3. Set the *H.263/H.261 Transcoding* field in the form to **y**. This field enables H.263/H.261 transcoding for endpoints operating at the same transfer rate. Value **y** indicates that H.263 and H.261 endpoints can coexist within the same conference.

Administering On-Screen Display

To administer On-Screen Display, complete the fields in the forms identified in the following table. Appropriate page references to the forms are provided. Detailed instructions for completing the forms are provided after the table

Table 2-59. Required Forms and Fields for On-Screen Display

Form	Field	Page Reference
Customer-Options	■ *Notification Package*	page 2-22
Conference Record	■ *Class* ■ *Continuous Presence* ■ *Speed Match* ■ *Entry/Exit Alert* ■ *Warning Alert*	page 1-3

To administer On-Screen Display (OSD), complete the forms discussed in the following sections as directed.

Customer-Options Form

The system administrator must set the *Notification Package* field to **y**.

Conference Record Form

Complete the following steps:

1. If you want to provide conference entry/exit notifications, populate the *Entry/Exit Alert* field, as appropriate. Enter **tone** (default) to provide an audible tone indicating that the conferee is joining or exiting the conference. Enter **text** to provide a text message indicating that the conferee is joining or exiting the conference. Enter **ANI** to display the Automatic Number Information (ANI) code of the endpoint for the associated conferee. Enter **text/tone** to provide both an audible tone and a text message indicating that the conferee is joining or exiting the conference. Enter **ANI/tone** to provide both an audible tone and the ANI code of the endpoint for the associated conferee.

   NOTE:
   Values **text**, **text/tone**, **ANI**, and **ANI/tone** cannot be administered unless the conference is a non-SM CPP conference. Therefore, the *Speed Match* field must contain **n** and the *Continuous Presence* field must contain **y**.

Value **text/tone** for entry/exit alert is not allowed if OneNumber Access is being used on the conference for either Audio Add-On or multimedia endpoints.

Enter **none** to provide no notification of a conferee joining or exiting the conference.

NOTE:
None of the values for the field can be changed dynamically during an active or in-use conference.

2. If you want to provide an end-of-conference warning notification, populate the *Warning Alert* field, as appropriate. This field appears if your MCU model has the Notification Package feature and if *Class* contains **reserved** or **file**. Enter **tone** (default) to provide the traditional 10-minute audible warning tone. Enter **text** to provide the text message for both the 10-minute and two-minute warnings.

   NOTE:
   Values **text** and **text/tone** cannot be administered unless the conference is a non-SM CPP conference. Therefore, the *Speed Match* field must contain **n** and the *Continuous Presence* field must contain **y**.

   Enter **none** to provide no conference end notification.

   NOTE:
   None of the values for the field can be changed dynamically during an active or in-use conference.

   If the conference end time is extended dynamically during a conference in response to the 10-minute notification, and if the field contains a valid value other than **none**, a subsequent 10-minute notification followed by a two-minute notification are provided starting 10 minutes prior to the newly established conference end time.

Administering RS366 Dialing

To administer RS366 Dialing, complete the fields in the forms identified in the following table. Appropriate page references to the forms are provided. Detailed instructions for completing the forms are provided after the table

Table 2-60. Required Forms and Fields for RS366 Dialing

Form	Field	Page Reference
Customer-Options	■ *DI Direct Connect*	page 2-22
Conference Record	■ *Type*	page 1-3
Access Endpoint	■ *Interface Type* ■ *Timing Mode* ■ *RS366 Dialing*	page 2-213

To administer RS366 Dialing, populate the forms discussed in the following sections as directed.

Customer-Options Form

The system administrator must populate the *DI Direct Connect* field, as appropriate. This field reserves MCU resources for point-to-point calls made from V.35, RS449, and EIA-530 endpoints using RS366 Dialing. Values for this field equate to the number of DI circuit packs (ports) used for point-to-point calls, and they include the range **0** through **32**. The number of MCU resources depends on the 384k transfer rate.

Conference Record Form

Ensure that the *Type* field contains value **DA** or **P64**.

Access Endpoint Form

Complete the following steps:

1. Populate the *Interface Type* field, as appropriate. Values include **V.35**, **RS449**, and **EIA530**.
2. Enter value **DCE** into the *Timing Mode* field.
3. Use the *RS-366 Dialing* field to indicate whether access endpoints can receive dial strings.

Administering Far-End Camera Control

To enable this feature, first access the Customer-Options form and set the *Chair Control Suite* field to **y**. Then access the Conference Record form and enter **y** into the *Far End Camera Control* field. Also, ensure that both the *Continuous Presence* field and the *T.120 Data Mode* field are set to **n** and that the *Cascade Mode* field is blank. (Basic/Enhanced?: Required value: **enhanced** or **h263_enh**)

NOTE:
Value **basic** can be administered for endpoints in FECC conferences. However, these endpoints do not have access to FECC.

Administering ISDN-BRI Trunk Interface

To administer ISDN-BRI Trunk Interface, complete the fields in the forms identified in the following table. Appropriate page references to the forms are provided. Detailed instructions for completing the forms are provided after the table

Table 2-61. Required Forms and Fields for ISDN-BRI Trunk Interface

Form	Field	Page Reference
Customer-Options	■ *Maximum Port Capacity* ■ *BONDed Use Same DS1*	page 2-22
Optional Features	■ *ISDN-BRI Trunks*	page 2-246
Feature Access Code	■ *ISDN Access Code*	page 2-213
System Capacities	■ *ISDN-PRI Ports*	page 2-246
BRI Trunk	(See the following “BRI Trunk Form” section.)	page 2-246
Trunk Group	(See the following “Trunk Group Form” section.)	page 2-251, page 2-122

To administer ISDN-BRI Trunk Interface, populate the forms discussed in the following sections as directed.

Customer Options Form

Populate the appropriate *Maximum Port Capacity* fields. Also, set the *BONDed Use Same DS1* field to **n**.

Optional Features Form

Ensure that the *ISDN-BRI Trunks* field in the Optional Features form is set to **y**. If the field is disabled, the BRI Trunk form is not accessible.

Feature Access Code Form

Populate the *ISDN Access Code* field with a four-digit code number. The number can contain any digits 0 through 9 along with the # and * symbols in any position (for example, ***256**).

System Capacities Form

Populate the I*SDN-BRI Ports* field with a value indicating the appropriate number of ISDN-BRI ports.

BRI Trunk Form

The BRI Trunk form is used to support the TN2185 circuit pack to enable Trunk-Side BRI. The following table lists the commands that can be used with the BRI Trunk form.

Table 2-62. Trunk-Side BRI Administration Commands

Action	Object	Qualifier
add	bri-trunk-board	PPCSS
change	bri-trunk-board	PPCSS
display	bri-trunk-board	PPCSS
remove	bri-trunk-board	PPCSS

An example of Page 1 of the BRI Trunk form for Trunk-Side BRI appears as follows.

```
add bri-trunk-board 01a09                                Page 1 of 2

                        ISDN-BRI TRUNK CIRCUIT PACK

              Location: 01A09                              Name: _________________
  Interface Companding: a-law_     DCP/Analog Bearer Capability: 3.1kHz
 T3 Timer Length (sec): 15_


Port  Interface   Side  Cntry/Peer  TEI    Time    Invert  Synch    Layer 1  Detect
                         Protocol          Fill    Bits?   Source?  Stable?  Slips?
  1:  user_______          12__     0___   flags     n       n        y       n
  2:  user_______          etsi     0___   ones_     y       y        y       y
  3:  user_______          2___     0___   flags     y       n        y       n
  4:  peer-slave_  b       QSIG     0___   flags     n       y        y       n
  5:  peer-slave_  a       QSIG     0___   ones_     n       n        y       n
  6:  ___________          ____     0___   ones_     n       n        y       n
  7:  ___________          ____     0___   ones_     n       n        y       n
  8:  ___________          ____     0___   ones_     n       n        y       n
```

Figure 2-88. BRI Trunk Form (ISDN-BRI Trunk Interface) — Page 1

To administer Trunk-Side BRI on the form, follow these procedures:

1. At the *command:* prompt, enter **change bri-trunk-board *PPCSS*,** where ***PPCSS*** is the location of the circuit pack. Specifically, ***PP*** is the two-digit port network number, ***C*** is the one-digit carrier identification, and ***SS*** is the two-digit slot number.
2. Verify that the screen displays the *BRI Trunk* form (see *BRI Trunk Form* for a list of fields and values).
3. Use TAB or RETURN to advance to the fields you want to change.
4. Submit the form.
5. To verify the form, enter **display bri-trunk-board *PPCSS***.

The following list identifies the fields on the BRI Trunk form and the possible values.

- *Location*

  Read-only field that shows the location of the TN2185 circuit pack in the form ***PPCSS***.

- *Name*

  15-character field that may be used to identify the circuit pack. Default is blank.

- *Interface Companding*

  Indicates the companding algorithm expected by the system at the far end. This parameter must be sent to the TN2185 circuit pack. Values include **a-law** (default) and **mu-law**. Blank is not a valid entry for this field.

  NOTE:
  You can change this field only if the circuit pack is busied out or if all ports are unadministered.

- *DCP/Analog Bearer Capability*

  Indicates how to encode the Bearer Capability IE for an outgoing call originated by a DCP or analog endpoint. Blank is not a valid entry for this field. Values include **3.1kHz** (default) and **speech**.

- *T3 Timer Length*

  Informs the TE side how long to wait for an inactive Layer 1 to become active. This parameter must be sent to the TN2185 circuit pack. Values include **1** through **127** seconds in increments of one second. The default is **15** seconds. Blank is not a valid entry for this field.

  NOTE:
  You can change this field only if the circuit pack is busied out or if all ports are unadministered.

Port parameters include the following:

NOTE:
The user is blocked from changing a port parameter unless the associated port is busied out or unadministered.

- *Interface*

  Tells the call processing software whether a particular port is connected to a user/network or a peer interface. For the TN2185 circuit pack, values include **user**, **network**, **peer-master**, **peer-slave**, and blank (default). Value **peer-slave** can be entered only if the QSIG Basic Call Setup optional feature has been enabled. Value **peer-master** can be entered only if *Cntry/Peer Protocol* is set to **QSIG**.

  NOTE:
  If the field contains a valid value when the form is submitted, the contents of all other fields for the associated port are validated and the port is enabled. If the field is blank when the form is submitted, all other fields for the port are reset to their default values, and the port is disabled. A port cannot be disabled if it is already administered as a member of a trunk group.

- *Side*

  Determines how glare conditions should be handled when *Interface* contains **peer-slave**. The field is not administrable when *Interface* contains **user**. Values include a and b (default). Blank is not a valid entry for this field.

- *Cntry/Peer Protocol*

  Tells the call processing software which ISDN protocol standard to apply. Values include **1** through **23**, **etsi**, and **QSIG**. The default is blank; however, the user cannot leave this field blank if *Interface* is set to a valid, non-blank value. Whenever *Interface* is set to **peer-slave**, *Cntry/Peer Protocol* must be set to **QSIG**. Whenever *Cntry/Peer Protocol* is set to **1**, *Protocol Version* is implicitly set to **b** even though the field is not administrable by the customer.

- *TEI*

  Layer 2 parameter. Values include **0** (default) and **auto**.

- *Time Fill*

  Identifies the bit pattern sent on the D-channel between valid LAPD packets. Values include **flags** and **ones** (default). Blank is not a valid entry for this field.

- *Invert Bits*

  Provides information needed by the TN2185 circuit pack. Enter **y**. Default is **n**.

- *Synch Source*

  Provides information needed by the TN2185 circuit pack. Enter **y**. Default is **n**.

- *Layer 1 Stable*

  Tells call processing and maintenance software whether to expect the network to drop Layer 1 when the BRI port is idle. Values include **yes** (default) and **no**. Blank is not a valid entry for this field.

- *Detect Slips*

  Tells maintenance software whether slips reported by the BRI port should be logged. Values include **no** (default) and **yes**. Blank is not a valid entry for this field.

Page 2 of the form deals with port parameters and appears as follows.

```
add bri-trunk-board 01a09                                Page 2 of 2


                        ISDN-BRI TRUNK CIRCUIT PACK

Port  XID      Endpt       SPID          Endpt      Max
      Test?    Init?                     Init?    NCA TSC
  1:    y        n                          __       20
  2:    y        y      908957200001        01       0_
  3:    y        y      0001________        __       0_
  4:    n        n                          __       0_
  5:    n        y      625761449___        __       0_
  6:    n        n                          __       0_
  7:    n        n                          __       0_
  8:    n        n                          __       0_
  9:    n        n                          __       0_
 10:    n        n                          __       0_
 11:    n        n                          __       0_
 12:    n        n                          __       0_
```

Figure 2-89. BRI Trunk Form, Page 2

NOTE:
The user is blocked from changing the values for *Endpt Init* and *SPID* unless the corresponding port is busied out or unadministered. All other fields on this page can be changed even if the corresponding port is active.

- *XID Test*

  Indicates whether the far end supports the Layer 2 XID test. Values include **yes** and **no** (default). Blank is not a valid entry for this field.

- *Endpt Init*

  Indicates whether the far end supports endpoint initialization. Values include **yes** and **no** (default). Blank is not a valid entry for this field.

  NOTE:
  If this field is set to **no**, *SPID* and *Endpt ID* must be blank. If this field is set to **yes**, *SPID* must not be blank.

- *SPID*

  12-digit field containing the SPID expected by the far end. Values include any string containing one to 12 digits and blank (default). Leading zeros are significant and are not ignored.

  NOTE:
  If *Endpt Init* is set to **no**, this field must be blank. If *Endpt Init* is set to **yes**, this field must not be blank.

- *Endpt ID*

  2-digit field containing the endpoint identifier expected by the far end. Values include any 2-digit string from **00** through **62** and blank (default). Leading zeroes are significant and are not ignored.

  NOTE:
  If *Endpt Init* is set to **no**, this field must be blank

- *Max NCA TSC*

  2-digit field providing the maximum number of Non-Call-Associated Temporary Signaling Connections allowed on the associated BRI D-channel. Values range from **0** (default) to **63**. Blank is not a valid entry for this field.

Trunk Group Form

Populate the form fields identified in the following list as directed.

- *Group Type*: **isdn**
- *Used for DCS*: **y**

  NOTE:
  However**,** if *DCS Signaling* is set to **d-chan**, *DCS Signaling* should be reset to **bx.25**.

- *Wideband Support*: **no**
- *Trunk Hunt*: blank
- *Outgoing Channel ID Encoding*: if *Used for DCS* is enabled, **exclusive**; if *Used for DCS* is disabled, **preferred**.

Administering Full Tie-Trunk Capabilities Using BRI Trunks

Full tie-trunk capabilites using BRI trunks are available via the TN556 circuit pack and by administering the BRI Trunk form.

BRI Trunk Form

The BRI Trunk form is used to support the TN556 circuit pack. The following table lists the commands that can be used with the BRI Trunk form.

Table 2-63. Trunk-Side BRI Administration Commands

Action	Object	Qualifier
add	bri-trunk-board	PPCSS
change	bri-trunk-board	PPCSS
display	bri-trunk-board	PPCSS
remove	bri-trunk-board	PPCSS

An example of the BRI Trunk form for full tie-trunk capabilities appears as follows.

```
add bri-trunk-board 01a09                                Page 1 of 2

                        ISDN-BRI TRUNK CIRCUIT PACK

              Location: 01A09                              Name: _________________
  Interface Companding: a-law_     DCP/Analog Bearer Capability: 3.1kHz


Port  Interface   Side  Cntry/Peer  TEI    Time    Invert
                         Protocol          Fill    Bits?
  1:  network____          12__     0___   flags     n
  2:  network____          etsi     0___   ones_     y
  3:  network____          2___     auto   flags     y
  4:  peer-master  b       QSIG     0___   flags     n
  5:  peer-master  a       QSIG     auto   ones_     n
  6:  ___________          ____     0___   ones_     n
  7:  ___________          ____     0___   ones_     n
  8:  ___________          ____     0___   ones_     n
  9:  ___________          ____     0___   ones_     n
 10:  ___________          ____     0___   ones_     n
 11:  ___________          ____     0___   ones_     n
 12:  ___________          ____     0___   ones_     n
```

Figure 2-90. BRI Trunk Form (Full Tie-Trunk Capabilities)

To administer full BRI tie-trunk capabilities on the form, follow these procedures:

1. At the *command:* prompt, enter **change bri-trunk-board *PPCSS*,** where ***PPCSS*** is the location of the circuit pack. Specifically, ***PP*** is the two-digit port network number, ***C*** is the one-digit carrier identification, and ***SS*** is the two-digit slot number.
2. Verify that the screen displays the *BRI Trunk* form (see *BRI Trunk Form* for a list of fields and values).
3. Use TAB or RETURN to advance to the fields you want to change.
4. Submit the form.
5. To verify the form, enter **display bri-trunk-board *PPCSS***.

The following list identifies the fields on the BRI Trunk form and the possible values.

- *Location*

  Read-only field that shows the location of the TN556 circuit pack in the form ***PPCSS***.

- *Name*

  15-character field that may be used to identify the circuit pack. Default is blank.

- *Interface Companding*

  Indicates the companding algorithm expected by the system at the far end. This parameter must be sent to the TN556 circuit pack. Values include **a-law** (default) and **mu-law**. Blank is not a valid entry for this field.

  NOTE:
  You can change this field only if the circuit pack is busied out or if all ports are unadministered.

- *DCP/Analog Bearer Capability*

  Indicates how to encode the Bearer Capability IE for an outgoing call originated by a DCP or analog endpoint. Blank is not a valid entry for this field. Values include **3.1kHz** (default) and **speech**.

Port parameters include the following:

NOTE:
The user is blocked from changing a port parameter unless the associated port is busied out or unadministered.

- *Interface*

  Tells the call processing software whether a particular port is connected to a user/network or a peer interface. For the TN556 circuit pack, values include **network**, **peer-master**, and blank (default). Value **peer-master** can be entered only if the QSIG Basic Call Setup feature has been enabled.

  NOTE:
  If the field contains a valid value when the form is submitted, the contents of all other fields for the associated port are validated and the port is enabled. If the field is blank when the form is submitted, all other fields for the port are reset to their default values, and the port is disabled. A port cannot be disabled if it is already administered as a member of a trunk group.

- *Side*

  Determines how glare conditions should be handled. The field is not administrable when *Interface* is set to **network**. Values include **a** (default) and **b**. Blank is not a valid entry for this field.

- *Cntry/Peer Protocol*

  Tells the call processing software which ISDN protocol standard to apply. Values include **2** through **23**, **etsi**, and **QSIG**. The default is blank; however, the user cannot leave this field blank if *Interface* is set to a valid, non-blank value. Whenever *Interface* is set to **peer-master**, *Cntry/Peer Protocol* must be set to **QSIG**. Whenever *Cntry/Peer Protocol* is set to **10**, **12**, **13**, or **etsi**, *Protocol Version* is implicitly set to **b** even though the field is not administrable by the customer.

- *TEI*

  Layer 2 parameter. Values include **0** (default) and **auto**.

- *Time Fill*

  Identifies the bit pattern sent on the D-channel between valid LAPD packets. Values include **flags** and **ones** (default). Blank is not a valid value for this field.

- *Invert Bits*

  Provides information needed by the TN556 circuit pack. Enter **y**. Default is **n**.

Administering ClearPresence @716 Audio Coder

When administering a customer's system, the VTC should use the **list configuration all** command to verify that no TN788B circuit packs are in the system.

To administer ClearPresence @716, complete the fields in the forms identified in the following table. Appropriate page references to the forms are provided. Detailed instructions for completing the forms are provided after the table

Table 2-64. Required Forms and Fields for ClearPresence @716

Form	Field	Page Reference
Customer-Options	■ *ClearPresence @716*	page 2-22
Conference Record	■ *Audio Mode*	page 1-3

Customer Options Form

Administer this form as follows.

- *ClearPresence @716*

  Indicates whether the right-to use (RTU) fee required for ClearPresence @716 has been paid by the customer. Values include **y** and **n** (default). Also, indicates the set of audio modes available for conferences and the rationale for each mode. For value **y**, the audio modes supported include ClearPresence @716, G.722, G.728, G.711, auto-down_only, and auto-w/upgrade. Value **y** also indicates that G.728/G.722 interworking is available in Speed Match conferences. For value **n**, all of the audio modes identified in this bullet item except ClearPresence @716 are supported.

  NOTE:
  If a TN788B circuit pack is inserted after value **y** is entered into the *ClearPresence @716* field, an alarm occurs.

Conference Record Form

Administer this form as follows.

- *Audio Mode*

  To establish the endpoint as a ClearPresence @716 endpoint, enter value **ClearPres716**.

  NOTE:
  If the Lucent Technologies ClearPresence @716 right-to-use (RTU) option *is not enabled*, the default value **auto-w/upgrade** indicates that the MCU will automatically select either the **G.711**, **G.722** or **G.728** audio mode, depending on the bandwidth of the conference and of the participating video endpoints. (These three rates are fixed values.) Also, this value indicates that the MCU will automatically upgrade a conference where the audio mode has been downgraded whenever the endpoint causing the downgrade drops out of the conference. Value **auto-down_only** indicates that the MCU will upgrade the audio mode only of a conference that has been downgraded and after all endpoints drop.

  Refer to Table B-11 on page B-64 for a description of the per-conference audio video and capabilities received by endpoints per the ClearPresence @716 RTU and other factors for all valid *Audio Mode* field values.

Checking MCU Performance

3

Overview

The MCU provides a variety of reports upon request. These reports provide details about the performance and status of trunks and MCU ports.

The optional system printer, not the local printers connected to the MCU-MT or MCU-ST, automatically prints all reports when the command includes the schedule qualifier. The system printer is remotely connected to the MCU via a 7400B data module.

If you do not have a system printer, individual reports can be printed on local printers by adding the print qualifier to the command line.

Scheduling Reports

The schedule option appears as a qualifier for many MCU list, display and test commands. Whenever a command containing the schedule qualifier is executed, a J*ob Id* is generated and a Report Scheduler screen appears so you can change the *Print Interval*. A maximum of 50 *Job Ids* can be scheduled for printing on the system printer.

The *Print Interval* field specifies when each report is printed (**immediate**, **deferred**, or **scheduled**). The **immediate** option prints the report on-demand and is the default.

The **deferred** option schedules the report for a single printing at some day or time in the future. Once the report is printed, the *Job Id* is automatically deleted.

Both the **deferred** and **scheduled** options provide two additional fields for *Print TIme* and *Days of Week* to specify when the report is to be printed. The **scheduled** option differs from the **deferred** option because the specified report continues to print at the scheduled time until it is removed from the Report Scheduler.

Table 3-1 lists the commands that relate to the Report Scheduler feature.

Table 3-1. Report Scheduler Administration Commands

Action	Object	Qualifier
list	report-scheduler	
change	report-scheduler xx*	
remove	report-scheduler xx	

* xx = the Job ID

When a scheduled report needs to be changed with the **change report-scheduler** command, the *Job Status* field indicates whether the report is scheduled to be printed (**print-next**), currently printing (**printing**), finished printing (**printed**), or not scheduled to print (**waiting**) during the current 15-minute interval. To print a scheduled report immediately, change the *Print Interval* field to **immediate**.

Recent Change History

The Recent Change History Report allows a history of the most recent administration and maintenance changes to be viewed or printed. This report may be used for diagnostic or informational purposes.

The MCU maintains a log in the software buffer of up to 250 of the most recent administration and maintenance commands. This log is called the transaction log. The commands must be data-affecting and are called data commands (see Table 3-2).

The Recent Change History Report shows the command, date and time, MCU port (such as, MCU-MT, INADS, or CDR) and the login used to enter the command. The transaction log can be displayed on screen or printed using the following command:

Table 3-2. Recent Change History Administration Commands

Action	Object	Qualifier
list	history	print

The following list shows the data commands available on the MCU:

Table 3-3. MCU Data Commands

Command	Command	Command
add	enable	reset
busyout	get	rp
change	list	save
clear	mark	set
disable	monitor	status
display	recycle	test
download	release	upload
duplicate	remove	wp

Trunk Group Measurements Report

Three reports provide traffic measurements for trunk groups and Call-By-Call (CBC) reports. The Trunk Group CBC Report displays the last-hour traffic data for any specified trunk group. The Trunk Group Summary Report tracks the activity of the trunk group. The Trunk Group Measurements Hourly Report captures hourly trunk group usage.

Table 3-4 shows the commands that relate to obtaining trunk group measurements.

Table 3-4. Trunk Group Measurements Administration Commands

Action	Object	Qualifier
change	meas-selection trunk-group	print or schedule
display	meas-selection trunk-group	print or schedule
list	measurements trunk-group hourly ##	print or schedule
list	measurements trunk-group summary	yesterday-peak, today-peak, last-hour, print, or schedule
list	measurements cbc-trunk-group <number>	last-hour, print, or schedule

Trunk Group Measurements Call-By-Call Report

The Trunk Group Measurements CBC Report monitors the trunk group and determines if the usage selected by the Usage Allocation Plan (UAP) is optimal.

Each UAP specifies the network services and features assigned to the trunk group. By implementing the UAP, the CBC trunk group can be optimized within certain limits without involving any of the Inter-Exchange Carrier/Local Exchange Carrier (IXC/LEC) network services personnel. Each plan specifies the minimum number of reserved channels and maximum number of channels each service or feature may use at a given time.

The UAP may be administered as **fixed** or **scheduled**. With the **fixed** method, a specified plan remains in effect continuously. With the **scheduled** method, two or three UAPs may be scheduled to vary during the week or time of day. Before this report can be analyzed, the intent of the UAP must be known.

The report appears as follows:

```
list measurements cbc-trunk-group 99 last-hour
Switch Name: Cust_Switch_Name
Switch Name:                                  Date: 2:15 pm WED DEC 13, 1995

                        CBC TRUNK GROUP MEASUREMENTS
                   Peak Hour For CBC Trunk Group 99 : 1300
Queue Size          : 40                            Usage Allocation Plan Used
Calls Queued        : 23                   Plan Number:  0  0   0   0   0   0
Queue Overflow      : 0                       Duration:  0  0   0   0   0   0
Queue Abandonments  : 4
Out of Service      : 0
                  Min  Max Meas Total Total Inc.  Ovf  Ovf  Ovf  %   %   %Out
Service/Feature   Chn  Chn Hour Usage Seize Seize  TG  S/F  Max TBM ATB  BLK
  outwats-bnd       5   12 1300   240   333   0     0   0   23   10   2    1
  sdn               4    8 1300    40    30  22     0   0    0   62   1    0
  other             0   20 1300    70    41  36     0   0    0    0   0    0
```

Figure 3-1. Trunk Group Measurements Call-By-Call Report

The following list identifies the data collected by the Trunk Group Measurements CBC Report:

- *Queue Size*

  This field is set to **0**; no queue is administered.

- *Calls Queued*

  This field is not used.

- *Queue Overflow*

  This field is not used.

- *Queue Abandonments*

  This field is not used.

- *Out of Service*

  The number of trunks in the trunk group that are out-of-service at the time the measurements are taken. An individual trunk may be taken out-of-service automatically by the MCU if too many errors occur or by maintenance personnel to run tests.

- *Usage Allocation Plan Used*

  N/A on the MCU.

- *Service/Feature*

  The name of up to 10 services or features and the special identifier “other” for which the associated measurements are being reported. The following services can be administered in a UAP: ACCUNET digital service, SDN and SDDN. The identifier “other” is used to report activity for any service or feature not specified in the current UAP. If no UAP is in effect, no report is made. If public network access is to an LEC or IXC other than Lucent Technologies, that carrier’s ISDN services or features are listed, provided they have been administered as a user-defined service or feature.

- *Min Chn*

  The minimum number of channels in the ISDN-PRI CBC trunk group allocated to the specified service or feature at the time the measurements were collected.

- *Max Chn*

  The maximum number of channels in the ISDN-PRI CBC trunk group allocated to the specified service or feature at the time the measurements were collected.

- *Total Usage*

  The sum of time, in hundred-call-seconds (CCS) for all channels that were used by the specified service or feature during the measurement interval (60 minutes).

- *Total Seize*

  The total number of incoming calls that requested the specified service or feature through the ISDN-PRI CBC trunk group.

- *Inc. Seize*

  The total number of incoming calls that requested the specified service or feature through the ISDN-PRI CBC trunk group.

- *Tan Seize*

  This field does not apply. It should remain at **0**.

- *Ovf TG*

  This field does not apply. It should remain at **0**.

- *Ovf S/F*

  This field is not used.

- *Ovf Max*

  This field is not used.

- *% TBM*

  This field is not used.

- *% ATB*

  The percentage of time (**0** to **100** percent) during the measurement interval when the specified service or feature could not get a channel because all trunks in the ISDN-PRI trunk group were busy on a call or busied-out by maintenance, the service was above its minimum and all available trunks were reserved, or the specified service or feature was at its maximum number of channels.

- *% BLK*

  This field does not apply.

Trunk Group Summary Report

The Trunk Group Summary Report appears as follows:

```
list measurements trunk-group summary last-hour
Switch Name: Cust_Switch_Name                    Date: 1:58 pm MON DEC 11, 1995
                          TRUNK GROUP SUMMARY REPORT
                       Peak Hour For All Trunk Groups: 1000
Grp Grp  Grp  Grp Meas Total Total Inc.  Grp  Que  Calls Que Que Out   %  %Out
No. Size Type Dir Hour Usage Seize Seize Ovfl Size Qued  Ovf Abd Serv ATB Blk
20  1    did  inc 1200 0     0     0     0    0    0     0   0   0    0   *
30  1    tie  two 1200 36    0     0     0    0    0     0   0   0    100 0
37  22   tand two 1200 598   179   0     0    0    0     0   0   0    0   0
38  23   isdn two 1200 171   654   0     0    0    0     0   0   0    0   0
39  22   isdn two 1200 270   762   762   0    0    0     0   0   0    0   0
40  5    co   two 1200 61    32    0     6    0    0     0   0   0    6   15
```

Figure 3-2. Trunk Group Summary Report

The following list identifies the data collected by the Trunk Group Summary Report:

- *Peak Hour for All Trunk Groups*

  Peak hour and busy hour are one in the same. The peak hour is the highest usage for each day regardless of the average across days.

- *Grp No.*

  A number that identifies each trunk group associated with the displayed data.

- *Grp Size*

  The number of administered trunks in a trunk group.

- *Grp Type*

  The type of trunk (**isdn-pri** or **tie**) in the trunk group.

- *Grp Dir*

  The direction (**incoming**) of the trunk group.

- *Meas Hour*

  The hour (using the 24-hour clock) in which the measurement was taken.

- *Total Usage*

  The total time the trunks are busy during the one-hour measurement period.

- *Total Seize*

  The number of times a trunk in the trunk group was seized. The call may or may not have been completed.

- *Inc. Seize*

  The total number of times a trunk in the trunk group was seized to receive an incoming call. The call may or may not have been completed.

- *Grp Ovfl*

  This field is not used.

- *Que Size*

  This field is not used.

- *Calls Qued*

  This field is not used.

- *Que Ovf*

  This field is not used.

- *Que Abd*

  This field is not used.

- *Out Serv*

  The total number of trunks in the trunk group that are out-of-service (listed as **maintenance busy**) at the time data was collected. An individual trunk may be taken out-of-service by the MCU when an excessive number of errors occurs or by maintenance personnel to run diagnostic tests. If the trunks were taken out-of-service by the MCU, it may be appropriate to notify maintenance personnel.

- *% ATB*

  The percentage of time that all trunks in the trunk group were simultaneously in use during the measurement interval. In use means all trunks were busy either processing calls or they were busied-out by maintenance. A high number in this field probably indicates calls are being blocked.

- *% Out Blk*

  This field is not used.

Trunk Group Hourly Report

NOTE:
To generate this report, the number of trunks you want measured hourly must be entered in the *Trunk Group Number* field of the Trunk Group Measurement Selection. If no selections are made, no measurements are taken.

The following list identifies the data collected by the Trunk Group Hourly Report:

- *Grp No.*

  A number that identifies each trunk group associated with the displayed data.

- *Grp Size*

  The number of administered trunks in a trunk group.

- *Grp Type*

  The type of trunk (**isdn-pri** or **tie**) in the trunk group.

- *Grp Dir*

  The direction (**incoming**) of the trunk group.

- *Que Size*

  This field is not used.

- *Total Usage*

  The total time the trunks are unavailable due to the trunks being busy, busied-out by maintenance, or any other reason.

- *Maint Usage*

  The total usage of trunks in this trunk group for maintenance busy or any other non-call situation where trunks are unavailable.

- *Total Seize*

  The total number of seizures on the trunk group.

- *Inc. Seize*

  The total number of times a trunk in the trunk group was seized to receive an incoming call. The call may or may not have been completed.

- *Tandem Seize*

  This field is not used.

- *Grp Ovfl*

  This field is not used.

- *Calls Qued*

  This field is not used.

- *Que Ovf*

  This field is not used.

- *Que Abd*

  This field is not used.

- *Out Serv*

  The total number of trunks in the trunk group that are out-of-service (listed as **maintenance busy**) at the time data was collected. An individual trunk may be taken out-of-service by the MCU when an excessive number of errors occurs or by maintenance personnel to run diagnostic tests. If the trunks were taken out-of-service by the MCU, it may be appropriate to notify maintenance personnel.

- *% ATB*

  The percentage of time that all trunks in the trunk group were simultaneously in use during the measurement interval. In use means all trunks were busy either processing calls or they were busied-out by maintenance. A high number in this field probably indicates calls are being blocked.

- *% Out Blk*

  This field is not used.

Trunk Outage Report

This report lists up to a maximum of five trunks in each trunk group that were out-of-service when sampled. The number of times the trunks were out-of-service when sampled is also given. The trunk outage data is kept for the current day, the previous day and the last hour.

Table 3-5 shows the command that relates to obtaining trunk outage measurements.

Table 3-5. Trunk Outage Measurements Administration Commands

Action	Object	Qualifier
list	measurements outage-trunk	yesterday, today, print, or schedule

Trunk Outage Measurements Report

The following list shows the data collected for the Trunk Outage Measurements Report:

- *Grp No.*

  A number that identifies each trunk group associated with the displayed data.

- *Grp Type*

  The type of trunk (**isdn-pri** or **tie**) associated with the accumulated data.

- *Grp Dir*

  Identifies the direction (**inc** for incoming only) of the trunk group.

- *Grp Size*

  The number of trunks in the trunk group.

- *Grp Mbr #*

  The number that identifies a specific trunk member in the group that is out-of-service.

- *#Sampled Outages*

  The number of times the group member is sampled as out-of-service over the period covered by the report (**yesterday**, **today**, or **last hour**). If there are no outages then no data is displayed. The sampling period is once per hour. If the report covers several hours but the column indicates a small number of outages, the trunk member may be providing intermittent service. Use the facility test calls feature to determine whether a specific trunk member is functioning. If you suspect the trunk is causing problems, use the ACA feature to monitor the particular trunk group. If the trunk is out-of-service, you may choose to refer the problem to maintenance personnel.

Trunk Group Traffic Report

The Trunk Group Status Report gives a current indication of the load on various trunk groups. For comparative analysis, the trunk members in the group that are active on calls are also displayed. Table 3-6 shows the command that relates to obtaining trunk status reports:

Table 3-6. Trunk Status Administration Commands

Action	Object	Qualifier
monitor	traffic trunk-groups	

Trunk Status Report

The following list shows the data collected for the Trunk Status Report:

- *#*

  Group number that identifies each trunk group.

- *S*

  Group size that shows the number of trunks administered for the trunk group.

- *A*

  Shows the number of trunk members in the group that are active on a call. Busied-out trunks are not active.

- *Q*

  This field is not used.

- *W*

  This field is not used.

DS1 (UDS1) Measurements

There are two performance measurements for DS1 (UDS1) links. The DS1 (UDS1) Link Performance Measurements Summary Report provides an indication of the quality of a DS1 (UDS1) link that connects to a DS1 (UDS1) interface circuit pack. The DS1 (UDS1) Link Performance Detailed Log Report lists errored events for the past 24 hours. The errored event records are listed for each 15-minute interval. Included are 96 records (for each 15-minute interval) dating from the current time back to 24 hours before the current interval, beginning with the oldest record.

Check these reports if you receive complaints from video endpoints about the stability of the far-end handshake, especially when the Status of Conference screen reports **Handshake** as the drop reason. (See Chapter 5 for troubleshooting details.)

Table 3-7. DS1 (UDS1) Measurements Administration Commands

Action	Object	Qualifier
clear	measurements-ds1 (uds 1)	
list	measurements ds1 (uds 1)	location, print, or schedule
list	measurements ds (uds)1-log	location, print, or schedule

DS1 (UDS1) Link Performance Measurements Summary

The following list shows the data collected on the DS1 (UDS1) Link Performance Measurements Summary form:

- *Counted Since*

  The date and time the counters were last cleared and restarted. The counters are set to 0 and start accumulating data when the MCU is administered or reinitialized. The current system time appears in the field once the system clock is set. An error message appears if the system clock has not been set following a reinitialization.

- *Number of Seconds Elapsed*

  The total number of 15-minute intervals (**0** to **96**) in the past 24 hours that have valid values. An invalid value is any 15-minute interval when the time was changed, a system reinitialization occurred or the DS1 (UDS1) circuit pack was removed.

- *Worst 15-Min Interval*

  The date, ending time, and count for the 15-minute period that contained the maximum count in each error category. If there are no errors, the field displays a **0** with the oldest time.

- *Total of 24-Hour Count*

  The total count in each error category for the last 24 hours (**0** to **65535**).

- *Current 15-Min Interval Count*

  The count-so-far in each error category for the15-minute interval in progress when the report was requested. If no errors have occurred in any of the categories during the current 15-minute interval, the respective field will contain a **0**. If the system is busy performing call processing functions and cannot respond within eight seconds, the field will show **N/A**.

DS1 (UDS1) Link Performance Detailed Log Report

The following list shows the data collected on the DS1 (UDS1) Link Performance Detailed Log Report form:

- *Date*

  The date on which the event occurred (**yesterday** or **today**).

- *Time*

  The ending times for the 15-minute sampling intervals.

- *Errored Second*

  The number of errored seconds (maximum of **900**) for the specified interval. An errored second is any second in which one or more data transmission errors occurred. **N/A** indicates the count for that interval was not available because there was no data.

- *Bursty Err Sec*

  The number of bursty errored seconds (maximum of **900**) for the specified interval. A bursty errored second is any second in which two to 319 data transmission errors occurred. **N/A** indicates the count for that interval was not available. An error count of this severity causes a minor alarm. Depending on local arrangements, it may be appropriate to notify maintenance personnel if they have not already been alerted.

- *Severely Err Sec*

  The number of severely errored seconds for the specified interval (maximum of **900**). A severely errored second is any second in which 320 or more data transmission errors occurred. An error count of this severity results in a major alarm. Depending on local arrangements, it may be appropriate to notify maintenance personnel if they have not already been alerted.

- *Failed Second*

  The number of failed seconds states for the specified interval (maximum of 900). A failed second state exists any time that ten or more consecutive severely errored seconds have occurred. An error count of this severity results in a major alarm. **N/A** indicates the count for this interval was not available. Depending on local arrangements, it may be appropriate to notify maintenance personnel if they have not already been alerted.

- *Valid Interval*

  Specifies whether or not a valid count is provided by the DS1 (UDS1) interface circuit pack. A value of **y** indicates all counts are valid for the interval. An invalid interval is any 15-minute interval during which the system clock was changed, a system reinitialization occurred, or the specified circuit pack was removed.

Hunt Group Measurements

There are three measurement reports for hunt groups. The Hunt Group Measurements Report monitors the hunt group. This report shows hunt group measurements for yesterday's peak hour, today's peak hour, and the last hour. A peak hour is the hour within a 24-hour period when there was the greatest usage for the specified day.

The Hunt Group Performance Report gives the slowest hourly average speed of answer for each hunt group. The Hunt Group Status Report provides an instantaneous indication of the load that is pending for various hunt groups. Table 3-9 lists the commands related to hunt group measurements.

Table 3-8. Hunt Group Measurements Administration Commands

Action	Object	Qualifier
list	measurements hunt-group	yesterday-peak, today-peak, last hour, print, or schedule
list	performance hunt group	yesterday, today, print, or schedule
monitor	traffic hunt-groups	

Hunt Group Measurements Report Form

The Hunt Group Measurements Report appears as follows:

```
list measurements hunt-group today-peak

                                                Date: 9:14 am WED DEC 20, 1995
                              HUNT GROUP MEASUREMENTS

Grp Grp              Grp  Grp  Meas Total  Calls Calls Que  Calls Time  Speed
No. Name             Size Type Hour Usage  Ans.  Aban. Size Que.  Avail Ans(sec)

1   Netcon Channels  4    ucd  800  0      0     0     0    0     144   0
```

Figure 3-3. Hunt Group Measurements Report

The following list shows the data collected on the Hunt Group Measurements Report form:

- *Grp No.*

  A number that identifies each hunt group.

- *Grp. Name*

  Name assigned, during administration, to the hunt group.

- *Grp Size*

  The number of extensions assigned to the hunt group.

- *Grp Type*

  The type of hunt group.

- *Meas Hour*

  The peak hours for the specified hunt group. Depending on the time of day the report is generated, today's peak hour data may vary.

- *Total Usage*

  The sum of all times that the members of a hunt group are busy on hunt group calls.

- *Calls Ans*

  The total number of all hunt group calls answered by the hunt group.

- *Calls Aban*

  The total number of calls which attempt to reach the hunt group but abandon the attempt before being answered.

- *Queue Size*

  This field is not used.

- *Calls Queued*

  This field does not apply.

- *Time Avail*

  The total time that the hunt group extensions are not in use but are available to receive hunt group calls during the measurement hour.

- *Speed of Answer*

  The average time interval (in seconds) from when the call first enters the hunt group until the call is answered by a hunt group member.

Hunt Group Performance Report

The following list shows the data collected on the Hunt Group Performance Report:

- *Grp No.*

  A number that identifies each hunt group.

- *Grp Size*

  The number of extensions assigned to the hunt group.

- *Grp Type*

  The type of hunt group.

- *Slowest Speed of Answer (sec)*

  A bar graph representation of the slowest hourly average speed of answer for the report interval (**yesterday** or **today**).

- *Speed Ans (sec)*

  The number of seconds corresponding to the slowest hourly average speed of answer (longest amount of time to answer) for the report interval (**yesterday** or **today**).

- *Meas Hour*

  The starting time (using the 24-hour clock) of the hour during which the data was recorded.

- *Daily Avg.*

  The number corresponding to the 24-hour daily average speed of answer for each hunt group.

Hunt Group Status Report

The Hunt Group Status report appears as follows:

```
monitor traffic hunt-groups

                        HUNT GROUP STATUS          9:18 am WED DEC 20, 1995

    #    S    A    Q    W    LCIQ        #    S    A    Q    W    LCIQ
    1    4    0    0    0    0

(#: Group; S: Grp Size; A: Active Members; Q: Q Length; W: Calls Waiting)
(LCIQ: Longest Call in Queue in seconds                                  )
```

Figure 3-4. Hunt Group Status Report

The following list shows the data collected for the Hunt Group Status Report:

- *#*

  Group number that identifies each hunt group.

- *S*

  Group size that shows the extensions assigned to the hunt group.

- *A*

  Shows the number of hunt group members that are active on a call.

- *Q*

  This field is not applicable.

- *W*

  This field is not applicable.

- *LCIQ*

  This field is not applicable.

Security Violations Reports

There are two security violation reports. The Security Violations Measurements Report measures the number of attempts at system entry through an invalid login. The Security Violations Status Report provides current status information on invalid system administration terminal login attempts and remote access violations.

Invalid login attempts and remote access violations are logged at the time they occurred. The login is entered and the port that was accessed during the failed login attempt is recorded. For remote access violations, the trunk group number is also reported. A total of 16 entries are maintained for invalid access. The data displayed by this report is refreshed every 30 seconds.

NOTE:
If login security violations occur, change the password for the login immediately.

Table 3-10 lists the commands that are used to obtain security violation reports.

Table 3-9. Security Violation Administration Commands

Action	Object	Qualifier
clear	security-violations	
list	measurements security-violations	print or schedule
monitor	security-violations	print

Security Violations Measurements Report

The following list shows the data collected on the Security Violations Measurements Report form:

- *Login ID*

  The login that was attempted.

- *Port Type*

  The port that was used for the login attempt.

- *Successful Logins*

  The total number of login attempts for each login that completed.

- *Invalid Passwords*

  The number of passwords that were entered for the login that were incorrect.

- *Counted Since*

  The time at which the counts were last cleared and started accumulating again or when the system was initialized.

Security Violations Status Report

The Security Violations Status report appears as follows:

```
monitor security-violations

                        SECURITY VIOLATIONS STATUS

                                          Date: 9:21 am WED DEC 20, 1995

     SYSTEM MANAGEMENT VIOLATIONS              REMOTE ACCESS VIOLATIONS

 Date  Time  Login  Port        Ext       Date  Time    TG No. Mbr   Ext
 05/04 15:22 testid SYS-PORT    71111     05/03 21:14     3     50   81111
 05/04 15:23 testit SYS-PORT    73456     05/04 22:10    21     43   83333
 05/04 15:25 testx  SYS-PORT    72198     05/04 22:13    21     43   83333
```

Figure 3-5. Security Violations Status Report

The following list shows the data collected on the Security Violations Status Report:

- *Date*

  The date of the invalid login attempt.

- *Time*

  The time of the invalid login attempt.

- *Login*

  The login ID used in the invalid login attempt. The failure may have been due to an invalid password but the login (valid or invalid) is also listed.

- *Port*

  The port on which the invalid login attempt was made. **SYSAM-LCL** is for the MCU-MT; **SYSAM-RMT** is for the **MCU-ST**; and **SYS-PORT** is for the dial-up remote maintenance port.

- *Ext*

  The extension assigned to the port.

- *Date*

  The date of the remote access attempt.

- *TIme*

  The time of the remote access attempt.

- *TG No.*

  The trunk group number associated with the trunk used in the invalid remote access attempt.

- *Mbr*

  The trunk group member number of the trunk used in the invalid remote access attempt.

- *Ext.*

  The remote access extension.

System Status Reports

There are two monitor system status reports. Monitor System View1 Report includes maintenance status and last hour's traffic data for hunt groups and trunk groups. The Monitor System View2 Report provides maintenance status and last hour's traffic for trunk groups only (no hunt group data). For both reports, data for maintenance status is updated every minute. Data for traffic status is collected on an hourly basis from existing measurements.

NOTE:
Consider requesting these reports as the last request during your current login. The screens are exited by pressing CANCEL, which also logs you off the system.

Table 3-11 lists the commands related to obtaining system status reports.

Table 3-10. System Status Reports Administration Commands

Action	Object	Qualifier
monitor	system view1	
monitor	system view2	

Monitor System View1 Report

The Monitor System View1 report appears as follows:

```
monitor system view1

                                                MAINTENANCE STATUS

                                              # of alarms for trunks: 0
                                              # of alarms for stations: 0
                                              # of alarms for other res: 2
                                              INADS has been informed? n


                                TRAFFIC STATUS
                              Measurement Hour: 8
        Trunk Group Measurement              Hunt groups Measurement
 (4 grps with highest %time ATB)     (4 grps with highest # of qued calls)
    Grp no:                                    Grp no:
   Grp dir:                                 Calls qued:
Calls qued:                                 Calls aban:
 %Out blkg:
 %Time ATB:
                                                      9:26 WED DEC 20 1995
               - press CANCEL to quit -
```

Figure 3-6. Monitor System View1 Report

The following list shows the data collected for the Monitor System View1 Report:

- *Maintenance Status*

  Shows the number of alarms (including minor and major alarms) that may indicate problems on trunks, stations (maintenance alarm terminals) or other resources. If any alarm exists in the system or if remote maintenance (for example, INADS) has acknowledged an alarm, indications are shown on the report. Use the **display alarms** command to determine exactly what alarms currently exist. A **y** in the *INADS* field indicates the alarm has been acknowledged.

- *Traffic Status*

  Displays the call handling status for trunk groups and hunt groups. The report indicates the number of abandoned calls during the previously completed measurement interval for the identified group. For the trunk group measurements, only the four trunk groups numbers with the highest percentage of blocking are listed.

- *Grp Dir*

  Displays the trunk group direction (**incoming**).

- *%Time ATB*

  Shows the percentage of all trunks busy. For incoming trunk groups that report a high number in this field, enter the **list performance trunk-group** command to check the trunk group for problems.

Monitor System View2 Report

The following list shows the data collected for the Monitor System View2 Report:

- *Maintenance Status*

  Shows the number of alarms (including minor and major alarms) that may indicate problems on trunks, stations (maintenance alarm terminals) or other resources. If any alarm exists in the system or if remote maintenance (for example, INADS) has acknowledged an alarm, indications are shown on the report. Use the **display alarms** command to determine exactly what alarms currently exist. A **y** in the *INADS* field indicates the alarm has been acknowledged.

- *Traffic Status*

  Displays the call handling status for trunk groups only. The report indicates the number of abandoned calls during the previously completed measurement interval for the identified group. Only the four trunk groups numbers with the highest percentage of blocking are listed.

- *Grp Dir*

  Displays the trunk group direction (**incoming**).

- *%Time ATB*

  Shows the percentage of all trunks busy. For incoming trunk groups that report a high number in this field, enter the **list performance trunk-group** command to check the trunk group for problems (see “Performance Reports” for details).

Performance Reports

There are two performance reports. The Performance Summary Report summarizes the peak-hour trunk blocking, ARS traffic data, trunks out-of-service, and trunks not used. The report captures data from the previous day or the current day. The Trunk Group Performance Report gives a graphical and numerical display of the peak-hour blocking for each trunk group. This report captures data from the previous day or the current day.

Table 3-12 gives the commands related to obtaining performance reports.

Table 3-11. Performance Reports Administration Commands

Action	Object	Qualifier
list	performance summary	yesterday, today, print, or schedule
list	performance trunk-group	yesterday, today, print, or schedule

Performance Summary Report

The Performance Summary Report appears as follows:

```
list performance summary yesterday                              Page 1
Switch Name:  Customer_Switch_Name
SUMMARY PERFORMANCE REPORT                  Today: 4:38 pm  SAT DEC 16, 1995
PEAK HOUR TRUNK BLOCKING                         DAILY ARS CALLS CARRIED
Grp - %Out Blocking or % ATB - Grp      Grp -  % Calls Per Group Type -   %
No. 1 2 3 4 5 6 7 8 9 10 20 50 Blk
54  ////////////////////////   42       Type 1 10 20 30 40 50 60 80 100 Calls
59  ////////////////////////   39
58  ////////////////////////   36
63  ////////////////////////   34
61  ////////////////////       10
      TRUNKS OUT OF SERVICE                TRUNKS NOT SERVICE
Grp Trunks Out Of Service All Day       Grp Trunks With No Calls All Day
No. ----------------------------        No. ----------------------------
41    9 19                              55    1
73    7                                 60    9
211   1 2 3 4 5 6 7 8 9 10              223   19 20 21 22 23
      more out of service                     more out of service
```

Figure 3-7. Performance Summary Report

The following list shows the data collected for the Performance Summary Report:

- *Peak-Hour Trunk Blocking*

  Lists up to the maximum of five trunk groups that have the highest percentage of blocking in a measurement hour. For incoming trunk groups, the percentage of blocking is shown in the *% ATB* field. The number in this field represents the percentage of time all trunks in the trunk group were simultaneously in use during the measurement interval.

- *Daily ARS Calls Carried*

  This field is not used.

- *Trunks Out of Service*

  Lists four trunk groups with out-of-service trunks over the report interval. A list of the first 10 trunks that are out-of-service is also given. The message `more out of service` is displayed when more than four trunk groups are out of service or more than ten members of any trunk group are out of service. This measurement is a summary of the Outage Trunk Measurement Report.

- *Trunks Not Used*

  Lists the four trunk groups with trunks that have not been used over the report interval. A list of the first five trunks, in each of the identified groups, that have not been used are also listed. The message `more out of service` displays when more than four trunk group have not been used or whenever more than five members of any group have not been used. You should determine the exact reason why trunks are not being used. If there is only one trunk member identified for a trunk group and that member has zero calls, the trunk may be defective.

Trunk Group Performance Report

The Trunk Group Performance Report appears as follows:

```
list performance trunk-group yesterday
Switch Name:  Cust_Switch_Name
Trunk Group Performance                          Today: 4:28 pm  SAT DEC 16, 1995
            HIGHEST HOURLY TRUNK GROUP BLOCKING
Grp Grp      Grp Grp   --% Outgoing Blocking or % ATB-- %Out  %Time Meas  Total
No. Type     Dir Size  1 2 3 4 5 6 7 8 9 10 20 30 40 50 Blkg  ATB   Hour  Calls
1   isdn-pri in  6     /////////////////                      9     1200  876
2   isdn-pri in  5     ////////////////////////////            30    1400  94
3   tie      in  14    /////////////                          36    1300  312
5   tie      in  10    ////////////////////////               99    1300  542
12  tie      in  18    ////////////////////                   96    1400  614
23  tie      in  7     ////////////////////                   81    1400  359
41  tie      in  8     /////////////////////////              91    1300  411
221 tie      in  5     ////////////////////                   77    1300  109
```

Figure 3-8. Trunk Group Performance Report

The following list shows the data collected for the Trunk Group Performance Report:

- *Group No.*

  A number that identifies the trunk group associated with the displayed data.

- *Grp Type*

  The type of trunk (**isdn-pri** or **tie**) associated with the accumulated data.

- *Grp Dir*

  The direction (**incoming**) of the trunk group.

- *Grp Size*

  The number of trunks in the trunk group.

- *% Outgoing Blocking or % ATB*

  A graphical representation equivalent to the numerical value of all trunks busy. For incoming trunks, peak hour is the largest *% ATB*. Since outgoing blocking is meaningless for incoming trunks, it is displayed as ***** in that column.

- *%Out Blkg*

  This field is not used.

- *%Time ATB*

  The percentage of time that all trunks in the trunk group were simultaneously in use during the measurement interval. In use means the trunks were busy processing calls or are busied-out by maintenance.

- *Meas Hour*

  The starting time (using the 24-hour clock) of the hour during which the data was recorded.

- *Total Calls*

  Because the value collected is not relevant for incoming trunks, it is displayed as ***** in that column.

Occupancy Reports

These four reports track the percentage of time the MCU is busy performing call processing tasks, maintenance tasks, administration tasks, and the operating system support for each of these tasks. The percentage of time when the MCU is not used is referred to as Idle Occupancy (*Idle Occ*). These reports determine usage, available capacity and help to diagnose certain customer-reported problems.

Table 3-13 lists the commands that relate to occupancy measurements.

Table 3-12. Occupancy Reports Administration Commands

Action	Object	Qualifier
clear	measurements occupancy	
list	measurements occupancy	summary, print, or schedule
list	measurements occupancy	last-hour, print, or schedule
list	measurements occupancy	busiest-intervals, print, or schedule
list	measurements communications-links	print or schedule

Occupancy Summary Report

The Occupancy Summary Report appears as follows:

```
list measurements occupancy summary                          Page 1

                                                Date: 3:12 pm MON DEC 6, 1999
                          OCCUPANCY SUMMARY MEASUREMENTS
                           Peak Hour For Occupancy: 1000
Meas    Stat   CP    Sm    Idle   Total   Total            Inc     Out
Hour    Occ    Occ   Occ   Occ    Calls   Atmpt            Atmpt   Atmpt
1400    4      13    22     61    28463   43908              0       0
1300    5      14    23     58    35626   51094              0       0
1200    5      14    23     58    35632   51120              0       0
1100    5      14    23     58    35645   51130              0       0
1000    5      16    24     55    24342   31756              0       0

             press CANCEL to quit  --  Press NEXT PAGE to continue
```

Figure 3-9. Occupancy Summary Report

The following list shows the data collected for the Occupancy Summary Report:

- *Meas Hour*

  The starting time (using the 24-hour clock) of the hour during which the data was recorded. Data is listed beginning with the most recent.

- *Meas Min*

  The start 3-minute interval for which the measurement is taken. It takes the form **hh:mm**.

- *Date of Occur*

  The date and time that coincides with a line of busiest-interval data. It takes the form of **MM/dd/hh:mm** where **MM** is the month, **dd** is the day, **hh** is the hour, and **mm** is the minute.

- *Stat Occ*

  The amount of time taken by high-priority background processes to support call processing, maintenance, and administration functions.

- *CF Occ*

  The amount of time taken by call processing level processes, such as ISDN and CDR. It is not desirable for the MCU to operate at 100 percent processor occupancy. Rather, the *CP Occ* and *Stat Occ* fields should combine to no more than 70 percent. This allows the system to perform conferencing and administration functions at optimum levels.

- *Sm Occ*

  The amount of time taken by lower-priority activities, such as maintenance and administration command processing and error logging.

- *Idle OCC*

  The amount of time that the processor is unused. This value should not fall below 15 percent for a sustained period.

- *Total Calls*

  This field is not used and is marked with an *****.

- *Total Atmpt*

  The number of incoming trunk seizure attempts made during the measurement interval.

- *Out Atmpt*

  This field is not used.

Occupancy Last-Hour Measurements Report

The data collected for the Occupancy Last-Hour Measurements Report is the same as the Occupancy Summary Report except that the measurements are taken for the last hour. This report helps to identify potential load related problems that may have occurred during the last hour. Be sure to check the hardware error log for an excessive amount of maintenance activity when investigating a problem.

The Occupancy Last-Hour Measurements Report appears as follows:

```
list measurements occupancy last-hour                          Page 1

                                                Date: 3:13 pm MON DEC 6, 1999
                        OCCUPANCY LAST-HOUR MEASUREMENTS
Meas    Stat   CP    Sm    Idle   Total   Total            Inc     Out
Min     Occ    Occ   Occ   Occ    Calls   Atmpt            Atmpt   Atmpt
15:11    2      5    26    67      646    1421               0       0
15:08    4      5    22    69      641    1412               0       0
15:05    5      6    25    64      639    1410               0       0
15:02    4      6    24    66      645    1420               0       0
14:59    3      4    22    71      639    1411               0       0
14:56    3      5    24    68      639    1412               0       0
14:53    6      6    24    64      645    1418               0       0
14:50    4      5    27    64      641    1418               0       0
14:47    4      6    19    71      648    1429               0       0
14:44    2      5    26    67      639    1405               0       0
14:41    4     14    19    63     1624    2399               0       0
14:38    6     15    28    51     1786    2556               0       0
            press CANCEL to quit  --  Press NEXT PAGE to continue
```

Figure 3-10. Occupancy Last-Hour Measurements Report

Occupancy Busiest-Interval Measurements Report

The data collected for the Occupancy Busiest-Interval Measurements Report is the same as the Occupancy Summary Report except that the measurements are a collection of the 20 busiest 3-minute intervals (instead of one-hour intervals) within the past two months. This report is useful to maintenance personnel who need to investigate habitual performance problems or problems that were not reported when they occurred. Be sure to check the hardware error log for an excessive amount of maintenance activity when investigating a problem.

An Occupancy Busiest-Interval Measurements Report for 3-minute interval measurements appears as follows:

```
list measurements occupancy busiest-intervals                    Page 1
                                                  Date: 3:13 pm MON DEC 6, 1999
              OCCUPANCY BUSIEST 3-MINUTE INTERVALS MEASUREMENTS

Date of      Stat  CP   Sm   Idle  Total  Total          Inc   Out
Occur        Occ   Occ  Occ  Occ   Calls  Atmpt         Atmpt Atmpt
11/11/10:20  16     9   26   49     686   1225            0     0
11/11/11:14   8    16   27   49    1788   2558            0     0
11/11/12:38   7    15   21   57    1786   2554            0     0
11/11/13:41   6    16   26   52    1786   2553            0     0
11/11/14:11   7    15   25   53    1780   2557            0     0
            press CANCEL to quit  --  Press NEXT PAGE to continue
```

Figure 3-11. Occupancy Busiest-Interval Measurements Report

Occupancy Communications Link Measurements Report

The Occupancy Communications Link Measurements Report appears as follows:

```
list measurements communications-links                          Page 1

                                                Date: 1:55 pm TUE DEC 12, 1995

                          COMMUNICATION LINK MEASUREMENTS

 Meas      Link      Link      Link      Link      Link      Link      Link      Link
 Hour      9         10        11        12        13        14        15        16
 1200      10471     576       24        4         0         40        2         0
 1100      13764     612       24        14        0         313       4         0
 1000      12217     550       24        4         0         36        9         0
 900       12365     601       26        4         0         32        2         0
 800       12630     559       28        4         0         36        4         0
 700       12714     412       24        4         0         36        4         0
 600       12531     299       24        4         0         40        4         0
 500       12407     352       24        4         0         42        2         0
 400       12173     311       34        4         0         32        2         0
 300       12121     301       24        4         0         36        4         0
 200       12561     412       24        4         0         36        4         0
 100       12501     478       24        4         0         36        2         0

           press CANCEL to quit  --  Press NEXT PAGE to continue
```

Figure 3-12. Occupancy Communications Link Measurements Report

The following list shows the data collected for the Occupancy Communications Link Measurements Report:

- *Meas Hour*

  The starting time (using the 24-hour clock) of the hour during which the data was recorded. Data is listed beginning with the most recently completed hour.

- *Link #*

  The links are identified by numbers **9** through **16**. The numbers in each column represent the number of messages traversing the link. Once a link is established and traffic begins flowing over it, the messages are counted automatically; no command is required.

Tone Receiver Measurements Report

The Tone Receiver Measurements Report provides traffic data on the TN744C, TN420C, or TN748D Tone Detector circuit pack. Reports can be requested for yesterday's peak hour, today's peak hour, or the last hour. You may want to request this report as conferees report they are unable to join a conference when other conferences are also active.

Table 3-14 shows the command related to obtaining the tone receiver measurements.

Table 3-13. Tone Receiver Measurements Administration Command

Action	Object	Qualifier
list	measurements tone-receiver	last-hour, today-peak, yesterday-peak, print, or schedule

The Tone Receiver Measurements Report appears as follows:

```
list measurements tone-receiver last-hour

Switch Name:  Cust_Switch_Name                    Date: 4:16 pm TUE DEC 19, 1995

                          TONE RECEIVER MEASUREMENTS
                 Total   Total   Peak   Total    Peak     Total    Peak
      Hour   Type Avail    Req    Req    Queued   Queued   Denied   Denied
      1400   DTMF   40      23     13      0        0        0        0
      1500   GPTD    6      94      7                        1        1

                       PN      PN        Peak      Total      Peak
       PN      Type    Req     Alloc     Alloc     Off-PN     Off-PN
       1       DTMF     8       8         200         0          5
       1       GPTD    12      12         3          0          0

             Press CANCEL to quit   -   press NEXT PAGE to continue
```

Figure 3-13. Tone Receiver Measurements Report

The following list shows the data collected for the Tone Receiver Measurements Report:

- *Hour*

  The starting time (using 24-hour clock) of the hour during which the greatest number of requests for tone receivers were made.

- *Type*

  The type of tone receiver being measured. Each TN748 circuit pack provides four DTMF ports (for touchtone reception) and two GPTD ports (for call progress tone reception).

- *Total Avail*

  The systemwide total number of DTMF and GPTD receivers that are available and not busied-out for maintenance.

- *Total Req.*

  The systemwide total number of requests by the tone receivers during the listed hour. The total number of requests is calculated by incrementing a counter for each request.

- *Peak Req*

  The systemwide peak number of simultaneous requests for tone receivers that occurred at any one time for the listed hour. The peak (or maximum) number is calculated by incrementing a counter for each request and decreasing the counter when the request fails or a tone receiver is released. Denied requests fail and are given the reorder tone.

- *Total Queued*

  The systemwide total number of tone receivers that were queued during the listed hour. There is no queuing for GPTD receivers.

- *Peak Queued*

  The systemwide maximum number of DTMF receivers that were queued at any one time during the listed hour. The system has a maximum queue size of four. GPTD receivers cannot be queued.

- *Total Denied*

  The systemwide total number of requests for tone receivers that were denied because none were available.

- *Peak Denied*

  The systemwide peak number of requests for tone receivers that were denied because no receivers were available during the listed hour.

- *PN*

  The port network where the circuit pack is located.

- *Type*

  The type of tone receiver being measured (**DTMF** or **GPTD**).

- *PN Req*

  The port number requests.

- *PN Alloc*

  The total number of tone receivers located in the port network that were allocated for use during the listed hour.

- *Peak Alloc*

  The peak number of tone receivers located in listed port network that were used simultaneously during the listed hour.

- *Total Off-PN*

  This field is not used.

- *Peak Off-PN*

  This field is not used.

Conference Control Features

4

Overview

This chapter discusses three key MCU conference control features, including the following:

- Universal Conference Control *Star*
- Conference control commands from the MCU-MT
- OpCenter

Using Universal Conference Control *Star*

Universal Conference Control (UCC) *Star* allows the user to control conferences in a manner similar to Chair Control conference mode. UCC *Star* enables the use of a regular touchtone telephone as the controlling terminal. This section discusses setting and using UCC *Star*.

Setting Up the Universal Conference Control *Star* Controller

The following list indicates the requirements for a terminal to be used as a Universal Conference Control (UCC) *Star* controller at a customer location.

- Terminal must be able to access the MCU via PRI trunks or non-ISDN DS1 trunks.
- Terminal must be a terminal capable of transmitting Dual-Tone Multifrequency (DTMF) touchtones to the MCU.
- If the terminal has a speakerphone, it is recommended that you “mute” audio feeding into the terminal. This is done to prevent audio frequencies that may be interpreted as touchtones from being transmitted to the MCU. If no mute capability is available and you are using a speakerphone, it is recommended that you turn off the speakerphone and place the handset next to the phone. Since the handset is less sensitive than the speakerphone, this reduces the chance of stray tones reaching the MCU.

Connecting the UCC *Star* Controller

Once the conference is scheduled with the reservations agent, the convener is provided with a confirmation report from CRCS (if available). This report includes instructions on how to access UCC *Star* features via touchtone dialing. The convener is responsible for ensuring that the controlling location receives the UCC *Star* dialing instructions along with its dial-in number. Included with the instructions are the endpoint numbers of each location reserved in the conference. These numbers are necessary to use the endpoint-specific options in UCC *Star*, such as selecting an endpoint for the broadcaster or dropping an endpoint. If CRCS is not available for the conference, the convener must manually write down the feature access codes and endpoint numbers and then provide this information to the controlling location.

The UCC *Star* party is identified on the conference form, and it is given a dial-in number that has been reserved for the UCC *Star* controller. This number is the dial-in number that the controlling endpoint subscriber will dial to reach the MCU. Once the controlling party successfully completes optional passwording procedures and a TN744 touch-tone detector is connected to the port, a feedback tone is provided to the party to indicate that the MCU is ready to accept UCC *Star* commands.

Failure to enter a correct password or to connect a tone detector causes the MCU to drop the party. Accordingly, the Status Conference form indicates the drop reason as either a password failure or a failure to connect a tone detector resource.

The joined controlling party does not receive media (that is, audio, video, or data), and it does not transmit media to the multimedia conference. Its audio is not summed into the audio portion of the conference, and it does not receive conferenced audio. Because the MCU never considers the UCC *Star* controller a valid video source, it does not attempt to make the controller a video broadcaster in the VAS switching algorithm. Accordingly, an entry tone is not provided to endpoints already connected to the conference when the controller is joined.

UCC *Star* Command Interface

The controller accesses UCC services by entering the telephone keyboard commands as shown in the next table. These feature access codes and endpoint numbers of each location are provided to the controlling location prior to the conference.

Accessing and Implementing Universal Conference Control Services

The controller (UCC terminal) can be used to access UCC services by entering the appropriate feature access code for the service. The following table identifies the available UCC services, indicates the means to access each service, and describes each service:

Table 4-1. UCC Services Access

Desired Service	Required Keypad Entry	Description
Unlock Video	****8**	Releases any fixed broadcaster and returns the conference to Voice Activated Switching (VAS) mode.
Select a site to be a video broadcaster	*** *N***	Locks in the specified site ***N*** as the broadcaster (broadcasting site). The broadcaster sees the site that is currently speaking, or, if no other sites are speaking, the site that spoke most recently.
Browse sites (after selecting a broadcaster)	****2**	Shows the video from one specific site to the broadcasting site. All other sites, including the site whose video is being shown to the broadcaster, continue to view the broadcaster. Pressing "#" allows the broadcaster to view each of the other sites in the prescribed order.
Conduct Rollcall of all sites	****7**	Shows the video from a specific site to all the sites. Pressing "#" shows the video from each of the other sites, in the prescribed order, to all the sites in the conference.
Scan to the Next Site (when in Rollcall or Browse option)	**#**	Pressing "#" displays the next site in Rollcall and Browse modes. The sites are scanned in order of their site numbers on the Conference Record form.
Change image from full-screen to quad-screen	****4#**	Partitions the full-screen image into four areas (quadrants). This allows up to four different sites to be viewed at one time.
Assign site to quadrant 1 (top left), 2 (top right), 3 (bottom left), or 4 (bottom right)	****4*QN***	Enables the user to choose what is seen in each quadrant (***Q***). The video from a single site can be locked into a quadrant. Also, a quadrant can display video according to VAS. To set a quadrant according to VAS, use ***0*** as the site number (***N***).
Change image from quad-screen to full-screen	****1**	Returns the quad-screen image back to full-screen. The layout of the quad-screen is remembered.

Table 4-1. UCC Services Access — *Continued*

Desired Service	Required Keypad Entry	Description
Transition to upper-panoramic three-image layout	****46**	Displays a three-image CPP screen layout where the top half of the screen displays one image while the bottom half of the screen is segmented into two screen displays.
Transition to lower-panoramic three-image layout	****47**	Displays a three-image CPP screen layout where the bottom half of the screen displays one image while the top half of the screen is segmented into two screen displays.
Transition to dual-panoramic screen layout	****48**	Displays a two-image CCP screen layout where the top half of the screen displays one image and the bottom half of the screen displays one image.
Transition to Executive Conferencing Mode	****49**	Displays a dual-panoramic screen layout for the three broadcaster endpoints and an upper-panoramic screen layout for the other conference participants.
Mute site	****61 *N***	Mutes the incoming audio of site ***N***. Other sites do not hear anyone at the muted site, but the muted site continues to hear audio from other unmuted sites. **NOTE**: This command overrides the “mute” button of the video equipment at Site *N*.
Unmute site	****62 *N***	Unmutes the incoming audio of site ***N.*** This site can be heard by all the other sites. **NOTE**: This command does not override the “mute” button of the video equipment at Site *N*.
Solo Audio	****63 *N***	Mutes the audio of all the sites except site ***N.*** Therefore, only the “solo audio” site can be heard by all the other sites.
Unmute all sites	****6#**	Unmutes all the sites. All the sites can speak and be heard.
Adjust Audio Volume for Audio Add-On Party	**64**	(This is the command code used within the commands for the specific services provided.)
— Decrease	****64 *N* 7**	Decreases the volume of endpoint ***N*** by 3 db.
— Reset	****64 *N* 8**	Resets the volume of endpoint ***N*** to the default (6 db loss).
— Increase	****64 *N* 9**	Increases the volume of endpoint ***N*** by 3 db.

Table 4-1. UCC Services Access — *Continued*

Desired Service	Required Keypad Entry	Description
Manipulate Most Recently Joined Endpoint	**99**	(This is the command code used within the commands for the specific services provided.)
— Broadcast	***99**	Broadcasts the most recently joined endpoint.
— Mute	****6199**	Mutes the most recently joined endpoint.
— Unmute	****6299**	Unmutes the most recently joined endpoint.
— Solo Audio	****6399**	Performs Solo Audio on the most recently joined endpoint.
— Set in Quad	****4Q99**	Sets the most recently joined endpoint into a quadrant.
— Drop	****399**	Drops the most recently joined endpoint.
Drop site	****3*N***	Drops a single site ***N*** from the conference.
End conference	****91**	Drops all sites in the conference, including the UCC port, and terminates the entire conference.

NOTE:
Refer to Appendix B for a complete description of the UCC services.

To select, for example, endpoint 2 to be a video broadcaster, enter either "*** 2**" or "*** 02**." To drop endpoint 12, enter "****3 12**."

NOTE:
Whenever you are using the keypad to access UCC services, be sure to strike the appropriate keys in a deliberate manner (that is, firmly and not too quickly). Otherwise, you may not get the desired result.

The "#" key or double-digit dialing (for example, ***02**, ***03**, etc.) can be used to override the interdigit timer. This causes the MCU to process the command immediately; as a result, you do not have to wait for the timer to expire.

Feedback Tones

For each command you enter, a feedback tone is provided to the UCC controller. The tone indicates to the controller that the MCU has collected and (if possible) acted upon a command. It also indicates that the MCU is ready to accept another command.

The UCC controller hears one of two tones. A “positive” feedback tone is played whenever either of the following is true:

- Controller successfully joins the conference.
- Request (command entered) is valid and the MCU is able to honor the request.

The “positive” feedback tone consists of one short tone.

A “negative” feedback tone is played whenever either of the following is true:

- Request is invalid.
- Request is well-formed, but the MCU is unable to honor it.

The “negative” feedback tone consists of five short tones in rapid succession.

Command Collection and Interpretation

The MCU responds to the digits entered by the UCC controller according to a number of different circumstances, as described in the following sections.

Valid Requests

The MCU recognizes any string of digits illustrated in the previous table as a complete request. For the command to be valid, it must satisfy the conditions for the particular command as described in the “Feature Description” chapter. The following list identifies, interprets, and comments on some valid requests:

- ****8**

  Unlock video (valid only if the current broadcaster is not broadcasting due to an MCV request).

- ****91**

  Drop all endpoints (always valid and honored).

  NOTE:
  This command cancels a previous command to select the broadcaster and also cancels Rollcall and Browse.

- ***8**

  Broadcast endpoint 8 (valid only if endpoint 8 is joined to the conference and is a valid video source).

  NOTE:
  This command overrides a previous command to select the broadcaster, and it cancels Rollcall and Browse.

- ****34**

  Drop endpoint 4 (valid only if endpoint 4 [which could be the controller] is joined to the conference).

- ****2**

  Begin Browse (valid only if the broadcaster is fixed and if the conference is currently viewing full screen).

For each of the previous cases, the MCU provides positive feedback tone and honors the request.

Invalid Requests

If the MCU detects a set of digits that does not adhere to the access code rules in the previous table, the MCU plays a negative feedback tone, disregards the request, and waits for another command. If the request is recognized as completed according to the previous table, but the MCU cannot honor the request because it does not satisfy the conditions required for the request, negative feedback tone is provided. The following list identifies, interprets, and comments on some invalid requests:

- ****5**

  (Invalid digit following “**.”

- **1 or 3**

  (Invalid first digits. All commands must be preceded by either “*” or “**”.)

- ****8**

  Unlock video (invalid if the current broadcaster is broadcasting due to an MCV request).

- ***2**

  Broadcast endpoint 2 (invalid if endpoint 2 is not reserved in the conference, has not been joined to the conference, or is not video source capable).

- ***26**

  Broadcast endpoint 26 (invalid because the endpoint number is not in the range 1 through 25).

- ****311**

  Drop endpoint 11 (invalid if endpoint 11 is not reserved in the conference or is not joined to the conference).

Interdigit Timer Expiration

If, while making a request, you wait too long after entering a digit, the MCU processes the digits already entered, attempts to honor the request, plays a feedback tone, and waits for another command.

The interdigit timing varies according to your current dialing stage. Specifically, a 5-second timer is set immediately after the MCU detects an initial "*". Also, a 5-second timer is set once the MCU recognizes that you intend to make a "drop endpoint" or "mute endpoint" request. (This timer is set before you enter any digits for the endpoint number.) Finally, a 2-second timer is set after you enter the first digit of an endpoint number for the "select broadcaster" or "drop endpoint" request.

Timing is disabled immediately after you hear a feedback tone. This gives you an indefinite amount of time before you enter the next request.

The following list provides some interdigit timer scenarios and consequences.

- ****** <2-seconds>

  A timeout occurs and the MCU processes the request because the user waited too long to enter a digit after the user entering "**". The MCU plays a negative feedback tone and disregards the request.

- ****3** <5-seconds>

  The user failed to enter, within five seconds, an endpoint number after "3." The user may have intended to drop an endpoint and then decided against doing so. Alternatively, the user may have had the same intention but could not locate in time the endpoint number of the location to be dropped. Since this is an incomplete command, the MCU plays a negative feedback tone and disregards the request.

- ***1** <2-seconds>

  Broadcast endpoint 1. The MCU cannot immediately process the digits prior to the 2-second timeout due to an ambiguity in the request. For example, if the MCU receives "*1" to broadcast an endpoint, it is not immediately known whether the user intends to broadcast endpoint "1" or to broadcast an endpoint with a number between "10" and "19" inclusive. Therefore, the MCU gives the user two seconds after "1" is dialed to enter a second digit for the endpoint number. This example shows that the user probably intended to enter endpoint number "1." The MCU provides a positive feedback tone if it is able to broadcast the location with endpoint number "1".

- ****32** <2-seconds>

  Drop endpoint 2. This timeout causes the MCU to drop the location with endpoint 2. The MCU must wait for the 2-second timer to expire before it can act on this request because it is not immediately known whether the user intends to drop endpoint “2” or an endpoint with a number between “20” and “25” inclusive. This example shows that the user probably intended to enter “2” for the endpoint number. The MCU provides a positive feedback tone if it is able to drop the location with this endpoint number.

Overriding the Interdigit Timer

The “#” key or double-digit dialing (for example, ***02**, ***03**, etc.) can be used to override the interdigit timer. This causes the MCU to process the command immediately and, as a result, you do not have to wait for the timer to expire.

The “#” key allows you to resolve ambiguity in the endpoint number. For example, “*1#” causes the MCU to broadcast endpoint 1 immediately without waiting for the timer to expire.

Entering equivalent 2-digit endpoints numbers also resolves ambiguous cases. For example, instead of entering “*2” to broadcast endpoint 2, you could enter “*02” to broadcast endpoint 2 immediately (and without waiting for the timer to expire).

Handling Dialing Mistakes

Once the MCU intercepts a command that it recognizes as “valid,” the MCU immediately honors the command (if possible) and provides feedback tone. However, if you enter a partial or incomplete request (for example, “*”, “**”, or “**3”), the expiration of the interdigit timer essentially causes the request to be “canceled.” For example, if you intend to enter a “drop endpoint” command but fail to enter the endpoint number after “**3” within five seconds, the MCU disregards the request and provides a negative feedback tone.

Dropping the UCC Controller

Whenever the UCC party is disconnected from a conference, it is assumed that the conference's primary method of control is no longer in effect. Therefore, to avoid leaving the conference “stuck” in a mode with a fixed full-screen broadcaster (via a Browse, Rollcall, or Select Broadcaster request, for example), any video locks caused by these commands are automatically canceled when the UCC party is disconnected. All active Mute features (such as Solo Audio and Mute Endpoint) are also canceled.

Disconnecting the UCC party has no affect on an MCV broadcaster. If an endpoint is broadcast due to an MCV request, that endpoint remains broadcasting until it sends a cancel-MCV request.

The remaining parties do not hear an exit tone after the controller is dropped from the conference.

If Continuous Presence *Plus* is enabled, quadrant video locks are not cancelled after the UCC drops.

NOTE:
Refer to the “Universal Conference Control” section in Appendix B for complete details on the interactions between UCC and Continuous Presence *Plus*.

Using UCC Options

The following sections discuss selected UCC options and present several command line examples for each option. For a complete description of these and all the other UCC options, refer to Appendix B.

Browse

Browse is an option of UCC that allows the fixed broadcaster (that is, the presenter or “selected broadcaster” via UCC) to scan endpoints in the conference without specifying an endpoint number for each scanned endpoint. Refer to Appendix B for a complete description of Browse.

Command Set. The following commands pertain to Browse.

- ****2**

  Starts Browse. This command is valid in presentation mode with a fixed broadcaster in effect (that is, if ***N** [explained later in this list] was previously entered and executed successfully) and also if the conference is currently viewing full-screen video. If the command is successful, the MCU immediately locks return video on an endpoint and provides a positive tone after entering Browse mode.

- **#**

  Selects the next return video in Browse. The current broadcaster is skipped in the browse unless it is the only party in the conference. A positive tone is provided to indicate a switch to a new return video.

- ***N**

  Selects a new UCC broadcaster and cancels Browse.

- ****8**

  Clears all controlled video locks and, if viewing full-screen video, cancels Browse.

- ****91**

  Ends the conference and cancels Browse.

- ****2**

  Starts Browse.

- ****7**

  Starts Rollcall. (This is denied in presentation mode and allowed in VAS mode).

Using Rollcall

Like Browse, Rollcall is an option of UCC that allows the UCC broadcaster to scan endpoints in the conference without specifying an endpoint number for each scanned endpoint. Unlike Browse, Rollcall allows all the endpoints in the conference to see the change in video caused by the scan. This feature allows the broadcaster to be seen by all the endpoints in the conference including the broadcaster itself (the broadcaster receives looped video). Refer to Appendix B for a complete description of Rollcall.

Command Set. The following commands pertain to Rollcall.

- ****7**

  Starts Rollcall. Once this command is entered, the MCU immediately locks the broadcast video and provides the broadcaster with looped video. A positive tone is given if this is successful.

- **#**

  Selects the next endpoint in the roll call. A positive tone is provided to indicate a switch in broadcast video.

- ***N**

  Selects a new UCC broadcaster and automatically cancels Rollcall.

- ****8**

  Clears all controlled video locks and, if viewing full-screen video, cancels Rollcall.

- ****91**

  Ends the conference and cancels Rollcall.

- ****7**

  Starts Rollcall.

- ****2**

  Starts Browse. (This allowed in Presentation mode and denied in VAS mode.)

UCC Options Permissions Summary

The following table provides a summary of permissions for the UCC options.

Table 4-2. Summary of Permissions for UCC Options

UCC Option	VAS conferences?	Presentation Conferences?	Viewing Full Screen	Viewing Quad Screen?
Select Broadcaster	yes	no	yes	yes, but causes an automatic transition to full screen
Browse Endpoints	yes	yes	yes	no when in VAS mode, yes when in presentation mode
Rollcall of all Endpoints	yes	no	yes	yes, but causes an automatic transition to full screen
Unlock Video	yes	yes	yes	yes
Mute Controls	yes	yes	yes	yes
Force Image to Quad Screen*	yes	yes	yes	N/A, has no effect
Set Quadrant	yes	yes	yes, saved, but does not cause transition to quad screen	yes
Force Image to Full Screen	yes	yes	N/A, has no effect	yes
Drop Endpoint	yes	yes	yes	yes
End Conference	yes	yes	yes	yes
Adjust Audio Volume for Audio Add-On Party	yes	yes	yes	yes
Manipulate Most Recently Joined Endpoint	yes	yes	yes	yes
Executive Conferencing	yes	no	yes	N/A

* Allowed only in conferences with Continuous Presence *Plus*

Using Conference Control Commands

Conference control commands can be issued from the MCU-MT to control MCU conferences. A number of these commands are based on the Universal Conference Control (UCC) feature and are consistent with the operation of this feature. On the other hand, some of the commands are intended for troubleshooting purposes. Once a command is issued and successfully completes, the message “Command successfully completed” is displayed. If the command does not complete successfully, the appropriate error message is displayed.

Operations and Commands

This section indicates the possible operations available via the MCU-MT conference control commands, the command for each operation, and a description for each command.

Unlocking Video

Syntax:

control conference *CONFERENCE_ID* unlock video *SUB-PICTURE*

Description:

For full-screen video, this command switches back to the voice-activated broadcaster, and it terminates the Rollcall, Browse, and Select Broadcaster options of UCC. If the conference previously went from quad-screen or panoramic video to full-screen video due to Rollcall or Select Broadcaster, this command does not return the conference to quad-screen or panoramic video.

For quad-screen or panoramic video, this command switches back to the administered quad-screen or panoramic view. If ***SUB-PICTURE*** is not specified, all four quadrants or the three images are returned to the administration view. If ***SUB-PICTURE*** is specified, only the specified quadrant is returned to the administration view. Whenever ***SUB-PICTURE*** command is not seen until the conference switches to quad-screen or panoramic video.

For the command to be successful, the following must be true:

- ***CONFERENCE_ID*** must be all digits and in the range from ***1*** through ***200***.
- ***SUB-PICTURE*** must be all digits and in the range from ***1*** through ***4***.
- Specified conference must be in use, and it must be able to support Continuous Presence *Plus*.
- Command must be consistent with the conference mode.
- Current broadcaster must *not* be MCV.

Selecting a Full-Screen Broadcaster

Syntax:

control conference *CONFERENCE_ID* broadcaster *ENDPOINT_ID*

Description:

This command selects a full-screen broadcaster.

For a Continuous Presence *Plus* conference in quad-screen or panoramic video, the command switches the conference to full-screen video. The command locks an endpoint as a video broadcaster. (The “unlock-video” or “quad-screen” (transition to quad) command releases this lock.) This command is not allowed in Presentation mode conferences.

For full-screen conferences, the command auto-cancels Rollcall or Browse, and it overwrites the current MCV broadcaster. Once the video is unlocked, the endpoint with the MCV becomes the broadcaster once again.

For the command to be successful, the following must be true:

- ***CONFERENCE_ID*** must be all digits and in the range from ***1*** through ***200***.
- ***ENDPOINT_ID*** must be all digits and in the range from ***1*** through ***25***.
- Specified conference must be in use.
- Command must be consistent with the conference mode.
- Specified endpoint must be joined to the conference, and it must be a valid video source.

Selecting a Broadcaster or Viewed Endpoint for a Quadrant

Syntax:

control conference *CONFERENCE_ID* set-sub-pict *SUB-PICTURE* *ENDPOINT_ID*

Description:

This command selects (locks) the broadcaster for the specified quadrant (**1, 2, 3**, or **4**). If the command is issued when the conference is viewing full screen, command information is stored and becomes effective once the conference switches to quad-screen or panoramic video (via the "quad-screen" command). If ***ENDPOINT_ID*** is ***0,*** the specified quadrant is locked as a Voice-Activated Switching (VAS) quadrant even if the quadrant was previously administered as a fixed quadrant. The quadrant can be returned to the administered view via the "unlock-video" command.

For conferences in Presentation mode, the command selects the endpoint videos to be viewed by the presenter.

For the command to be successful, the following must be true:

- ***CONFERENCE_ID*** must be all digits and in the range from ***1*** through ***200***.
- ***ENDPOINT_ID*** must be all digits and in the range from ***0*** through ***25,*** where ***0*** is VAS.
- ***SUB-PICTURE*** must be all digits and in the range from ***1*** through ***4***.
- Specified conference must be in use, and it must be able to support Continuous Presence *Plus*.
- Command must be consistent with the conference mode.
- Specified endpoint must be joined to the conference, and it must be a valid video source, but it must not be the Presenter.

Swapping Quadrants

Syntax:

control conference *CONFERENCE_ID* swap-sub-pict *SUB-PICTURE* *SUB-PICTURE*

Description:

This command exchanges the characteristics of the two specified quadrants.

For the command to be successful, the following must be true:

- ***CONFERENCE_ID*** must be all digits and in the range from ***1*** through ***200***.
- ***SUB-PICTURE*** must be all digits and in the range from ***1*** through ***4***.
- Specified quadrants must be different.

- Specified conference must be in use, and it must be able to support Continuous Presence *Plus*.
- Command must be consistent with the conference mode.

Beginning Browse

Syntax:

control conference *CONFERENCE_ID* browse-begin

Description:

This command starts the Browse option of UCC. However, the full-screen broadcaster must be selected first. The next endpoint serving as the return video source is selected via the “next-endpoint” command. The “unlock-video” command returns the conference back to VAS mode.

The command switches a quad-screen or panoramic conference to full-screen video. The “unlock-video” command terminates Browse, but it does not automatically return the conference to quad-screen or panoramic video.

For the command to be successful, the following must be true:

- ***CONFERENCE_ID*** must be all digits and in the range from ***1*** through ***200***.
- ***ENDPOINT_ID*** must be all digits and in the range from ***1*** through ***25***.
- Specified conference must be in use, and it must not be in Rollcall mode.
- Command must be consistent with the conference mode.
- Fixed broadcaster must be established.
- Current broadcaster must not be MCV.

Beginning Rollcall

Syntax:

control conference *CONFERENCE_ID* rollcall-begin

Description:

This command starts the Rollcall option of UCC, and it releases the currently fixed broadcaster. The next endpoint serving as the video source is selected via the “next-endpoint” command. The “unlock-video” command returns the conference back to VAS mode.

The command switches a quad-screen or panoramic conference to full-screen video. The “unlock-video” command terminates Rollcall, but it does not automatically return the conference to quad-screen or panoramic video. Rollcall is not allowed in Presentation mode conferences.

For the command to be successful, the following must be true:

- ***CONFERENCE_ID*** must be all digits and in the range from ***1*** through ***200***.
- Specified conference must be in use.
- Command must be consistent with the conference mode.

Selecting the Next Endpoint

Syntax:

control conference *CONFERENCE_ID* next-endpoint

Description:

This command selects the next endpoint as a video source for the Browse and Rollcall features of UCC.

For the command to be successful, the following must be true:

- ***CONFERENCE_ID*** must be all digits and in the range from ***1*** through ***200***.
- Specified conference must be in use.
- Command must be consistent with the conference mode.
- Rollcall or Browse, as appropriate, must be activated.

Dropping an Endpoint

Syntax:

control conference *CONFERENCE_ID* drop-endpoint *ENDPOINT_ID*

Description:

This command disconnects the designated endpoint from the specified conference.

For the command to be successful, the following must be true:

- ***CONFERENCE_ID*** must be all digits and in the range from ***1*** through ***200***.
- ***ENDPOINT_ID*** must be all digits and in the range from ***1*** through ***25***.
- Specified conference must be in use.
- Command must be consistent with the conference mode.
- Specified endpoint must be joined to the conference.

Dropping All Endpoints

Syntax:

control conference *CONFERENCE_ID* drop-all

Description:

This command disconnects all endpoints from the specified conference.

For the command to be successful, the following must be true:

- ***CONFERENCE_ID*** must be all digits and in the range from ***1*** through ***200***.
- Specified conference must be in use.
- Command must be consistent with the conference mode.

Muting an Endpoint

Syntax:

control conference *CONFERENCE_ID* mute-endpoint *ENDPOINT_ID*

Description:

This command mutes the specified endpoint (that is, it excludes the endpoint's audio from the audio sum). For this process, the Status Conference form displays value **M** in the *Ts* field. Value **M** takes priority over value **m** (mute by an endpoint). This pertains to all connected endpoint types that have audio capabilities.

For the command to be successful, the following must be true:

- ***CONFERENCE_ID*** must be all digits and in the range from ***1*** through ***200***.
- ***ENDPOINT_ID*** must be all digits and in the range from ***1*** through ***25***.
- Specified conference must be in use.
- Command must be consistent with the conference mode.
- Specified endpoint must be joined to the conference, and it must have audio capabilities.

Unmuting an Endpoint

Syntax:

control conference *CONFERENCE_ID* unmute-endpoint *ENDPOINT_ID*

Description:

This command unmutes the designated endpoint (that is, it includes the endpoint's audio into the audio sum if the endpoint hasn't muted itself via AIM).

For the command to be successful, the following must be true:

- ***CONFERENCE_ID*** must be all digits and in the range from ***1*** through ***200***.
- ***ENDPOINT_ID*** must be all digits and in the range from ***1*** through ***25***.
- Specified conference must be in use.
- Command must be consistent with the conference mode.
- Specified endpoint must be joined to the conference.

Unmuting All Endpoints

Syntax:

control conference *CONFERENCE_ID* unmute-all

Description:

This command unmutes in the specified conference all the endpoints that were muted via conference control (and do not control AIM).

For the command to be successful, the following must be true:

- ***CONFERENCE_ID*** must be all digits and in the range from ***1*** through ***200***.
- Specified conference must be in use.
- Command must be consistent with the conference mode.

Enabling Solo Audio

Syntax:

control conference *CONFERENCE_ID* solo-audio *ENDPOINT_ID*

Description:

This command mutes all but the specified endpoint in the conference. Endpoints joining the conference subsequently join muted.

For the command to be successful, the following must be true:

- ***CONFERENCE_ID*** must be all digits and in the range from ***1*** through ***200***.
- ***ENDPOINT_ID*** must be all digits and in the range from ***1*** through ***25***.
- Specified conference must be in use.
- Command must be consistent with the conference mode.
- Specified endpoint must be joined to the conference, and it must have audio capabilities.

Suppressing Video

Syntax:

control conference *CONFERENCE_ID* suppress-video *ENDPOINT_ID*

Description:

This command stops the conference from using the specified endpoint's video as a video source. Feature interactions are similar to VIS (video suppressed at the request of the endpoint).

For the command to be successful, the following must be true:

- ***CONFERENCE_ID*** must be all digits and in the range from ***1*** through ***200***.
- ***ENDPOINT_ID*** must be all digits and in the range from ***1*** through ***25***.
- Specified conference must be in use.
- Command must be consistent with the conference mode.
- Specified endpoint must be joined to the conference, and it cannot be an audio-only or UCC party.

Unsuppressing Video for an Endpoint

Syntax:

control conference *CONFERENCE_ID* unsuppress-video *ENDPOINT_ID*

Description:

This command unsuppresses the video of an endpoint that was suppressed via the "suppress-video" command. In other words, the command allows the conference to start using the endpoint's video as a video source if the endpoint has not suppressed its own video via VIS and if the endpoint's video is still valid.

For the command to be successful, the following must be true:

- ***CONFERENCE_ID*** must be all digits and in the range from ***1*** through ***200***.
- ***ENDPOINT_ID*** must be all digits and in the range from ***1*** through ***25***.
- Specified conference must be in use.
- Command must be consistent with the conference mode.
- Specified endpoint must be joined to the conference, and it cannot be an audio-only or UCC party.

Unsuppressing Video for all Endpoints

Syntax:

control conference *CONFERENCE_ID* unsuppress-all-video

Description:

This command unsuppresses the video for all the endpoints for which video was suppressed via the “suppress-video” command.

For the command to be successful, the following must be true:

- ***CONFERENCE_ID*** must be all digits and in the range from ***1*** through ***200***.
- Specified conference must be in use.

Redialing an Endpoint

Syntax:

control conference *CONFERENCE_ID* redial-endpoint *ENDPOINT_ID*

Description:

This command redials this dial-out endpoint. The effect is identical to the redial flag being set via the Conference Record form.

For the command to be successful, the following must be true:

- ***CONFERENCE_ID*** must be all digits and in the range from ***1*** through ***200***.
- ***ENDPOINT_ID*** must be all digits and in the range from ***1*** through ***25***.
- Specified conference must be active.
- Specified endpoint must be a dial-out endpoint.

Switching a CPP or Executive Conferencing Conference to Full-Screen Video

Syntax:

control conference *CONFERENCE_ID* full-screen

Description:

This command switches a CPP or Executive Conferencing conference to full-screen video.

For the command to be successful, the following must be true:

- ***CONFERENCE_ID*** must be all digits and in the range from ***1*** through ***200***.
- Specified conference must be in use, and it must be able to support Continuous Presence *Plus* or Executive Conferencing.

Switching a CPP Conference to Quad-Screen

Syntax:

control conference *CONFERENCE_ID* quad-screen

Description:

This command switches a CPP conference back to quad-screen video (if quad-screen resources are still available).

For the command to be successful, the following must be true:

- ***CONFERENCE_ID*** must be all digits and in the range from ***1*** through ***200***.
- Specified conference must be in use, and it must be able to support Continuous Presence *Plus*.

Switching a CPP Conference to Dual-Panoramic

Syntax:

control conference *CONFERENCE_ID* dual-panoramic

Description:

This command switches a CPP conference to dual-panoramic video (if dual-panoramic resources are still available).

For the command to be successful, the following must be true:

- ***CONFERENCE_ID*** must be all digits and in the range from ***1*** through ***200***.
- Specified conference must be in use, and it must be able to support Continuous Presence *Plus*.

Switching a CPP Conference to Lower-Panoramic

Syntax:

control conference *CONFERENCE_ID* lower-panoramic

Description:

This command switches a CPP conference to lower-panoramic video (if lower-panoramic resources are still available).

For the command to be successful, the following must be true:

- ***CONFERENCE_ID*** must be all digits and in the range from ***1*** through ***200***.
- Specified conference must be in use, and it must be able to support Continuous Presence *Plus*.

Switching a CPP Conference to Upper-Panoramic

Syntax:

control conference *CONFERENCE_ID* upper-panoramic

Description:

This command switches a CPP conference to upper-panoramic video (if upper-panoramic resources are still available).

For the command to be successful, the following must be true:

- ***CONFERENCE_ID*** must be all digits and in the range from ***1*** through ***200***.
- Specified conference must be in use, and it must be able to support Continuous Presence *Plus*.

Switching a Full-Screen Conference to Executive Conferencing

Syntax:

control conference *CONFERENCE_ID* executive

Description:

This command switches a full-screen conference to Executive Conferencing (if Executive Conferencing resources are still available).

For the command to be successful, the following must be true:

- ***CONFERENCE_ID*** must be all digits and in the range from ***1*** through ***200***.
- Specified conference must be in use, and it must be able to support Executive Conferencing.

Forcing CIF Video

Syntax:

control conference *CONFERENCE_ID* cif-floor

Description:

This command forces the video mode to CIF, and it takes effect when viewing full-screen video. The highest common MPI rate (auto) is implied.

For the command to be successful, the following must be true:

- ***CONFERENCE_ID*** must be all digits and in the range from ***1*** through ***200***.
- Specified conference must be in use, and it must be able to support H.261.

Forcing QCIF Video

Syntax:

control conference *CONFERENCE_ID* qcif-now

Description:

This command forces the video mode to QCIF, and it takes effect when viewing full-screen video. The highest common MPI rate (auto) is implied.

For the command to be successful, the following must be true:

- ***CONFERENCE_ID*** must be all digits and in the range from ***1*** through ***200***.
- Specified conference must be in use, and it must be able to support H.261.

Forcing an Audio Mode

Syntax:

control conference *CONFERENCE_ID* audio-mode *AUDIO_MODE*

Description:

This command forces the conference to the specified audio mode (***g.711***, ***g.722***, or ***g.728***). If the administered audio mode is not ***g.722***, any changes to ***g.722*** are disallowed unless an “auto” audio mode is administered and the transfer rate is greater than 2B. Audio changes are not allowed for data conferences.

Values for ***AUDIO_MODE*** depend on the administered transfer rate and audio mode of the conference when the conference becomes active (that is, after the first endpoint joins the conference).

The following tables show the allowable values when the ClearPresence @716 RTU is optioned and when it is not optioned.

NOTE:
The ***AUDIO_MODE*** values for the ClearPresence @716 RTU not optioned are identical for those for the R5.0 MCU.

Table 4-3. Values for the *AUDIO-MODE* Parameter of the Conference Control Audio-Mode Command (ClearPresence @716 RTU Not Optioned)

Audio Mode **Field Value Administered in Conference Record Form**	**Transfer Rate**	**Values Allowed for *AUDIO-MODE* Parameter***	**Values Not Allowed for *AUDIO-MODE* Parameter***
G.722	any	***G.722, G.711***	***G.728***
G.728	any	***G.728, G.711***	***G.722***
G.711	any	***G.728, G.711***	***G.722***
auto-down_only or **auto-w/upgrade**	less than or equal to 2B	***G.728, G.711***	***G.722***
auto-down_only or **auto-w/upgrade**	greater than 2B	***G.722, G.711***	***G.728***

***** Values ***auto-down_only*** and ***auto-w/upgrade*** are not allowed for any conference.

Table 4-4. Values for the *AUDIO-MODE* Parameter of the Conference Control Audio-Mode Command (ClearPresence @716 RTU Optioned)

***Audio Mode* Field Value Administered in Conference Record Form**	**Transfer Rate**	**Values Allowed for *AUDIO-MODE* Parameter***	**Values Not Allowed for *AUDIO-MODE* Parameter***
ClearPres716	any	***ClearPres716, G.722, G.728, G.711***	***-----***
G.722	any	***ClearPres716, G.722, G.728, G.711***	***-----***
G.728	any	***G.728, G.711***	***ClearPres716, G.722***
G.711	any	***G.728, G.711***	***ClearPres716, G.722***
auto-down_only or **auto-w/upgrade**	less than or equal to 2B	***ClearPres716, G.722, G.728, G.711***	***-----***
auto-down_only or **auto-w/upgrade**	greater than 2B	***ClearPres716, G.722, G.728, G.711***	***-----***

***** Values ***auto-down_only*** and ***auto-w/upgrade*** are not allowed for any conference.

For the command to be successful, the following must be true:

- ***CONFERENCE_ID*** must be all digits and in the range from ***1*** through ***200***.
- Specified conference must be in use, and it must be able to support H.261.
- ***AUDIO_MODE*** must be ***ClearPres716***, ***g.711***, ***g.722***, or ***g.728.***
- Resources must be available for a conference administered in g.722 audio mode.
- Conference cannot be a data conference.

Moving an Endpoint to Another Conference

Syntax:

control conference *CNF_ ID -From* move *EPT_ID-From CNF_ID-To*

This command can do the following:

- Move a conferee to his or her destination conference (by entering **control conference *CNF_ID-From* move *EPT_ID-From***).
- Move a conferee to another conference with the option of specifying a specific endpoint index (by entering **control conference *CNF_ID-From* move *EPT_ID-From CNF_ID-To***.

For the command to be successful, the following must be true:

- Specified conferences must be in use.
- ***CNF_ID-From*** and ***CNF_ID-To*** omust be all digits and in the range from***1*** through ***200***.
- ***EPT_ID-From*** and ***EPT_ID-To*** must be all digits and in the range from ***1*** through ***25***.

Reporting Conference Record Translation Changes

Syntax:

control conference *CONFERENCE_ID* xln-clear-flag

NOTE:
This command is used exclusively with CRCS OpCenter.

Whenever a translation for a conference record is changed, the corresponding conference translation flag is set and reported in the Status All Conference form. This allows the standalone OpCenter to extract the translation change. Use this command to clear the translation change flag for the specified conference once the standalone OpCenter has the new translation.

For the command to be successful, the following must be true:

- ***CONFERENCE_ID*** must be all digits and in the range from ***1*** through ***200***.

OpCenter

OpCenter is a PC-based real-time graphical user interface (GUI) within the CRCS platform that enables conference management. OpCenter allows agents to find and correct typical multipoint conferencing problems quickly and also provide value-added services. For more information, refer to the appropriate CRCS documentation.

Diagnosing and Solving Problems

5

Overview

The reservations agent, system administrator, and the Video Technical Center (VTC)/Customer Care Center are responsible for troubleshooting MCU conferences. As the primary contact for making MCU reservations, the reservations agent is the first person whom conferees and conference conveners will call when something goes wrong with their conference. This section describes how to resolve scheduling conflicts and diagnose minor problems. Such problems can include the following:

- User is unable to join or remain joined to a conference, or is not dropped from the conference.
- Conference has poor video quality as a result of its being downgraded due to automatic algorithms.
- User is not receiving full service (for example, the endpoint is administered as an audio-only endpoint and not with “mlp-var” or “ld-celp” [G.728]).
- Dial-out call (including that for a cascade link or for other types of multimedia users) is unable to be joined by the user’s endpoint equipment.
- Audio-only endpoint is unable to join or remain joined to the conference or is not dropped from the conference.
- Conferee is not seen by other users due to interworking problems.
- User is not able to participate in a Multipoint Communication Service (data) conference.

When conferees call to report a problem, you need to isolate the problem before you attempt to resolve it or escalate it. Since the MCU brings together a variety of video endpoints through the network (sometimes through another telephone system), it may take some investigating to figure out where the trouble lies. This troubleshooting section details some of the most likely possibilities and best actions to take.

NOTE:
Many of the problems in this chapter relevant to Dial-Out and Cascading can be resolved only during an active conference and if Dynamic Resizing is allowed (customer option). Without dynamic resizing, modifications to the Conference form can be made only to the *Stop Time* and *Class* fields. Specifically, the *Class* field must be changed to “file” to stop the conference. Then, once the proper modifications are made, this field must be changed back to “reserved” or “dedicated.”

NOTE:
Dial-out problems can be retried by setting the *Redial?* flag in the Conference Record to **y**.

There are three commands available for troubleshooting an MCU conference, including:

- **status conference**
- **status conference endpoint**
- **monitor conference**

The following sections discuss and illustrate these commands. Thereafter, several sections discuss troubleshooting MCU feature-related problems.

Status of Conference Screen

The Status of Conference screen displays information about active, in-use and completed conferences that can be used to diagnose a problem. You can display data for a number of conferences. In such a case, active conferences are displayed first (in order of conference ID), followed by completed conferences (in order of completion). If you reserve a conference that has not yet begun using the same conference ID as a completed conference, the Status of Conference screen will show the completed conference until the reserved conference has begun. (There is no data or information for conferences that have not yet begun.) For a completed conference, the *Status* field in the screen shows **complete**.

The data for each conference is displayed in two parts. The first screen describes the status of the conference, and it indicates the modes and levels of the conference. The rest of the first screen and the second screen show information concerning endpoints (for example, which endpoints are in use and which endpoint(s) (if any) caused the conference operating mode to change). The rest of the screens display endpoint level data, including ports and drop reasons. Data for up to six endpoints per screen are displayed.

The command syntax for accessing the Status of Conference screen is **status conference *[all | conference-ID] [print | schedule]***. The default for the conference ID is **all**, which indicates all the stored data. Value ***conference ID*** is the number of the conference. A conference ID uniquely identifies a current

conference; however, a completed conference may have the same conference ID. Nonetheless, this ID causes the system to display a current conference with that ID. If there is no active or in-use conference, the most recently completed conference with that ID is displayed. Value **print** sends the screen to the printer attached to the MCU-ST. Value **schedule** sends the screen to the system printer at the specified time.

To view the screens for all conferences, enter **status conference** or **status conference *all***. You can add **print** to any of the preceding command lines (for example, **status conference *conference-ID* print)** to print a copy of the Status of Conference screen.

The first screen appears whenever you enter **status conference** and at least one valid conference is found. You can use the “Page” keys to browse through all the screens for a conference. The “Next Form” key enables you to move to the next conference (if any). To exit, press “Cancel.” Note that the data displayed is whatever the system had stored at the time the command was entered. To see the latest data, enter the command again. A repeat feature allows you to reissue the previously executed command. To do so, enter “R” followed by a return.

```
status conference                                                page  1 of  6


   STATUS OF CONFERENCE: 2__       Status: in-use__        Conf Type: ____________
Conference Name: LuTech 12/11/98_____                 Conference Mode: voice-activated
      Password: 112358__________                                Chair: __
 Password Scope: audio_               Cascade Mode: primary__     Audio Mode: G.711-A_
         Class: reserved_                Data Mode: none_____       MLP Rate: _______
Start Time/Date: 12:00 DEC 11       Admin Bandwidth: 64k__       Lo/Hi Interworking? n
 Stop Time/Date: 14:00 DEC 11        Conf Bandwidth: 56k__          Rate Adaptation? y
Video Processor: ____ ________       Format(in/out): CIF_____           SM Rate 1: _____
 VP Audio Delay: ____        FPS 16C: ___ 4C: ___ C: 30_ Q: 30_                2: _____
Q1: __  Q2: __  Fill Ept: __             Video Mode: H.261-down_only           3: _____
Q3: __  Q4: __   Vid Mix: full          H.263/H.261 Transcoding? n          FECC: __
                                Dial In ----Capability-----   SM Rt Bd
       Type  Ext  Meet-Me Number  Type Use Ch Au Vid T120 CC Gx Rt Ad Md Ts Vs
    1: P64_ 75000 (700)957-5939__  in_  y  y  y  y 1  _   _  y  _  y  _  t  b
    2: P64_ 75001 (700)957-5938__  in_  n  _  _  y 1  _   _  _  _  _  _  _  _
    3: AUD_ 75002 (700)957-5937__  in_  y  _  y  _ _  _   _  _  _  _  _  _  _
    4: CAS_ 75003 (700)957-5936__  out  y  y  y  y 1  _   _  y  _  y  _  t  r
    5: ____ _____ _______________  ___  _  _  _  _ _  _   _  _  _  _  _  _  _
    6: P64_ 75004 (700)957-5935__  out  c  y  c  y 1  _   _  e  _  y  _  _  _
    7: P64_ 75005 (700)957-5940__  out  c  y  y  y 1  _   _  y  _  c  _  t  _
    8: P64_ 75006 (700)957-5945__  in_  y  y  y  y 1  _   _  e  _  y  _  _  _
```

Figure 5-1. Sample Status of Conference Screen, Page 1

```
status conference                                              page 2 of 6

                    STATUS OF CONFERENCE ____    Status: ______

                                    Dial In  ----Capability----   SM Rt Bd
    Type  Ext  Meet-Me Number       Type Use Ch Au Vid T120 CC Gx Rt Ad Md Ts Vs
 9: ___ _____ ________________       ___  _   _  _  _ _  _    _  _  _  _  _  __
10: ___ _____ ________________       ___  _   _  _  _ _  _    _  _  _  _  _  __
11: ___ _____ ________________       ___  _   _  _  _ _  _    _  _  _  _  _  __
12: ___ _____ ________________       ___  _   _  _  _ _  _    _  _  _  _  _  __
13: ___ _____ ________________       ___  _   _  _  _ _  _    _  _  _  _  _  __
14: ___ _____ ________________       ___  _   _  _  _ _  _    _  _  _  _  _  __
15: ___ _____ ________________       ___  _   _  _  _ _  _    _  _  _  _  _  __
16: ___ _____ ________________       ___  _   _  _  _ _  _    _  _  _  _  _  __
17: ___ _____ ________________       ___  _   _  _  _ _  _    _  _  _  _  _  __
18: ___ _____ ________________       ___  _   _  _  _ _  _    _  _  _  _  _  __
19: ___ _____ ________________       ___  _   _  _  _ _  _    _  _  _  _  _  __
20: ___ _____ ________________       ___  _   _  _  _ _  _    _  _  _  _  _  __
21: ___ _____ ________________       ___  _   _  _  _ _  _    _  _  _  _  _  __
22: ___ _____ ________________       ___  _   _  _  _ _  _    _  _  _  _  _  __
23: ___ _____ ________________       ___  _   _  _  _ _  _    _  _  _  _  _  __
24: ___ _____ ________________       ___  _   _  _  _ _  _    _  _  _  _  _  __
25: ___ _____ ________________       ___  _   _  _  _ _  _    _  _  _  _  _  __
```

Figure 5-2. Sample Status of Conference Screen, Page 2

```
status conference                                              page 3 of 6

                          STATUS OF CONFERENCE 2__

                    Sum Grp: Group1  Group2  Group3  Group4
                         L1: 1a0716  1a0714  _______  _______
                         L2: 1a0715  1a0713  _______  _______

        Join  Drop      Drop     AC  -------Ports---------  Sum
    Ext Time  Time      Reason   Num Trunk  Video  Aud/ESM  Grp Software
1: 75000 12:01 _____ ___________ ___ 1a1201 1a0801 1a0905    1  _________
         12:01 _____ ___________ ___ 1a1202 1a0802 _______       _________
2: 75001 12:06 12:07 Handshake_ ___ _______ _______ _______    _  _________
         _____ _____ ___________ ___ _______ _______ _______       _________
3: 75002 12:02 _____ ___________ ___ 1a1203 _______ 1a0907    _  _________
         _____ _____ ___________ ___ _______ _______ _______       _________
4: 75003 12:01 _____ ___________ _12 1a1215 1a0805 1a0908    1  _________
         12:01 _____ ___________ _13 1a1216 1a0806 _______       _________
5: _____ _____ _____ ___________ ___ _______ _______ _______    _  _________
         _____ _____ ___________ ___ _______ _______ _______       _________
6: 75004 12:02 _____ ___________ _14 1a1205 1a0803 1a0906    1  _________
         12:03 _____ ___________ _15 1a1206 1a0804 _______       _________
```

Figure 5-3. Sample Status of Conference Screen, Page 3

```
status conference                                              page 4 of 6

                              STATUS OF CONFERENCE 2__

          Join  Drop      Drop      AC  -------Ports--------  Sum
     Ext  Time  Time     Reason     Num Trunk   Video Aud/ESM  Grp Software
 7:  75005 12:01 _____ __________ _16 1a1001 1a0807 1a1107    1  ________
           12:01 _____ __________ _17 1a1002 1a0808 ______       ________
 8:  75006 12:01 _____ __________ ___ 1a1005 1a0807 1a1108    1  ________
           12:01 _____ __________ ___ 1a1006 1a0808 ______       ________
 9:  75007 12:01 _____ __________ ___ 1a1007 1a0807 1a1105    1  ________
           12:01 _____ __________ ___ 1a1008 1a0808 ______       ________
10:  75008 12:01 _____ __________ _18 1a1202 1a0810 1a1109    2  ________
           12:01 _____ __________ _19 1a1203 1a0811 ______       ________
11:  _____ _____ _____ __________ ___ ______ ______ ______    _  ________
           _____ _____ __________ ___ ______ ______ ______       ________
12:  _____ _____ _____ __________ ___ ______ ______ ______    _  ________
           _____ _____ __________ ___ ______ ______ ______       ________
13:  _____ _____ _____ __________ ___ ______ ______ ______    _  ________
           _____ _____ __________ ___ ______ ______ ______       ________
14:  _____ _____ _____ __________ ___ ______ ______ ______    _  ________
           _____ _____ __________ ___ ______ ______ ______       ________
```

Figure 5-4. Sample Status of Conference Screen, Page 4

NOTE:
Pages 4 through 6 of the Status of Conference screen display the same information as Page 3 when there are more than six MCU-extensions assigned to a conference.

```
status conference                                          page 7 of 7

              STATUS OF CONFERENCE ___          Endpoint/VP Info
   In       Fr  Ept                              In       Fr  Ept
   Use Vs   Err VCU                              Use Vs   Err VCU
 1: _  __   _______                           13: _  __   _______
 2: _  __   _______                           14: _  __   _______
 3: _  __   _______                           15: _  __   _______
 4: _  __   _______                           16: _  __   _______
 5: _  __   _______                           17: _  __   _______
 6: _  __   _______                           18: _  __   _______
 7: _  __   _______                           19: _  __   _______
 8: _  __   _______                           20: _  __   _______
 9: _  __   _______                           21: _  __   _______
10: _  __   _______                           22: _  __   _______
11: _  __   _______                           23: _  __   _______
12: _  __   _______                           24: _  __   _______
                        VP Decoder:  1  2  3  4  5
                        VP Dec Vcu: __ __ __ __ __
                        VP Dec Ept: __ __ __ __ __
                    VP Dec Framing: _  _  _  _  _
```

Figure 5-5. Sample Status of Conference Screen, Page 7

The following list describes the fields in the screen:

- *Conference ID*

  Display-only field that shows the number (**1** through **200**) that identifies this conference.

- *Conference Name*

  The purpose of the meeting for identification purposes. This is chosen by the convener.

- *Password Scope*

  If this field is blank, no password is required. **Audio** indicates that the per-conference password is in effect and is required only for an Audio Add-On endpoint. If the field is set to **all**, the per-conference password is in effect and is required for all endpoints.

  NOTE:
  The value “all” does not appear on the form if no H.320 endpoints in the conference support passwords.

- *Password*

  Identifies the per-conference password. Per-conference passwords are not required to gain entry into a conference unless the per-conference or per-user password is set. If the per-conference password is set, each endpoint entering the conference is checked against the *Password Scope* field to determine whether password validation is done (see *Password Scope*). The per-conference password is blank on the Status Conference form if *Password Scope* is blank or **per-user**. Per-user passwords are not displayed on the Status Conference forms.

- *Class*

  The choices are **reserved** or **dedicated**. A **reserved** conference is a conference that is scheduled to begin within 24 hours of the current time. A **dedicated** conference is a conference that is always available and can be joined at any time.

- *Start Time/Date*

  Indicates the starting time for the conference (in 24-hour notation) along with the month and calendar day.

- *Stop Time/Date*

  Indicates the ending time for the conference (in 24-hour notation) along with the month and calendar day. If *Status* (discussed later in this list) contains **complete**, the actual ending time is displayed. Otherwise, the field is blank.

- *Chair*

  Identifies current chair token holder. Value is an endpoint from **1** through **25** corresponding to the slot number associated with the endpoint.

- *FECC*

  Identifies the current camera control (LSD/H.224) token handler. Value is an endpoint from **1** through **25** corresponding to the slot number associated with the endpoint.

- *Format (in/out)*

  Indicates the video format for single screen or multi-screen conferences. For SM conferences, indicates the video format for the primary rate of the conference. Values include **16CIF**, **4CIF**, **CIF**, **QCIF**, **QCIF-SQCIF**, **H.CTX**, **H.CTX+**, **SG4**, and **MRV**.

- *FPS 16C*

  Indicates the rate (frames per second) at which an H.263-only endpoint can receive 16CIF frames. Values include **' '**, **30**, **15**, **10**, **7.5**, **6**, **5**, **3**, **2**, and **1**.

NOTE:
There is no indication of either the maximum transmit frame rate or current frame rate that can be detected by the MCU. In practice, the frame rate changes as a function of the amount of motion in the input image.

The field is always blank for proprietary modes, H.261 modes, and conferences using a video processor (including CPP, SM, and same rate H.263/H.261 transcoding conferences). The field is also blank for the 4CIF, CIF, and QCIF video formats.

- *FPS 4C*

  Indicates the rate (frames per second) at which an H.263-only endpoint can receive 4CIF frames. Values include **' '**, **30**, **15**, **10**, **7.5**, **6**, **5**, **3**, **2**, and **1**.

  NOTE:
  There is no indication of either the maximum transmit frame rate or current frame rate that can be detected by the MCU. In practice, the frame rate changes as a function of the amount of motion in the input image.

  The field is always blank for proprietary modes, H.261 modes, and conferences using a video processor (including CPP, SM, and same rate H.263/H.261 transcoding conferences). The field is also blank for the 4CIF, CIF, and QCIF video formats.

- *FPS C*

  Indicates the rate (frames per second) at which an H.263-only endpoint can receive CIF frames. Values include **' '**, **30**, **15**, **10**, **7.5**, **6**, **5**, **3**, **2**, and **1**. Values **6**, **5**, **3**, **2**, and **1** apply only to H.263 video mode; values **' '**, **30**, **15**, **10**, and **7.5** apply to both H.263 video mode and H.261 video mode.

  NOTE:
  There is no indication of either the maximum transmit frame rate or current frame rate that can be detected by the MCU. In practice, the frame rate changes as a function of the amount of motion in the input image.

  For conferences using a video processor (including CPP, SM, and same rate H.263/H.261 transcoding conferences), the field indicates the highest common CIF frame rate of all the endpoints and the rate of the video mixer board (which may be lower than that of the rest of the participants).

  The field also contains the frame rate value for proprietary modes.

  The field is blank for QCIF full-screen conferences or QCIF CPP conferences using the TN2237 Video Processor circuit pack. For Presentation mode quad-screen conferences, the field indicates the highest common CIF frame rate of all the participants.

- *FPS Q*

  Indicates the rate (frames per second) at which an H.263-only endpoint can receive QCIF frames. Values include **' '**, **30**, **15**, **10**, **7.5**, **6**, **5**, **3**, **2**, and **1**. Values **6**, **5**, **3**, **2**, and **1** apply only to H.263 video mode; values **' '**, **30**, **15**, **10**, and **7.5** apply to both H.263 video mode and H.261 video mode.

  For conferences using a video processor (including CPP, SM, and same rate H.263/H.261 transcoding conferences), the field indicates the highest common QCIF frame rate of all the endpoints and the rate of the video mixer board (which may be lower than that of the rest of the participants).

  NOTE:
  The QFPS frame rate cannot be greater than the CIF frame rate announced by the presenter.

  The field is blank for proprietary modes.

NOTE:
For SM conferences, the previous five fields are blank if no endpoints dial in at the administered rate. Also, if one or more endpoints dial in at this rate, all endpoints administered at this rate display the same values. This is true even for endpoints that have not joined the conference.

- *Video Mode*

  The video mode of a non-cascaded conference can be changed during an active conference via administration or conference control. Conference control does not modify the administered video mode. Each video mode except “H.261-CIF-Floor” and “H.261-QCIF-Now” can be set only via administration. These two video modes can also be set via conference control. The following are video mode values:

  — **H.263/1-down_only** — indicates the highest common video mode that allows either its frame rate or video to be downgraded to H.261 video mode. Whenever an endpoint that caused a video downgrade disconnects from the conference, no attempt is made to upgrade video quality. This video mode can downgrade to H.261. The format and associated frame rates for the H.261 and H.263 video modes can also be downgraded.

  — **H.263/1-w/upgrade** — indicates the highest common video mode that allows either its frame rate or video to be downgraded (to H.261) or upgraded video (to H.263) depending on the capabilities of the endpoints currently in the conference. This video mode can downgrade to H.261. The format and associated frame rates for the H.261 and H.263 video modes can also be downgraded.

  — **H.261-down_only** — indicates the highest common video mode that allows only a downgrade of video. Whenever an endpoint that caused a video downgrade disconnects from the conference, no attempt is made to upgrade the video quality.

— **H.261-w/upgrade** — indicates the highest common video mode that allows a downgrade or an upgrade of the video depending on the capabilities of the endpoints currently in the conference.

— **H.261/3-down_only** — Indicates a downgrade from H.263/1-down_only; otherwise, provides the same functionality as "H.261-down_only." Indicates the highest common video mode that allows only downgrade of video. Whenever an endpoint that caused this video downgrade disconnects from the conference, no attempt is made to upgrade the video quality to H.263. Also, no attempt is made to upgrade the format and associated frame rates for the video mode.

— **H.261/3-w/upgrade** — Indicates a downgrade from H.263/1-w/upgrade; otherwise, provides the same functionality as "H.261-w/upgrade." Indicates the highest common video mode that allows either a downgrade or an upgrade of video depending on the capabilities of the endpoints currently in the conference. This video mode can upgrade the conference to H.263-w/upgrade video mode if all the endpoints causing the downgrade disconnect from the conference.

— **H.261-CIF-Floor** — forces the video format to CIF. CIF frame rate floor may be specified via administration. Whenever it is changed via conference control, the CIF frame rate is assumed to be "auto."

— **H.261-CIF-Now** — forces the video to CIF, and forces the CIF frame rate to a value specified via administration.

— **H.261-QCIF-Now** — forces the video to QCIF, and forces the QCIF frame rate to a value specified via administration. Whenever it is changed via conference control, the QCIF frame rate is assumed to be "auto."

— **H.CTX** — indicates the H.CTX proprietary video format.

— **H.CTX+** — indicates the H.CTX+ proprietary format.

— **MRV** — indicates that the video mode of the conference is MRV video mode. If the conference is using MRV video, the values for the CIF Minimum Picture Intervals reflect the current maximum frame rate at the MCU.

— **SG4** — indicates the SG4 proprietary video format.

■ *Q1* through *Q4*

Indicate the quadrants. Values include **1** through **25** (representing the endpoints that are part of the quad image), **F** (indicating that the displayed video is a default board "fill" image), or **FE** (indicating that the displayed video is that of the fill endpoint). (See the explanaton for the *Fill Ept* field.)

Also, the *Q4* field displays **XX** for a CPP upper-panoramic, CPP lower-panoramic, or Executive Conferencing screen layout; the *Q3* and *Q4* fields display **XX** for a dual-panoramic screen layout; and the *Q1* through

Q4 fields display **XX** for a CPP full-screen video screen layout. In each case, value **XX** indicates that the relevant quadrant(s) are not being used currently.

- *Broadcasters 1:...2:...3:*

  Replaces the *Q1* through *Q4* field whenever *Executive Conferencing* is set to **y**. Identifies the endpoint association with decoders 1, 2, and 3 of the Video Processor circuit pack.

- *Fill Ept*

  Indicates the fill endpoint. Values include **1** through **25.** Whenever the video of an endpoint that is fixed in a quadrant cannot be used as a valid video source (because the endpoint has not joined the conference, has disconnected from the conference, or has invalid video), a fill video is displayed in the designated quadrant. The fill can be another endpoint's video or the default board fill image. Value blank indicates a default fill.

  Fill video is also displayed in VAS quadrants whenever not enough endpoints have joined the conference.

- *Vid Mix*

  Indicates the picture layout for the conference. The following list identifies and discusses the values for this field.

  — blank — indicates a non-CPP full-screen conference. For this value, the *Q1* through *Q4* fields are also blank.

  — **exec** — indicates an Executive Conferencing conference in mixed image mode. For this value, the *Q1* through *Q3* fields are used to indicate the Executive Conferencing broadcast endpoints. Also, the *Q4* field displays **XX** to indicate that this quadrant is currently not being used.

  — **full** — indicates a CPP conference in full-screen mode. For this value, the *Q1* through *Q4* fields display **XX** to indicate that the four quadrants are currently not being used.

  — **dual** — indicates a dual-panoramic mixed-image conference. For this value, the *Q3* and *Q4* fields display **XX** to indicate that they are currently not being used.

  — **up-panor** — indicates a three-image conference where the top half of the screen displays one image while the bottom half of the screen is segmented into two screen displays. For this value, the *Q4* field displays **XX** to indicate that the fourth quadrant is currently not being used.

  — **lo-panor** — indicates a three-image conference where the bottom half of the screen displays one image while the top half of the screen is segmented into two screen displays. For this value, the *Q4* field displays **XX** to indicate that the fourth quadrant is currently not being used.

  — **quad** — indicates a quad-screen mixed-mode conference.

- *H.263/H.261 Transcoding*

  Indicates if H.263/H.261 transcoding is performed within the same rate. Values include **y** and **n**. Value **y** also indicates that TN2237 video processor resources are being used.

- *Video Processor*

  This field has two components. The first component is the board address for the TN2237 resource handling all four quadrants. Whenever there are problems with the video processor resources, a failure reason (the second component) is displayed along with the board address.

  The following list identifies and defines the keyword values associated with Continuous Presence *Plus* status:

  — blank — video processor resources were successfully allocated for the CPP, SM, or same rate H.263/H.261 transcoding conference.

  — **mtcbusy** — available resources were maintenance busy at the start of the conference.

  — **went oos** — an active video processor resource went OOS during an active CPP, SM, or same rate H.263/H.261 transcoding conference. For the TN2237 resource, this indicates a failure of the Video Processor circuit pack.

  — **time slot** — Continuous Presence *Plus* time slot resources are not available at the start of the conference.

  — **not ready** — the TN2237 Video Processor circuit pack is not ready for connection.

- *VP Audio Delay*

  Indicates the current conference audio delay for multimedia endpoints associated with the primary rate of the conference. The field value is blank for full-screen (non-CPP) non-SM conferences and CPP non-SM conferences in full-screen mode.

  For CPP conferences in quad mode, the value of the delay depends on the value for the *Lip Sync Setting* field in the Conference Record form. The following table illustrates the appropriate variations:

Lip Sync Setting Value	Corresponding CPP Audio Delay Value
“none”	**0**
“auto”	Value uplinked by the TN2237 circuit pack
“custom”	Same as “custom” value (0 through 1000 ms)

- *Status*

  This display-only field reports the current status of a conference as it relates to the *Class* of the conference. A reserved conference can be **active, in-use**, or **complete**. **Active** means the conference is available to receive calls. **In-use** means the conference is in progress and one or more conferees has joined in. **Complete** means a conference has ended (the *Stop Time* has been reached). A dedicated conference is always **active** (available to receive calls).

- *Conference Mode*

  Depending on your MCU model, this field offers up to four conference modes. **Voice-activated** allows conferees to view the current speaker. The video source changes when another speaker begins talking. **Chair** allows the chairperson to determine which video-sending endpoint is broadcast to all the other conferees. **Presentation** mode allows one location to be viewed constantly by all conferees. If a conferee asks a question or makes a comment, the other conferees hear the speaker but continue to view the presenter. **Broadcast/scan** mode allows one location to be viewed and heard constantly by all conferees. The broadcaster receives video (not audio) from all conferees on a rotating basis.

- *Control Ext.*

  This field is displayed only if the operating mode has a controlling endpoint.

- *Scn Int.*

  This field appears only when the *Conference Mode* selected is **Broadcast/scan**. This is the amount of time (in seconds) that the broadcaster will view each conferee before the MCU switches to the next conferee.

- *Cascade Mode*

  If the conference includes a CAS link, this mode must be either **primary** or **secondary.** On the other hand, if a cascaded endpoint cannot be administered, the mode must be **blank.** Finally, if endpoints can be administered but the link is not enabled, the mode is **disabled.**

- *Audio Mode*

  This field identifies the current operating audio mode from among the following: **G.711** (PCM-A), **G.711** (PCM-mu), **G.728** (LD-CELP), **G**.**722** (7kHz), or **CP716** (ClearPresence @ 716).

The following list assumes that the *ClearPresence @716 RTU is not optioned*, and it discusses the interactions between G.711, G.728, and G.722 along with the results generated by including into the *Audio Mode* field of the Conference Record the following values, coupled with the accompanying circumstance:

- — ***auto-down_only*** or ***auto-w/upgrade*** *and the conference bandwidth is 112k or 128k* — the system starts out with the highest common audio, G.728 (LD-CELP). When a G.711 (PCM) endpoint joins such a conference, the conference audio mode is downgraded to G.711 (PCM). If the endpoint that caused the downgrade to G.711 disconnects, and if the conference is administered with ***auto-w/upgrade***, the conference is upgraded to G.728.
- — ***auto-down_only*** or ***auto-w/upgrade*** *and the conference bandwidth is greater than 128k* —the system starts out with the highest common audio, G.722 (7kHz). When a G.711 (PCM) endpoint joins such a conference, the conference audio mode is retained, and the PCM endpoint interworks with the conference. The *Capability...Aud* field displays value **e** to indicate G.722/G.711 interworking.
- — ***G.722*** *(7kHz), and a G.711 (PCM) endpoint joins such a conference* — the conference audio mode is not downgraded to G.711 (PCM). The Status Conference form indicates that such an endpoint does not support G.722 (7kHz) audio by displaying **e** in the *Capability...Aud* field. In this case, value **e** implies G.711/G.722 interworking.
- — ***G.728*** *(LD-CELP), and a G.711 (PCM) endpoint joins such a conference* — the Status Conference form indicates that such an endpoint does not support G.728 (LD-CELP) by displaying **e** in the *Capability...Aud* field. In this case, value **e** implies G.728/G.711 interworking. This type of endpoint is an invalid video source. The MCU may or may not send video to it.

For information on interactions based on the ClearPresence @716 RTU optioned (value **CP716**), refer to the “ClearPresence@716” section in Appendix B.

- *Data Mode*

  Indicates the data mode capability for this conference. Entries include **none** and **any-mlp.**

- *MLP Rate*

  Indicates the MLP Data Rate for the conference. Entries include **var-MLP, 6.4k, 8k, 14.4k, 16k, 22.4k, 24k, 30.4k, 32k, 38.4k, 40k, 46k,** and blank.

- *Admin Bandwidth*

  This field is used to specify the bandwidth selected for the conference. Values include **1x56k**, **1x64k**, **2x56k**, **2x64k**, **112k, 128k, 168k, 192k, 224k, 256k, 280k, 320k, 336k, 448k, 512k, 672k, 768k, 1008k, 1152k, 1288k, 1344k, 1472k, 1536k, 1680k,** and **1920k**.

- *Conf Bandwidth*

  Indicates the current operating channel conference bandwidth. This value can differ from the administered bandwidth due to Rate Adaptation.

- *Rate Adaptation*

  Indicates whether the conference supports rate adapation. Value is **y** or **n**.

- *Lo/Hi Interworking*

  Indicates whether the conference supports Low-/High-Speed Interworking. Value is **y** or **n**.

- *SM Rate 1 (2, 3)*

  Identifies the additional Speed Match rates for the conference. Values include blank, **1x56k**, **1x64k**, **2x56k**, **2x64k**, **112k**, **128k**, **224k**, **256k**, **336k**, **384k**, **448k**, **512k**, **672k**, and **768k**. Within the *Ept Rt* field, values **1**, **2**, and **3** are used to associate endpoints with the Speed Match rates 1, 2, and 3 identified in the *SM Rate 1 (2, 3)* field.

- *Type*

  Indicates the type of conferee. Possible entries include **P64** (audio/video), **AUD** (Audio Add-On), **CAS** (cascade link), **BOND** (BONDing), **DA** (Dedicated Access), **BCAS** (cascade BONDing), or **UCC** (Universal Conference Control).

  NOTE:
  BONDing calls use up to 12B-channels to form one H.261 channel.

  A maximum of one UCC terminal per MCU per conference is allowed.

- *Dial Type*

  Indicates whether dial-in **(in)** or dial-out **(out)** is being used to join the endpoint to the conference.

- *Ext*

  Indicates the extension for the endpoint. This extension is selected at administration time. For dial-in calls, the extension is the Lucent Technologies MCU internal mapping for the external Meet-Me Number that the user dials. For dial-out calls, the extension is the MCU-extension used to originate the call.

- *Meet-Me Number*

  The MCU automatically displays the data entered on the MCU-extension form in the associated Meet-Me Number field. The Meet-Me Number is usually the network number the conferee dials to join a conference or the location of the conferee.

- *In Use*

  Indicates the status of endpoints and/or conferences. You can use this field to identify general connection or capability problems quickly. By checking other fields, you may identify specific capability or connection problems. This field is blank until the first call is made from or to the endpoint. The following list identifies other possible entries and their meaning:

  — blank — the call was never attempted.

  — **n** — the endpoint was connected but dropped or was dropped, or the endpoint attempted to connect but failed.

  — **y** — the endpoint is in use and is fully connected on all media. For a data call, at least MLP level ‘2’ connectivity was reached (and DLCI0 was established).

  — **c** — the endpoint is in use and is fully connected (as it is for **y**). However, the endpoint has changed the conference audio or video capability, or it has changed the rate of the conference because of rate adaptation. This condition requires a closer analysis of the endpoint’s capabilities and mode fields to determine the missing capability.

  — **e** — the endpoint is in use but it has a capability problem. Specifically, it does not have one of the required capabilities (channel, video, or data) to be a full participant This condition requires a closer analysis of the endpoint’s capabilities and mode fields to determine the missing capability.

  — **f** — the endpoint is in use but is not connected to all media. This indicates that the endpoint has declared all the required capabilities (channel, video, audio, or data) but is not fully connected to all conference media at this time. It could also indicate that the endpoint is in the process of connecting or an invalid video source (in which case the *Vs* flag would be set to **i**).

  — **t** — the endpoint has been moved to another conference. Refer to the appropriate Status of Conference Endpoint screen for details.

- *Capability*

  This field contains several subfields indicating the endpoint's current status and also its ability to receive video, audio, and data.

  — *Ch*

  Provides information on the quantity and quality of channels. The following list identifies other possible entries and their meaning:

  - **y** — the endpoint has the required number of channels.
  - **e** — the endpoint has not declared support for the correct number of channels and cannot participate fully in the conference.
  - **n** — the endpoint has declared the correct number of channels, but all the channels have not yet joined the conference due to a problem with either the network or an endpoint.

  This field is always blank for audio-only endpoints.

  — *Au*

  Provides information on the required audio capability. The following list identifies possible entries and their meaning:

  - **y** — the endpoint has the required audio capability. This field is always set to **y** once an audio-only endpoint joins the conference.

    c — Whenever the administered audio mode of a conference is set to **auto** and the ClearPresence@ 716 RTU is optioned, the system starts out with the highest common audio of ClearPresence @716. A G.711 or G.711/G.722-only endpoint that joins such a conference changes the conference video quality by changing its operating audio from CP716 to G.722. Any G.728 endpoints that join this conference interwork with ClearPresence @716 and do not change the conference audio.

    Whenever ClearPresence @716 is not optioned and the conference bandwidth is less than or equal to 128k, the system starts out with G.728. A G.711 or G.711/G.722-only endpoint that joins such a conference changes the conference video quality by changing its operating audio from G.728 to G.711.

    Whenever ClearPresence @716 is not optioned and the conference bandwidth is greater than 128k, the system starts out with G.722. A G.711-only endpoint that joins such a conference does not change the conference video quality because the conference audio in such a case remains at G.722 and the endpoint's G.711 audio interworks with G.722.

- **e** — either a G.711 (PCM) only endpoint was unable to support the administered audio mode of G.728 (LD-CELP) or G.722 (7kHz), or a G.728/G.711 endpoint could not support the administered audio mode of G.722 (7kHz). Such endpoints operate with G.711 (PCM) audio and interwork with the current operating audio mode.

This field is blank until the first call is made from or to the endpoint, or if the endpoint is a UCC endpoint.

— *Vid*

Two column field that provides information concerning video.

Values in the first column indicate if the associated endpoint has the required video capability and is receiving video. The following list identifies possible values and their meaning:

- **y** — the endpoint has the required video capability and should be receiving video if the *Ch, Au,* and *Dat* fields are each set to **y**.
- **c** — the endpoint downgraded the video quality of the conference. For example, H.263 could be downgraded to H.261. Also, within H.263 and H.261, the video format and the associated frame rates could be downgraded. For example, H.261 could be downgraded from CIF to QCIF. H.261 video mode is set by default to CIF and, if a QCIF-only endpoint joins the conference, the entire conference is set to operate in QCIF with a downgrade in video clarity. In addition, the conference frame rate is initially set to the highest frame rate, which can subsequently be reduced by an endpoint. Whenever the conference video mode is not administered with an upgrade capability, and if the video parameters for a conference are “downgraded,” they are not “upgraded” until all endpoints disconnect from the conference.
- **e** — the endpoint has not declared any video capability in its cap-set.
- **n** — the endpoint is an audio-only endpoint and is not receiving video (due possibly to an audio or data problem).

The first column is always blank for Audio Add-On endpoints.

Values in the second column indicate the standard video type. Values include blank (for a proprietary video), **1** (for H.261 video), and 3 (for H.263 video).

— *T120*

Provides information on MLP data capability, the state of the data connection in the T.120 stack, and the state of the Control Link to the ESM (T.120 stack terminator). The following list identifies possible entries and their meaning:

- **a —** the endpoint has the MLP data capability, but the endpoint's MLP channel was not joined to the data conference because the endpoint is administered as one that uses a non-compliant application.
- **e** — the endpoint does not have the MLP capability. This endpoint is joined to the conference as a secondary audio-only endpoint.
- **m** — the endpoint has the MLP data capability, but the MLP channel is not opened to the endpoint because the endpoint did not confirm the **MLP on** request from the MCU, or the endpoint is viewed as a "secondary" endpoint. (For example, an endpoint that supports CIF is joining a Continuous Presence *Plus* conference that does not support CIF.) As long as the MLP channel is not opened, this endpoint is joined to the conference as a secondary audio-only endpoint.
- **t** — the endpoint has confirmed the **MLP on** request, and the MLP channel is opened toward the endpoint. If the endpoint has no T.120 capability, or if it did not confirm the **T.120 on** message, the endpoint is joined to the conference as a audio/video endpoint.
- **0** — the endpoint has the MLP data capability and the MLP channel is opened to the endpoint, but the endpoint is not currently connected to the data conference due to a connection problem to the ESM.
- **1** — GCC initiated a Conference Invite Request to the endpoint, or GCC received Conference Join Request from the endpoint.
- **2** — the endpoint accepted a GCC Conference Invite Request. The endpoint has confirmed the Conference Invite/Join Request and, as such, is now connected to the Multipoint Communications System (MCS).
- **y** — the endpoint announced its entry into the conference. Accordingly, the endpoint's MLP channel has been successfully joined to a data conference. The endpoint can now send and receive data as a party in the multipoint conference.
- ***** — the control link to the ESM (MCS Provider) is down, and the real state of the connection to the MCS domain is not known.

A blank entry indicates either that the endpoint is a UCC or Audio Add-On endpoint or that the data mode for the conference is **none** (the latter case indicates either that the data does not apply or that the endpoint never joined the conference).

— *CC*

Indicates the state of the H.221 LSD channel and also the H.224 capabilities for each endpoint in the conference. The following list identifies possible values and their meaning:

- **t** — indicates an H.224-capable endpoint. The endpoint is an H.224 LSD token holder (current LSD broadcaster).
- **y** — indicates an H.224-capable endpoint that has sent an H.224-LSD-on mode command. However, it does not indicate whether or not this endpoint supports FECC.
- **d** — an H.221 LSD channel opened successfully, but an H.224-LSD-on command has not been received from the endpoint.
- **n** — LSD channel failed to open successfully in an FECC conference (that is, LSD-6400-on mode was never sent by the endpoint) even though the endpoint announced the LSD capability at 6.4k. This is also true for endpoints administered as “basic” on the Conference Record form. Therefore, the endpoint is joined to the conference as a secondary audio-only endpoint.
- **e** — Endpoint has not announced the 6.4k data capability in an FECC conference. Therefore, the endpoint is joined to the conference as a secondary audio-only endpoint.
- blank — FECC is not optioned for this conference; therefore, the data does not apply, the endpoint never joined the conference, or the endpoint is not a video-capable endpoint.

NOTE:
If FECC is administered for the conference, all the endpoints in the conference must open the 6.4k LSD channel for FECC to be fully functional. If an endpoint does not open the channel, it becomes an audio-only endpoint.

— *Gx*

Provides information on the Still-Frame Graphics capability of the endpoint. The following list identifies possible values and their meaning:

- **y** — the endpoint has this capability.
- **e** — the endpoint did not declare this capability. The conference retains the Still-Frame Graphics capability when a noncompliant endpoint joins the conference.

A blank entry indicates that the endpoint is either a UCC or Audio Add-On endpoint, or that the endpoint never joined the conference.

— *SM Rt*

Identifies endpoints with rates that are different from the primary conference rate. Values include **1**, **2**, and **3**. These values correspond to the respective rates entered into the *SM Rate 1 (2, 3)* fields.

— *Rt Ad*

Serves as a Rate Adaptation/interworking indicator. The following list identifies possible values and their meaning:

- **5** and **6** — pertain only to Low-/High-Speed Interworking. All other values pertain only to rate adaptation. The value **5** indicates that a low-speed (56k) endpoint joined a high-speed (128k or higher) conference. Such an endpoint is connected with audio-only capability and is not a valid video source and destination. The value **6** indicates that a low-speed (64k) endpoint joined a high-speed (128k or higher) conference. Such an endpoint is connected with audio-only capability and is not a valid video source and destination.
- **y** — the endpoint has joined the conference at the administered rate of 64k, but because another endpoint triggered rate adaptation to 56k, this endpoint has successfully rate-adapted down to 56k.
- **c** — the administered bandwidth of the conference is 64k and the endpoint joined the conference at 56k. The first 56k endpoint that joins a 64k rate adaptable conference triggers rate adaptation (see *Join Time* later in this list).
- **n —** assumes a 64k conference, and indicates that a rate adaptation to 56k was triggered by some other endpoint. This endpoint joined the conference at a 64k bandwidth but encountered problems in rate adaptating down to 56k. This endpoint may have audio capability and may be receiving video, but it is not a valid video source. Other conferees may hear, but not see, the conferee.

A blank entry indicates that rate adaptation was never triggered by any endpoint, and that if the endpoint is in use and connected, it has joined the conference at the administered bandwidth.

— *Bd Md*

Indicates the BONDing mode for BONDing calls. The entry **1** indicates BONDing Mode 1, and the entry **t** indicates transparent mode. This field is blank for any call not associated with BONDing.

— *Ts*

Provides information on the talking state of the endpoint. The following list identifies possible entries and their meaning:

- **t** — at the time of the request, voice energy was detected from the endpoint.
- **m** — at the time of the request, the endpoint informed the MCU that it (the endpoint) was muted.

  NOTE:
  There is no way for the MCU to find out if an endpoint is muted if the endpoint does not inform the MCU about this condition after it joins the conference. In such a case, the endpoint does not display a mute indication.

- **M** — can indicate either of the following:
  - — At the time of the request, the endpoint's audio was muted by a UCC/CRCS CRCS Agent interface.
  - — Both the endpoint and the UCC/CRCS CRCS Agent have muted the endpoint audio.
- **S** — at the time of the request, the endpoint's audio was muted due to the solo audio state set by the UCC/CRCS agent interface.
- blank — at the time of the request, voice energy was not being detected from the endpoint.

— *Vs*

Provides information on the MCU video state of the endpoint. For quad-screen conferences, an asterisk (*) is prepended to the value in the field to indicate that an endpoint is currently part of the mixed image. A pound sign (#) may be prepended to the field value to indicate that an endpoint was fixed to be in the mixed image (via administration or UCC/CRS Agent), but a "fill" video is shown in its place. This occurs whenever the video of an endpoint that is fixed in a quadrant cannot be used as a video source because the endpoint is currently not joined to the conference, has suppressed its video, or has invalid video to be the video source. A maximum of four endpoints can have * or # prepended to the field value.

For a quad-screen conference in VAS mode, the mixed image is broadcast to all the endpoints. For quad-screen conferences in Presentation mode, the mixed image is the return video to the Presenter.

The following list identifies possible values and their meaning:

- **a —** applies only to quad-screen conferences. Value ***a** indicates that the endpoint is part of the current mixed image and is fixed in one of the quadrants via administration. This number is prepended to the value in the *Vs* field. Value **#a** indicates that the endpoint is fixed in a quadrant but is not currently connected. (The fill image is being displayed.)

- **b** — for full-screen conferences, at the time of the request, the endpoint's video was being broadcasted to other sites, and the conference was in VAS, Broadcast, or Presentation mode.

  For a quad-screen VAS conference, the value is prepended with *****, and it indicates that the endpoint's video is part of the mixed image due to VAS. For a quad-screen Presentation conference, the value without the asterisk identifies the presenter as the broadcaster.

- **B** — at the time of the request, the endpoint's video was being broadcasted to other sites due to the UCC Rollcall option. This option can be performed only in full-screen mode.

- **c** — at the time of the request, the endpoint's video was broadcasted to other sites, the conference was in chair mode, and the broadcaster was designated by the chair. Chair features can be performed only in full-screen mode.

- **i** — at the time of the request, the endpoint was not a valid video source.

  If this endpoint is in a Continuous Presence *Plus* conference with fixed quadrant participants and is fixed in a specific quadrant, # is prepended to the value.

- **r** — for full-screen conferences, at the time of the request, the endpoint's video was the return video to the broadcaster.

  For Continuous Presence *Plus* conferences in Presentation mode, ***r** represents a VAS quadrant that is part of the mixed range.

- **R —** at the time of the request, the endpoint's video was the return video due to the UCC Browse option. Browse can be performed only in full-screen mode.

- **s** — at the time of the request, the endpoint's video was suppressed at the request of the endpoint.

  For Continuous Presence *Plus* conferences with fixed quadrant participants, if the endpoint is fixed in a specific quadrant, # is prepended to the value.

- **S** — at the time of the request, the endpoint's video was suppressed by the UCC/CRCS CRCS Agent interface. For Continuous Presence *Plus* conferences with fixed quadrant participants, if the endpoint is fixed in a specific quadrant, # is prepended to the value.

  This value is also displayed whenever both the endpoint and the UCC/CRCS CRCS Agent suppress the endpoint video.

- **u** — for full-screen conferences, at the time of the request, the endpoint's video was being broadcasted to other sites, the conference was in Voice Activated Switching (VAS) mode, and the endpoint was designated to be the broadcaster by the UCC/CRCS CRCS Agent interface.

  For quad-screen conferences, a UCC/CRCS Agent designated this endpoint as fixed in a quadrant. The value is prepended with ***** if the endpoint is currently part of the quad image, and # is prepended if the endpoint is not currently joined.

- **U** — pertains only to quad-screen conferences, and indicates that the UCC/CRCS Agent designated a quadrant as a VAS quadrant. A ***** is prepended to the value to indicate that this endpoint is part of the current quad image.

- **v** — at the time of the request, the endpoint's video was being broadcasted to other sites and the conference was in VAS mode, but the endpoint requested to be a broadcaster via a "See-Me" request and was granted a Multipoint Command Visualize (MCV) token. The See-Me features is performed only in full-screen mode.

A blank entry indicates that, at the time of the request, the endpoint's video was not broadcasted, used as return video, or part of the mixed image, but is a valid video source for endpoints currently joined to the conference.

- *Sum Grp*

  Identifies the VC audio Level 1 (L1) and Level 2 (L2) summer group ports for each assigned group (**1** through **4**). Summer ports are assigned only for conferences with more than six participants. Whenever a conference is operating at 7kHz audio mode (that is, the administered audio mode is either 7kHz or auto with a bandwidth higher than 128k), the system allocates "primary" and "secondary" L1 and L2 summer ports. These ports are allocated as adjacent port slots on the same circuit pack. Status conference displays only the "primary" summer ports. The "secondary" summer ports are always one slot higher than the displayed "primary" summer ports.

The remaining fields can contain up to two entries (one for each channel):

- *Join Time*

  Indicates (in 24-hour notation) when the channel joined the conference.

- *Drop Time*

  Indicates (in 24-hour notation) when the channel dropped from the conference. If the first channel has a drop time, the endpoint is not in use. Also, if there is a drop time, but no join time, the call was dropped without being joined to the conference.

- *Drop Reason*

  Indicates why the channel dropped. Refer to the “Conferee Dropped from Conference” section in this chapter for the list of drop reasons.

- *Ports Trunk*

  Identifies the trunk port that the channel is using.

- *Ports Video*

  Identifies the MMI port for the channel. This field is always blank for a UCC or Audio Add-On endpoint.

- *Ports Audio*

  If the endpoint type is not “UCC,” indicates the VC audio encoder port Indicates the VD audio encoder port (which is always paired to a decoder port) for the first channel. Because only one audio encoder port is allocated per endpoint, it appears together with the ESM data port in the endpoint’s channel 1 port slot position of the *Port...Aud/ESM* column. Ports 1 through 4 comprise encoder ports, while ports 5 through 8 comprise decoder ports.

  For endpoint type “UCC,” the channel 1 port slot position displays the allocated Call Classifier resource.

  For a UCC endpoint, this indicates a call classified resource.

- *Ports ESM*

  Indicates the Expansion Service Module (MMI) data port. This port is assigned only for channel 1. Because only one ESM MMI port is allocated per endpoint, it appears together with the audio port in the endpoint’s channel 2 port slot position of the *Port...Aud/ESM* column. This field is always blank for a UCC or Audio Add-On endpoint.

  NOTE:
  A missing ESM port for data calls indicates an ESM port resource problem. This condition can exist from the beginning of the call if the port is not available during the call setup, or it can occur during the active session if something happens to the allocated port. ESM port resources are not reallocated during an active call.

- *Sum Grp*

  Indicates the assigned summer group number for the endpoint (the summer group port assignments are shown on the first screen).

  This field is always blank for a UCC endpoint.

- *Software*

  Provides additional troubleshooting information. See the “Software Field Description” section later in this chapter.

- *AC Num*

  Indicates the Administered Connection Number (**1** through **128**). Administered Connection Numbers can be used to diagnose a problem more thoroughly by combining status conference information with the **status administered connection** command and data stored in the error and alarm logs. AC Numbers are only for dial-out calls; therefore, this field should be blank for dial-in calls.

The following fields on Page 7 are grouped with the *In Use*, *Vs*, and *Fr Err* fields, which also appear on page 1 of the form. For an explanation of these fields, refer to the appropriate bullet items earlier in this list.

- *Ept VCU*

  This is a circular hexadecimal counter that is used to indicate the number of endpoint and video processor Video Command Fast Update (VCU) requests from endpoints. Values include **0** through **FF**.

- *VP Dec Vcu*

  Video Processor Decoder VCU counter. This is a circular hex counter that is used to indicate the rate of the decoder's VCU requests. Values include **0** through **FF**.

- *VP Dec Ept*

  Endpoint that is currently associated with the video processor's decoder.

- *VP Dec Framing*

  Indicates whether or not the video processor's decoder has framing. Values include **y** and **n**.

Status of Conference Endpoint Screens

All data relevant to an endpoint is displayed in the Status of Conference Endpoint Screens. These screens display some conference level data and all endpoint-associated data. An example of page 1 of the screen set follows:

```
status conference 1 endpoint 3                                  page  1 of  6

              STATUS OF CONFERENCE: 1    ENDPOINT: 3_                Status: in-use__
 Ext: 72002                    Manufacturer/Country: 125_/1___       Product: 5___
    Terminal Name: MT-3c425-1_____        Data Mode: none_____      MLP Rate: ____
  Admin Bandwidth: 384k_             Conf Bandwidth: 384k_         Rate Adaptation? n
   Admin Ept Rate: 384k_              Endpoint Rate: 384k_       Lo/Hi Interworking? y
   Format(in/out): _________ FPS: ___     Meet-Me Number: (700)957-5957_____________
   Sum Grp: 1  L1: 1a0701     L2: 1a0702    Dial Out # 1: 917007654321_____________
                                                # 2: 917007654322_____________
  Conference Type: _____________ Destination Conf ID: ___        Destination Ept: __

                     ENDPOINT STATES/CAPABILITIES/MODES
                  In  Enh  -----Capability----- Rt Bd
                  Use BAS  Ch Au Vid T120 CC Gx Ad Md  Ts Vs
                   n   y   n  _  _ _  _   _  _  _  _   _  _

 OneNumber Access? assigned                 Channel Association Attempt: no

 Calling  #1: ____________________   #5: ____________________   # 9: ____________________
   Party  #2: ____________________   #6: ____________________   #10: ____________________
          #3: ____________________   #7: ____________________   #11: ____________________
          #4: ____________________   #8: ____________________   #12: ____________________
```

Figure 5-6. Sample Status of Conference Endpoint Screen (Page 1)

To view, on the MCU-ST, the Status of Conference Endpoint screen for an endpoint in a specific conference, log in and enter **status conference *conference-ID* endpoint *endpoint-ID***, where ***conference-ID*** is the number of the conference and ***endpoint-ID*** is the slot number in which the endpoint was administered in the Conference Record. To view the screens for all the endpoints in a conference, enter **status conference *conference-ID* endpoint *all*** or **status conference *conference-ID* endpoint *endpoint-ID*.** You can add **print** to any of the preceding command lines (for example, **status conference *conference-ID* endpoint *endpoint-ID* print**) to print a copy of the Status of Conference screen. Similarly, you can add schedule to any of these command lines to send the screen to the system printer at the specified time.

The first screen appears whenever you enter **status conference *conference-ID* endpoint *endpoint-ID*** and the specified conference ID is valid. This screen shows the first administered endpoint in the conference (the first administered endpoint may be in a slot other than the first slot). You can use the “Next Form” key to move to the next endpoint (if any) in the conference. Accordingly, the next screen shows the next administered endpoint; any empty slots are skipped. The command exits once the last endpoint is displayed. Also, you can exit the command by pressing “Cancel.” Note that the data displayed is whatever system

had stored at the time the command was entered. To see the latest data, enter the command again.

Basic States and Modes Screen—Page 1

The top of this screen (shown in the previous section) primarily reflects administration plans along with information relevant to format and summer groups. The middle of the screen contains information on endpoint level states, capabilities, and modes.

Most of the fields in the screen are the same as the fields in the Status of Conference screen (discussed earlier in this chapter). Therefore, the following list describes only those fields that are specific to the endpoint level command output.

- *Terminal Name*

  String of a maximum of 15 characters that represents the terminal name/identification parameters from the endpoint as defined in the ANSI version of the H.243 recommendation. Given that the MCU can poll and ascertain “naming/identification” parameters from endpoints, the contents of this field can be useful for site identification. For an Audio Add-On terminal, the term “audio-only” appears in this field.

- *Endpoint-ID*

  Identifies the slot number associated with the endpoint entered on the conference forms.

- *Sum Grp*

  Identifies the Summer Group number to which the endpoint belongs and also the VC audio Level 1 (L1) and Level 2 (L2) summer ports for this group.

  NOTE:
  These fields are populated only for conferences with more than five participants.

- *Product*

  Identifies the product identification number obtained from the endpoint.

- *Manufacturer/Country*

  Identifies the manufacturer identification number and the manufacturer’s country code obtained from the endpoint.

- *Admin Endpoint Rate*

  Indicates the SM rate at which the associated endpoint is administered. The field is blank if the endpoint is administered at the primary conference rate. Other values include **2x56k**, **2x64k**, **112k**, **128k**, **224k**, **256k**, **336k**, and **384k**.

- *Endpoint Rate*

  Indicates the actual rate at which the associated endpoint joined the conference. This rate can be different from the one indicated in *Admin Bandwidth* or *Admin Ept Rate*. If the rate is different from the rate identified in *Admin Ept Rate*, the endpoint has joined the conference as a secondary audio-only endpoint.

- *Destination Conf ID*

  Provides the ID of the conference to which the specified endpoint is currently related. The conference may be the administered conference or the conference to which the endpoint has been moved. Values include **1** through **200**.

  NOTE:
  If *In Use* contains **t**, the *Related Conf ID* field identifies the conference to which the endpoint has been moved. If *In Use* contains a value *other than **t***, the *Related Conf ID* field identifies the conference where the endpoint has been administered. Therefore, if the value for the *Related Conf ID* field and the value for the *Related Ept* field are the same as the value for the *Status of Conference* field and the value of the *Endpoint* field, respectively, the endpoint is being used in the conference in which it was administered.

- *Destination Ept*

  Provides the ID of the endpoint in the related conference. Values include **1** through **25**.

- *Enh BAS*

  Indicates the status of the Basic/Enhanced Service Flag. Entry **y** indicates that the endpoint supports the enhanced BAS commands/caps. Entry **n** indicates that the endpoint supports only the basic BAS commands/caps.

- *OneNumber Access*

  Indicates if the extension for the associated endpoint is a duplicated MCU-extension. Value **assigned** indicates that the MCU-extension assigned to the endpoint is also assigned to one or more other endpoints within the same conference. Value **no** indicates that the endpoint uses a unique MCU-extension. It does not indicate a failed attempt.

- *Channel Association Attempt*

  Because OneNumber Access allows the same MCU-extension to be administered for multiple endpoints in the same conference, there is a possibility of an incorrect association between the first and subsequent channels associated with a calling party. Value **yes** indicates that the subsequent channel association has been attempted for the associated endpoint. Value **no** indicates that the association attempt has not been made.

- *Calling Party Number [1 through 12]*

  Identifies the calling party numbers (CPNs) received from the network. CPNs may not always be available. Whenever a Billing Number is available, it may replace a CPN. Another name for the Billing Number is Automatic Number Identification (ANI).

Endpoint Channel Information Screen—Page 2

This screen contains channel data for up to 12 channels, which are required for BONDing calls. The number of channels used depends on the conference bandwidth. Non-BONDing calls use channels 1 and 2 or just channel 1.

```
status conference 1 endpoint 3                                  page 2 of 6

                                ENDPOINT CHANNEL INFORMATION

 Chan.  Join  Drop   ----Drop----    AC  ------------Ports-----------   Fr
 No.    Time  Time   Reason   Code   Num Trunk  Video Aud/ESM  BONDng   Err  Software
 1:     16:22 16:22 Busy_____ c3     1__ _______ _______ _______ _______  __  ________
 2:     _____ _____ __________ __    2__ _______ _______ _______ _______  __  ________
 3:     _____ _____ __________ __    3__ _______ _______ _______ _______  __  ________
 4:     _____ _____ __________ __    4__ _______ _______ _______ _______  __  ________
 5:     _____ _____ __________ __    5__ _______ _______ _______ _______  __  ________
 6:     _____ _____ __________ __    6__ _______ _______ _______ _______  __  ________
 7:     _____ _____ __________ __    ___ _______ _______ _______ _______  __  ________
 8:     _____ _____ __________ __    ___ _______ _______ _______ _______  __  ________
 9:     _____ _____ __________ __    ___ _______ _______ _______ _______  __  ________
10:     _____ _____ __________ __    ___ _______ _______ _______ _______  __  ________
11:     _____ _____ __________ __    ___ _______ _______ _______ _______  __  ________
12:     _____ _____ __________ __    ___ _______ _______ _______ _______  __  ________
```

Figure 5-7. Status of Conference Endpoint—Page 2

Most of the fields in the screen are the same as the fields in the Status of Conference screen (discussed earlier in this chapter). Therefore, the following list describes only those fields that are specific to the endpoint level command output.

- *Drop Code*

  Code that complements the drop reason and allows the VTC personnel to identify a more specific cause of the problem.

  NOTE:
  Drop codes are discussed later in this chapter.

- *Ports BONDng*

  Indicates the BONDing port for the channel.

- *Fr Err*

  Indicates, in hexadecimal notation (**1** to **FF**), the count of framing errors. An entry other than blank indicates a possible endpoint problem.

Communication Modes, Frame Alignment, and Miscellaneous Information Screen—Page 3

Page 3 provides information on the following:

- Frame alignment word (FAW)
- Conference communication modes
- Endpoint incoming and outgoing command modes
- Miscellaneous endpoint states
- Conference broadcaster/return video

All Page 3 values are zero (**0**, **0x00**, **n**, or **neutral**) if call is not active or the endpoint is an Audio Add-On endpoint (endpoint Type is **BAUD** on Page 1 of the forms.) Some values are zero during initial handshaking.

```
status conference endpoint                              page  3 of  6


CONFERENCE INFO: Broadcaster: xx (See-Me)       Next Broadcaster: xx
                 Return Vid: xx                 Next  Return Vid: xx

---------- MODE COMMANDS/COMMUNICATION MODES ------|   EPT MISC  | ------- FAW --
 CMD  STAT    CONF       EPT => MCU    MCU => EPT  |------ I  0 |       CH1 CH2
----- --- ------------ ------------  ------------  |  AIM: y  n |       --- ---
XRATE: y  384          384           384           |  VIS: y  n | A-OUT:  y   n
AUDIO: n  G728         neutral       G728          |  MIS:    n |  A-IN:  y   n
56/64: y  derestrict   derestrict    derestrict    |  MCV: n    | M-FRM:  y   n
VIDEO: n  H.261        H.261         H.261         |------------|   MFA:  y   n
  MLP: y  MLP-off      MLP-off       MLP-off       | TALK: y 05 |   MFN:  y   n
H_MLP: y  H-MLP_off    H-MLP_off     H-MLP-off     | VRCV: y 01 |   FAS:  1   0
  LSD: y  LSD-6400     LSD-6400      LSD-6400      | BCTK: n 02 |MCUFAL: 00  00
  HSD: y  HSD-off      HSD-off       HSD-off       | RTTK: y 01 | FEFAL: 00  00
CRYPT: y  encryp-off   encryp-off    encryp-off    | BCLS: n    |
  S/M: y  N-comp-6B-HO N-comp-6B-HO  N-comp-6B-HO  | RTLS: n    | --- DELAY ---
                                                   | HYPR: n 00 | AUDIO: 420
                                                   |DMUTE: n 00 |
                                                   |H.261: n 00 |
                                                   | VFMT: n 00 |
```

Figure 5-8. Status Conference Endpoint Screen—Page 3

Frame Alignment Word (FAW) Area

The FAW area provides the channel Frame Alignment Word information for the communication paths labeled CHL *1* and *2*. For 2B calls, both CHL *1* and *2* are used. For 1-channel calls (1x56k, 1x64k, 112k, 128k, 168k, 192k, 224k, 256k, 280k, 320k, 336k, 384k, 448k, 512k, 672k, 768k, 1008k, 1152k, 1288k, 1344k, 1472k, 1536k, 1680k, and 1920k), only CHL *1* is used. *A-OUT, A-IN, M-FRM, MFA* and *FN* are flags with values of **y** or **n**. The following list discusses these and other flags.

- *A-OUT*

  Indicates if the MCU has endpoint framing.

- *A-IN*

  Indicates if the endpoint has MCU framing.

- *M-FRM*

  Indicates if a multichannel frame alignment (alignment of both channels in 2B) was achieved.

- *MFA*

  Indicates if a multiframe alignment word (required in 2B call) was found.

- *MFN*

  Indicates if multiframe numbering (required in a 2B call) is present.

- *FAS*

  Frame Alignment Signal (FAS) channel number (1 or 2; should match column header).

- *MCUFAL*

  MCU Frame Alignment Loss (MCUFAL). This is a counter of the number of times the MCU indicates to the endpoint that it has lost endpoint FAW or multichannel synchronization (M-FRM). The MCU a-bit toggles when the MCU gains or loses endpoint multichannel synchronization (M-FRM). This counter (MCUFAL) starts with initial value of 0x00, it increments to 0xff and then wraps around to 0x01. The MCU FAL count is also shown in the *Fr Err* field of Page 1 output.

- *FEFAL*

  Far-End Frame Alignment Loss (FEFAL). This is a counter of the number of changes the MCU detects in the endpoint's a-bit (A-OUT). The endpoint a-bit toggles when an endpoint gains or loses MCU framing. This counter (FEFAL) starts with an initial value of 0x00. It increments to 0xff and then wraps around to 0x01.

Mode Commands and Communication Modes Area

The modes of a conference are the Selected Communication Modes. These modes are a collection of bandwidth allocations for the multiplex. The *MODES COMMANDS* include incoming and outgoing modes. “Incoming” pertains to what the MCU thinks the endpoint is doing based on the Bit-Rate Allocation Signal (BAS) codes the MCU receives from the endpoint or codec and the capabilities the MCU has declared. “Outgoing” pertains to what is going from the MCU toward the endpoint.

The *CMD* column contains labels for the various types of mode commands. The *STAT* column indicates if the conference and the incoming modes are compatible. A value of **y** indicates mode compatibility. A value of **n** indicates that modes are not compatible. A value of **n** in the *STAT* column for any command impacts video status.

NOTE:
The *56/64* field is an exception to the previous statements. See the note in the *56/64* description in the following list.

The *CONF* column defines the communication modes of the conference. The *EPT => MCU* column defines the communication modes coming in from an endpoint. The *MCU => EPT* column defines the communication modes sent out to an endpoint based on the number of channels connected and the capabilities of the endpoint.

For Speed Match conferences, there can be a maximum of four conference communication modes; that is, one mode per conference. The values in the *STAT* column corresponding to the XRATE, AUDIO, and 56/64 communication modes in the *CMD* column indicate the status of information associated with the Speed Match rate for the associated endpoint. This rate can be the primary conference rate. The values in the *CONF* column for the XRATE, AUDIO, and 56/64 communication modes provide information associated with the endpoint’s rate. This rate can be the primary conference rate or one of the additional Speed Match rates.

The following list identifies and discusses the Command/Communication modes:

- *XRATE*

  Transfer rate. The XRATE should be one of the supported rates as defined in the following table.

Table 5-1. XRATES and Corresponding Bandwidth(s)

XRATE	Bandwidth of the Call
1x64	1B (1x56k or 1x64k)
2x64	2B (2x56 or 2x64)
128	112k or 128k
196	168k or 196k
256	224k or 256k
320	280k or 320k
384	336k or 384k
512	448k or 512k
768	672k or 768k
1152	1008k or 1152k
1472	1288k or 1472k
1536	1344k or 1536k
1920	1680k or 1920k

XRATE may be **64** when the endpoint is just dialing in or if problems occur. This value implies that only one B-channel is being used.

With Speed Match enabled, this field indicates the transfer rate for the associated endpoint. This rate may differ from the primary rate administered for the conference.

- *AUDIO*

  Indicates the actual audio delay for the associated endpoint. (This information is retrieved from call processing.) Value 0 appears for full-screen conferences, Audio Add-On endpoints, UCC endpoints, and endpoints that are not joined to the conference. The displayed value is not retained once the endpoint drops from the conference.

  Audio rate (number of kilobits per second of bandwidth) of the conference and the endpoint must be the same, but the conference and the endpoint need not necessarily have the same mode (for example, G.711-A-48k/G.711-Mu-48k/G.722-48k and G.711-A-56k/G.711-Mu-56k/G.722-56k have the same rate but different modes.) When the audio rate of the conference and the endpoint are different, the endpoint's audio interworks, but the endpoint's video is invalid. The MCU may or may not send video to such an endpoint.

  The following table summarizes the expected endpoint audio mode with different configurations of transfer rate, facility bandwidth, and conference mode. The table also indicates whether or not all endpoints support the highest common audio mode (HC Audio) when the administered mode is y.**auto**. The highest common conference mode is dependent on the administered bandwidth. A dash (-) indicates that a value for the corresponding field does not apply.

Table 5-2. Summary of Expected Endpoint Audio Modes

Transfer Rate	CP716 RTU?	Administered Audio Mode	Facility BW	Conference Audio Mode	All Support Conf Mode?	Expected Endpoint Mode for Video
<=128k	no	auto	___	G.728	yes	G.728
			64k	G.711-A-56k	no	G.711-A-56k G.711-Mu-56k
			56k	G.711-A-48k		G.711-A-48k G.711-Mu-48k
	yes		___	CP716	yes	CP716, G.728
			64k	G.722-56k	no	G.722-56k G.711-A-56k G.711-Mu-56k
			56k	G.722-48k		G.722-48k G.711-A-48k G.711-Mu-48k
>128k	no		64k	G.722-56k	___	G.722-56k G.711-A-56k G.711-Mu-56k
			56k	G.722-48k		G.722-48k G.711-A-48k G.711-Mu-48k
	yes		___	CP716	yes	CP716
			64k	G.722-56k	no	G.722-56k G.711-A-56k G.711-Mu-56k
			56k	G.722-48k		G.722-48k G.711-A-48k G.711-Mu-48k
___	___	G.728	___	G.728	___	G.728
		G.711		G.711-A-56k		G.711-A-56k G.711-Mu-56k
				G.711-A-48k		G.711-A-48k G.711-Mu-48k
		G.722		G.722-56k		G.722-56k G.711-A-56k G.711-Mu-56k
				G.722-48k		G.722-48k G.711-A-48k G.711-Mu-48k

Other possible mode values for this field include **neutral** (neutralized I-channel) plus **Au-off** and **Frm** (no audio signal), which never match the conference mode.

NOTE:
Conferences using Speed Mach allow audio interworking between different facility bandwidths.

- *56/64*

  This field contains **derestrict** when operating at per-channel speeds of 1x64k, 2x64k, 128k, 192k, 256k, 320k, 384k, 512k, 768k, 1152k, 1472k, 1536k, or 1920k; it contains **restrict** when operating at speeds of 1x56k, 2x56k, 112k, 168k, 224k, 280k, 336k, 448k, 672k, 1008k, 1288k, 1344k, and 1680k.

  NOTE:
  If the conference is configured for Nx56k operation, the endpoint may signal either via capabilities or modes that are operating at the proper rate. Accordingly, if the **derestrict** value appears and therefore does not match the conference communication mode of **restrict** (with a STAT 56/64 of **n**), but the capability indicates **restrict** (MISC capability has **restrict** displayed on Page 3), the MCU assumes that the 56/64 compatibility exists for the conference and endpoint.

- *VIDEO*

  The value **H.263** (Recommendation H.263) **H.261** (Recommendation H.261), **H.CTX** (proprietary), **H.CTX+** (proprietary), or **SG4** (proprietary) indicates that the video is on in the direction indicated. The value **video-off** indicates that the video is off.

- *MLP*

  Multilayer Protocol data mode. When *Data Mode* is administered as **any_mlp**, the *MLP* mode should be consistent with the administered MLP Rate**.** These values include **var-MLP**, **MLP-6.4k**, **MLP-8k**, **MLP-14.4k**, **MLP-16k**, **MLP-22.4k**, **MLP-24k**, **MLP-30.4k**, **MLP-32k**, **MLP-38.4k**, **MLP-40k**, and **MLP-46.k**. Other values will affect video status.

  The *MLP* mode should be **MLP-off** when *Data Mode* is administered as **none**. Again, other values in this mode will affect video status.

- *H_MLP*

  The High-Speed MLP mode. This mode should be **H-MLP-off**. Other values in this mode will affect video status.

- *LSD*

  Low-Speed Data mode. This mode should be **LSD-off**. Other values in this mode will affect video status.

- *HSD*

  High-Speed Data mode. This mode should be **HSD-off**. Other values in this mode will affect video status.

- *CRYPT*

  Encryption mode. This mode should be **encrypt-off**. Other values in this mode will affect video status.

- *S/M*

  Single-/Multi-channel interoperability mode. The value **6B-H0-comp** indicates that the sender is interoperating multiple channels and a single channel (for example, a 6B channel and an H0 channel). The value **Not-comp-6B-H0** indicates that the sender is not interoperating between 6B and H0. Usually, this value is **Not-comp-6B-H0**. Other values in this mode will affect video status.

Endpoint Miscellaneous Information (EPT MISC) Area

The EPT MISC area contains miscellaneous states and counters for an endpoint. The flags can have a value of **y** or **n**. The counters start with initial value of 0x00, increment to 0xff and then wrap around to 0x01. **AIM** and **VIS** are BAS commands that can be sent as input (I) to the MCU from an endpoint or as output (O) from the MCU to an endpoint.

- **AIM**

  Audio Indicate Muted. Value **y** on input (I) indicates that this endpoint has muted its audio. MCU will not VAS to an endpoint displaying a mute indication. Value **n** on input indicates that this endpoint has not muted (this is possible only if the endpoint audio mode is turned on.) Value **y** on output (O) indicates that all other endpoints in the conference have muted their audio (have sent **AIM** to the MCU). By sending **AIM** to the endpoint, the MCU in turn tells the endpoint that there is no audio output from the MCU. Value **n** on output indicates that there is an audio path open across the bridge.

- **VIS**

  Video Indicate Suppressed. Value **y** on input (I) indicates that this endpoint has suppressed its video (indicated video is muted). Value **y** on output (O) indicates that the MCU is not sending video to this endpoint because there is no video broadcaster (that is, the broadcaster has not joined, or the broadcaster's video is not valid).

- **MIS**

  Multipoint Indicate Secondary Status. This command is sent only as output (O) to an endpoint. Value **n** indicates that the endpoint is viewed as capable of being a valid source (although not necessarily at this moment).Value **n** is the correct state to be in for video. Value **y** indicates

that MIS was sent to an endpoint and that this endpoint is viewed as a secondary endpoint and is therefore included in the audio portion of the conference but not in the video portion. Video will not be sent.

- **MCV**

  Multipoint Command Visualize. This command is sent only as input (I) from an endpoint. Value **y** indicates that an endpoint has requested to become a broadcaster. (This is used during Still-Frame Graphics and to force presentation mode.)

- **TALK**

  Value **y** in the TALK flag indicates that the VC board is detecting voice energy from the endpoint. The TALK counter indicates the number of times a start/stop was detected.

- **VRCV**

  Value **y** in the VRCV flag indicates if the endpoint is receiving video (MCU is sending video to the endpoint). The VRCV counter indicates the number of times video was sent or not sent to this endpoint.

- **BCTK**

  Value **y** in the BCTK flag indicates that the endpoint is the video broadcast source. The BCTK counter indicates the number of times this endpoint was the video broadcast source.

- **RTTK**

  Value **y** in the RTTK flag indicates that the endpoint is the return video source. The RTTK counter indicates the number of times this endpoint was the return video source.

- **BCLS**

  Value **y** indicates that the endpoint is watching the video of the broadcast source.

- **RTLS**

  Value **y** indicates that the endpoint is watching the video of the return source.

- **HYPR**

  Value **y** in the HYPR flag indicates hyperactivity from an endpoint (the MCU isolated an endpoint from the MCU conference due to a “thrashing” behavior) and that this hyperactivity affects the endpoint’s status as a video source (for five seconds of the hyperactivity timer). The HYPR counter indicates the number of times this endpoint was hyperactive.

- **DMUTE**

  Value **y** in the DMUTE flag indicates that the decoder was muted by the VC circuit pack or by the software in the MCU. The VC circuit pack mutes the decoder when it loses MMI, endpoint framing is lost, it receives an invalid audio code word, or when the endpoint is hyperactive. The only time that

the MCU software mutes the decoder of an endpoint is when it mutes all endpoints, other than the broadcaster, in a conference with broadcast with auto scan mode enabled. The DMUTE counter indicates the number of times this endpoint's decoder was muted by the VC circuit pack.

- **VFMT**

  Applies only to quad-screen conferences. Value **y** indicates that the video format is violated: the Video Processor circuit pack detected CIF instead of QCIF in the quad-screen conference. The VFMT counter indicates the number of times CIF was detected.

- **H.261**

  Value **y** indicates video framing loss. The H.261 counter indicates the number of times the framing was lost.

Conference Information Area

This area contains the following fields and corresponding values

- *Broadcaster*

  Indicates the endpoint number that is the current broadcaster. The following list identifies the associated keywords along with their meaning:

  — **Broadcast**

  Broadcast mode broadcaster.

  — **Chair**

  Broadcaster was designated by the chair.

  — **See-Me**

  Broadcaster is a result of MCV request from an endpoint.

  — **Presenter**

  Presentation mode broadcaster.

  — **Rollcall**

  Broadcaster was designated by the UCC via the Rollcall feature.

  — **UCC**

  Broadcaster was designated by the UCC.

  — **VAS**

  Voice-Activated Switching broadcaster.

- *Next Broadcaster*

  Indicates the endpoint number that is selected to be the next broadcaster.

- *Return Vid*

  Indicates the endpoint number that is the current return video. The following list identifies the associated keywords along with their meaning:

— **blank**

Return video is the previous broadcaster forced to be return video because of VAS, action by the Chair, action by UCC, or endpoint MCV request. The broadcaster qualifier identifies which action forced this endpoint to become return video.

— **Autoscan**

Auto-scan return video. This is true only when the conference mode is broadcast with auto-scan.

— **Browse**

Return video was designated by the UCC via the Browse feature.

— **VAS**

Voice-Activated Switching return video.

- *Next Return Vid*

  Indicates the endpoint number that is selected to be the next return video.

Endpoint Capabilities Screen—Page 4

The capabilities of an endpoint indicate what audio modes, video rates, and transfer rates the endpoint supports (can receive) on the current connection. All Page 4 values are blank when the call is not active or when the endpoint is either an Audio Add-On endpoint (endpoint *Type* is **AUD** on Page 1 of the form) or a UCC endpoint (endpoint *Type* is **UCC** on Page 1 of the form). Some values are blank during initial handshaking.

```
status conference 1 endpoint 3                          page  4 of 6

                        ENDPOINT CAPABILITY INFORMATION
------+--H.261 --+---------- H.263 -------------------------+----------------------
      | CIF:7.5   |  16CIF:  5 cmp umv amp ac pb HRD:4 BPP:4 | H.CTX       MRV:30fps
 VID  | QCIF: 30  |   4CIF:7.5 cmp umv amp ac pb HRD:4 BPP:4 | H.CTX+      da_sfg
      |           |    CIF: 15 cmp umv amp ac pb HRD:4 BPP:4 | SG4         vic
      |           | S/QCIF: 30 cmp umv amp ac pb HRD:4 BPP:4 | SG4_sfg     vim
------+-----------+-------------------------------------------+----------------------
MISC  | derestrict   S/M  mbe   dcomp     cic    encrypt  esccr   tic
------+-------------------------------------------------------------------------
  XR  | 64 64x2 64x3 64x4 64x5  64x6      384    384x2    384x3   384x4     384x5
      |    128  192  256  320   512  768  1152   1472     1536    1920
------+-------------------------------------------------------------------------
 AUD  | ntrl    711m 711a g728  g722_64   g722_48      au_iso
------+-------------------------------------------------------------------------
 LSD  | var 300 1.2k 4.8k 6.4K  8k   9.6k 14.4k  16k  24k 32k 40k 48k 56k 62.4k h224
------+-------------------------------------------------------------------| h224s
 HSD  | var 64k 128k 192k 256k  320k 384k 512k   768k  1152k  1536k h224
------+-------------------------------------------------------------------------
 MLP  | var 6.4k 8k 14.4k  16k   22.4k 24k  30.4k  32k  38.4k 40k 46.4k t120 h224
------+-------------------------------------------------------------------------
HMLP  | var 62.4k    68k  128k  192k 256k 320k   384k
```

Figure 5-9. Status Conference Endpoint Screen—Page 4

Fields on this page are displayed only if an endpoint declares the specific capability. For example, if an endpoint does not declare the VID CTX capability, the *CTX* field is not displayed.

Video Capabilities

The *VID* (video) fields in the *CAPABILITY* information provide statements about the type of video and frame rate the endpoint supports.

NOTE:
The format capabilities fields are not displayed if endpoint has no video capability.

An endpoint can have both H.261 and H.263 capabilities. In general, for larger screens, higher resolutions such as CIF display a sharper resolution video, while lower resolutions such as QCIF are not as sharp (but may run at higher frame rates). With very small displays, the differences between the resolutions are less observable.

Associated with each format is a frame rate; this indicates the minimum picture interval or the maximum frame rate (in frames per second or fps) at which the endpoint can receive video. A faster frame rate provides better quality motion in video. H.263 frame rate values include 30fps, 15fps, 10fps, 7.5fps, 6fps, 5fps, 3fps, 2fps, and 1fps. H.261 frame rate values include 30fps, 15fps, 10fps, and

7.5fps Therefore, for example, *CIF*: **30fps** indicates that the endpoint can receive one frame each 30th of a second with full CIF format.

NOTE:
The endpoint does not indicate to the MCU the transmit frame rate the endpoint is capable of or currently encoding.

Associated with the H.263 format are the following options:

- **CMP** — reserved for future use
- **UMV** — Unrestricted Vector Mode
- **AMP** — capable of using Advanced Motion Prediction
- **AC** — capable of using Arithmetic Coding
- **PB** — capable of using PB Frames
- **HRD** — using HRD-B scaling factor for a higher resolution format. Contains associated HRD-B value.
- **BPP** — using BPPmaxKB scaling factor for a higher resolution format. Contains associated BPPmaxKB value.

If the endpoint does not support a specific format, all values associated with that format (frame rate and options) are blank.

The following list identifies and comments on other values for the *Vid* fields.

- **da_sfg**

  Indicates support for H.261 Still-Frame Graphics transfers.

- **imp**, **iso**, and **aviso**

  These values are reserved for future use and will not usually appear.

- **H.CTX, H.CTX+**, **SG4**, and **MRV**

  Proprietary video format capabilities. Value **MRV** includes the associated frame rate.

- **SG4_sfg**

  Indicates support for SG4 Still-Frame Graphics.

- **vic**

  Video Indicate Compose. This may be declared by a cascaded MCU.

  NOTE:
  This is currently not supported by the Lucent Technologies MCU.

- **vim**

  Video Indicate Mixing. This may be declared by an endpoint.

  NOTE:
  This is currently not supported by the Lucent Technologies MCU.

Transfer Rate Capabilities

The *XR* (transfer rate) fields of the CAPABILITY information are statements about the speeds at which the endpoint can operate over the current connection and how it can operate a H.320 multiplex.

For a 384k (H0) call, the endpoint sets its capabilities to indicate 384k support; therefore, *XR* **384** is displayed. On a 336k call, the endpoint must signal 384k support. If an endpoint does not indicate support for 384k on a 384k/336k call, the MCU provides Audio-Only Communications Mode (AOCM).

For a 2B conference, the MCU sets a 2x64k rate. It expects endpoints to do likewise. If this capability is present, *XR* **64x2** is displayed. (If this value is not displayed, there is no 2x64k capability.)

Endpoints may occasionally take 2x64 (or the current channel rate 384k, 768k, 1472k, 1536k) out of their capability set. This is part of a customary procedure called Mode 0 forcing. The MCU provides AOCM if the endpoint does not signal support that matches the configuration of the conference (that is, the MCU's capabilities for that conference).

Audio Capabilities

The *AUD* (audio) fields are statements of the audio protocols that the endpoint supports. The following list identifies and discusses the associated values.

- **711m** and **711a**

  Indicate PCM (G.711) support, Mu and A-law, respectively. At least one value is required of endpoints.

- **728**

  Indicates whether G.728 is supported (LD_CELP). This value depends on the type of the endpoint and how that endpoint is currently configured.

- **722_48**

  Indicates endpoint support for G.722 (7kHz) at both 48kbps and 56kbps. If this value is displayed, the endpoint supports G.722 audio at both rates.

- **722_64**

  Indicates endpoint support for G.722 at 64k in an unframed mode (not supported by the MCU).

- **au_iso**

  Reserved for future use (will not be displayed).

- **cp716**

  Indicates support for the ClearPresence @716 audio mode.

Miscellaneous Capabilities (MISC)

A **restrict** value is one way for an endpoint to indicate that it is operating at 56k per channel. Another way is the **56/64** command mode (see page 2) with the value **restrict**. An endpoint on a 56k conference must send one of the two indications (or both) that it is operating at 56k before it can become a video source in a 56k conference. If it signals either way that it is operating at 56k in a 64k (or 384k) conference, it is an audio-only source. However, the MCU continues to send Selected Communication Mode (SCM) toward it whenever possible.

A MISC capability of **derestrict** and a **56/64** command with **derestrict** (see page 2) together indicate that an endpoint is operating at 64k. If either the capability or the command has the value **restrict,** 56k operation is signaled.

Other values are identified and discussed in the following list.

- **dcomp**

  Data compliance indicator. This is reserved for future use and should not appear.

- **mbe**

  Indicates support for Multi-Byte Extension (MBE). MBE capability is used for the exchange of passwords, terminal names, and other special capabilities

- **cic**

  Chair Indicate Capability. Indicates chair control capability.

- **S/M**

  Indicates capability of Single-/Multi-Channel interoperability. This is reserved for future use and should not appear.

- **encrypt**

  Indicates encryption capability. This is reserved for future use and should not appear.

- **esccr**

  This is reserved for future use and should not appear.

- **tic**

  Terminal Indicate Capability. Indicates the Terminal Indicate Exchange (TIX) commands capability. This is reserved for future use and should not appear.

Low-Speed Data Capabilities

The *LSD* fields indicate the capabilities for Low-Speed Data conferencing and for H.224 over the LSD channel. Value **h224s** indicates that H.224 in the LSD channel can function simultaneously with T.120 protocols in the MLP channel.

High-Speed Data Capabilities

The *HLP* fields indicate the capabilities for High-Speed Data conferencing and for H.224 over the HSD channel.

Multilayer Protocol Data Capabilities

The *MLP* fields indicate the capabilities for MLP data conferencing, including support for the T.120 stack and H.224 capabilities over the MLP channel. The 4k, 62.4k, and 64k rates are not displayed.

High-Speed Multilayer Protocol Data Capabilities

The *HMLP* fields indicate the conference's capability for High-Speed MLP data conferencing.

Endpoint Call Status Information Screen—Page 5

This page summarizes call-related status. The page contains fields for per-channel join counts, join/drop times, drop reasons, drop codes, and auxiliary bonding drop codes. It also contains fields for drop codes and software from the previous call.

```
status conference 1 endpoint 3                          page  5 of 6

                        ENDPOINT CALL STATUS INFORMATION
     Join   Join  Drop      -------------- Drop --------  --- Previous -----
Chan Count  Time  Time      Reason       Code    BondCode  DropCode  Software
 1:   8_    16:22 16:22     Busy_______   c3       __        c3       ________
 2:   1_    ______ ______   __________   __       __        __       ________
 3:   1_    ______ ______   __________   __       __        __       ________
 4:   1_    ______ ______   __________   __       __        __       ________
 5:   1_    ______ ______   __________   __       __        __       ________
 6:   1_    ______ ______   __________   __       __        __       ________
 7:   __    ______ ______   __________   __       __        __       ________
 8:   __    ______ ______   __________   __       __        __       ________
 9:   __    ______ ______   __________   __       __        __       ________
10:   __    ______ ______   __________   __       __        __       ________
11:   __    ______ ______   __________   __       __        __       ________
12:   __    ______ ______   __________   __       __        __       ________
```

Figure 5-10. Status Conference Endpoint Screen—Page 5

This screen is always retained, and it groups together all call-related fields. All the fields in this section appear on Page 1 of the form except for the *Join Count* field. This field displays a count of the number of times the endpoint joined the associated conference during a specific conference session. This counter starts with initial value of 0, increments to 64, and then wraps around back to 1.

The *Previous DropCode* and *Previous Software* fields reflect the values of the *Drop Code* and the *Software* fields (see Page 1 of the form) during a previous call or call attempt.

Administered Connections Information Screen—Page 6

This screen contains information on the administered connections for the associated endpoint. The screen can be viewed only while the conference is active. The screen appears as follows:

```
status conference 1 endpoint 3                          page  6 of 6

                    ADMINISTERED CONNECTIONS INFORMATION

     AC                                                    Retry   Failure
Chan Num    Dial-out Number           Connection State     Count   Cause
 1:  1__    40010_____________________ waiting to retry__   5_      11
 2:  2__    40010_____________________ disabled___________  0_      00
 3:  3__    40010_____________________ disabled___________  0_      00
 4:  4__    40010_____________________ disabled___________  0_      00
 5:  5__    40010_____________________ disabled___________  0_      00
 6:  6__    40010_____________________ disabled___________  0_      00
 7:  ___    __________________________ ____________________  __      __
 8:  ___    __________________________ ____________________  __      __
 9:  ___    __________________________ ____________________  __      __
10:  ___    __________________________ ____________________  __      __
11:  ___    __________________________ ____________________  __      __
12:  ___    __________________________ ____________________  __      __
```

Figure 5-11. Status Conference Endpoint Screen—Page 6

The following list identifies and discusses fields for the screen.

- *Dial-Out Number*

  Contains actual numbers that are dialed out by the administered connections for each channel in the call.

  NOTE:
  Dial Out #1 and *#2* on Page 1 of the form display only the administered numbers. This is particularly useful with BONDing dial-out calls.

- *Connection State*

  Indicates the current call state of the AC. The following are connection states associated with dial-out ACs:

  — **enabled**

  Transient in nature, indicates that an AC is about to enter the “attempting to connect” state.

— **disabled**

Indicates one of the following:

- AC has reached an administered retry threshold and all retries have been stopped. This can be verified by checking the Hardware Error Log and checking whether an error type of 9 is logged against the AC.
- AC was in a “connected” state and the far end has disconnected.
- Initial channel call has not yet been connected. No dial-out call attempt is made for the additional channel(s) until the initial channel has reached a “connected” state.

— **not scheduled**

Transient in nature, indicates that an AC is about to enter the “attempting to connect” state.

— **waiting to retry**

Indicates that the AC is inactive (sleeping) waiting for the retry timer to expire. Once the timer expires, the AC will launch a dial-out call and enter the “attempting to connect” state. ACs in this state also indicate that the dial-out call has failed at least once.

— **attempting to connect**

Indicates that the AC is active on a call, but the call has yet to be connected.

— **connected**

Indicates that the call associated with the AC has been answered and joined cut-thru to the conference.

- *Retry Count*

Indicates how many retries have been attempted for this AC during this join attempt. This field is not cleared when the AC is connected. However, it is cleared when a new join attempt is made via a Redial feature. Note that this is different from “Join Count,” which counts the number of times the channel joined the conference during this conference session.

- *Failure Cause*

Shows an ISDN or CPTR cause value (values lower than 0x7f) recorded when the last dial-out call was dropped. Values above 0x7f are generated internally. The following table lists all possible failure cause values displayed by this field and its associated description. The *Failure Cause* value is logged in the Hardware Error Log and can be displayed via the **dispay errors** command. (*Err Type* is normally displayed in decimal.)

Table 5-3. Values for Status AC Failure Cause Field

Failure Cause Hex (decimal)	Description
0x00 (0t0)	N/A
0x01 (0t1)	Incorrect destination address
0x02 (0t2)	Reason unknown
0x06 (0t6)	Reason unknown
0x10 (0t16)	Normal call clearing
0x11 (0t17)	Endpoint not available
0x12 (0t18)	ISDN timer expired
0x15 (0t21)	Reason unknown
0x12 (0t22)	Destination address changed
0x1C (0t28)	Bad destination or access denied
0x1D (0t29)	Access denied
0x1F (0t31)	Reason unknown
0x22 (0t34)	Trunks unavailable
0x26 (0t38)	Temporary or facility failure
0x29 (0t41)	Temporary or facility failure
0x2A (0t42)	Resources unavailable
0x2C (0t44)	Resources unavailable
0x32 (0t50)	Access denied
0x34 (0t52)	Access denied
0x36 (0t54)	Access denied
0x3A (0t58)	Resources unavailable
0x41 (0t65)	Required capability not implemented
0x42 (0t66)	Required capability not implemented
0x45 (0t69)	Required capability not implemented
0x51 (0t81)	ISDN protocol error
0x52 (0t82)	Required capability not implemented
0x58 (0t88)	Incorrect destination number
0x60 (0t96)	ISDN protocol error
0x61 (0t97)	ISDN protocol error

Table 5-3. Values for Status AC Failure Cause Field — *Continued*

Failure Cause Hex (decimal)	Description
0x62 (0t98)	ISDN protocol error
0x64 (0t100)	ISDN protocol error
0x66 (0t102)	ISDN timer expired
0x7f (0t127)	Reason unknown
0xC2 (0t194)	Ring no answer
0xC8 (0t200)	High and dry: no feedback detected
0xC9 (0t201)	Cascade link administered incorrectly
0xCA (0t202)	CPTR not available to detect failure

Debugging Hints

The additional information provided on the “Endpoint Capability Information” page (page 4) of the form is useful when attempting to understand why an endpoint may not be receiving video information or why the MCU may not be using an endpoint's video information, as well as other media troubles.

NOTE:
This section is not applicable to audio-only endpoints.

Follow these steps:

1. Verify the framing status or the frame alignment word (*FAW*). These fields detail the status of low level communication with the endpoint. If the *A-OUT* indication is **y**, the MCU has the endpoint's framing in that channel. If the call is dual channel, both channels should have *A-OUT* set to **y**. If a channel is not connected or does not have framing (both cases show **n**), the MCU will not use media from that endpoint in the conference.

   The *MCUFAL* count indicates how many times the MCU has lost the endpoint's framing (see also the *Fr Err* column on Page 1 of the form). Similarly, the *FEFAL* count indicates how many times the endpoint has lost MCU framing. If either of these counts is increasing over time (multiple commands are necessary to access the form over time), the endpoint is most likely not showing video from the MCU very well (due to freezing, tiling, etc). Both counts are indications of the stability of the connection(s). Incrementing counters show problems are occurring in the network between the MCU and the endpoint. A few “hits" during call establishment are normal.

2. If the MCU has the endpoint's framing in all channels, check the information under the "MODE COMMANDS / COMMUNICATION MODES" section. In the *STAT* column, all media (*CMD*s) except **56/64** should be **y**. If any value other than **56/64** is **n**, that endpoint's video will be affected. If any **CMD** is **n**, the MCU will not use that media in the conference. The *Audio* field is an exception to this.

   The indicator shows if the endpoint is matching exactly what the MCU wants (this is indicated in the *CONF* column). The endpoint may not be matching what the MCU wants and still participate in the audio portion of the conference. Since the information in the form is structured to show why an endpoint's bandwidth allocations for each media are invalid, it is necessary to indicate when the audio is at the wrong rate rather than when the MCU cannot decode the audio information.

3. Check the *EPT MISC* data.

   AIM indicates that the endpoint has muted the audio path to the MCU. The MCU will not VAS to an endpoint that has sent AIM even if it receives speech energy from the endpoint. The MCU transmits AIM if all other endpoints in the conference have sent the MCU AIM.

   TALK shows when speech energy is being received from an endpoint. The counter can help indicate a "stuck" speaker or provide supporting information about silent sites in the conference.

   VIS shows if an endpoint has suppressed the video stream. It is similar to muting the audio stream. The MCU will not VAS to an endpoint that has sent the MCU VIS. Endpoints usually send this code to the MCU when they have no video input. (For example, if the camera is off, selected video source is not providing video information to the endpoint.) The MCU transmits VIS if it has no input video to transmit.

   The *EPT MISC* field with **MIS** shows the user when the MCU has determined that the endpoint capability information shows that the endpoint can never meet the "Selected Communications Mode." Value **y** in this field indicates an endpoint from which the MCU will not use video or to which it will not send video. This indicator occurs when the selection is for a fixed audio mode (usually G.728) and when the endpoint does not support G.728.

   An endpoint that has the **MCV** indicator set to **y** is requesting that it be made the broadcaster. In a VAS or Chair conference, MCV forces the MCU to broadcast the site requesting MCV. If an endpoint does not send "cancel-MCV," the MCU will not be able to voice-activate switch the video. Therefore, users may complain about "locked" video. Endpoints use this BAS code when transferring still-frame graphics, and if the MCU does not receive the cancel-MCV indicator, it will get "stuck." Corrections include sending the graphic again or disconnecting and then reconnecting the offending endpoint.

   BCLS and **RTLS** indicate that the endpoint is "listening" to the video sent by the current broadcaster or return video source. It is unusual for both flags to be set to **y**; however, it is not an error.

BCTK and **RTTK** indicate that the endpoint is “talking” on the broadcast or return video channels. The counters are useful for checking if the MCU ever switched to a site and to note the “requently accessed” sites. Zeros for both counters indicate the MCU never switched to the endpoint as a video source.

HYPR indicates if the MCU thinks the endpoint is hyperactive. Hyperactive endpoints send too many changes in a small window of time. This behavior can cause problems for all endpoints connected to the MCU. Therefore, the hyperactive endpoint is isolated for five seconds and then allowed to “return” to the conference. While the endpoint is isolated, it is not allowed to be a video source. Some types of hyperactivity associated with audio mode changes can cause the endpoint to be audio muted by the MCU. The counter is useful to note problems that are momentarily transient in nature.

MUTE indicates if the TN788 audio decoder has muted the audio input. This will happen if the input protocol is unknown, the MCU has lost endpoint framing, or if the endpoint is hyperactive. The counter is useful to note problems that are momentarily transient in nature.

CONFERENCE INFO indicates the endpoint numbers (from the conference form) of the current video broadcaster and video return source. *Next Broadcaster* and *Next Return Vid* indicate the endpoints that will replace the broadcaster return video sources within a few seconds. If the fields are the same, a video switch is not pending.

Software Field Description

The software field appears on Pages 3 through 6 of the **status conference** command output and on Page 2 of the **status conference endpoint** command output. The software reason code is a 32-bit word that is divided into five parts. The code is illustrated in the following table.

Table 5-4. Software Reason Code for Conference and Endpoint Status

Part	Mask								Position	Comment
1	X	-	-	-	-	-	-	-	Upper Nyble Byte 4	Low Nyble of drop reason
2	-	X	-	-	-	-	-	-	Lower Nyble Byte 4	Unused
3	-	-	X	X	-	-	-	-	Byte 3	Frame Alignment
4	-	-	-	-	X	X	-	-	Byte 2	MCU FAL count
5	-	-	-	-	-	-	X	X	Byte 1	Port Type, FEFAL count or supplementary BONDing drop codes

The following list contains statements concerning the code.

- The high-order nyble of the high-order byte (left-most nyble of Byte 4) contains the unique information that isolates specific causes within a drop reason class.

  For the Status Conference form, the internal representation of the drop reason is split into two fields. Internally, the drop reason is stored as the unsigned character 0xFF. (See the *Drop Code* field on Page 1 of the **status conference endpoint** command output.) The high-order nyble (high four bits) 0xF0 is displayed as the ASCII drop reason. (However, in most cases, Dial-Out and BONDing are exceptions to this.) The low-order nyble (low 4 bits) 0x0F is displayed as the highest order nyble in the Software Reason field (0xF0000000). The aforementioned drop codes are described in the following table.

- Lower nyble of Byte 4 is not used.

- Byte 3 details Frame Alignment Signal (FAS) information described in the "Frame Alignment Signal Information (FAS)" section later in this chapter. (See also the FAW section on Page 2 of the **status conference endpoint** command output described earlier in this chapter.)

- Byte 2 is the MCU gain/loss H.221 framing count. This count is also shown in the *Fr Err* field of Page 1 output.

- The last byte (right-most byte or Byte 1) details the port type if the drop reason is **Bandwidth.** (See the "Port Type—Byte 1" section later in this chapter.)

  If the drop reason is **BondHshake**, the last byte contains supplementary bonding drop codes. (See the "Supplementary BONDing Drop Codes" section later in this chapter.) If the drop reason is anything else (including blank due to an active call), the low-order byte contains the far-end frame alignment loss count (FEFAL). This is a count of number of times an endpoint loses MCU framing. Framing loss is most likely due to network problems, such as noise, slips, etc. The FEFAL count also appears in the FAW section on Page 2 of the **status conference endpoint** command output described earlier in this chapter.

Table 5-5. Drop Codes Associated with the Software Field

Drop Reason	Software Reason	Drop Code	Comment
Far-end	0	00	Far-end disconnected (default)
Handshake	1	11	Frame Alignment Signal (FAS was not found
	2	12	Unsuccessful capability exchange
	3	13	No end of capability commands detected
	4	14	Proper Bit-rate Allocation Signal (BAS) channel was not found
	5	15	Initialization timer expired (while waiting for more than four events)
	6	16	BAS channel number does not match the FAS channel number
	8	18	BAS code received indicating initial BAS channel was lost
	9	19	FAS has been lost for more than 40 seconds
System	1	21	Reset System 2 occurred
	2	22	Internal maintenance has detected an invalid conference
Password	1	31	Password received was incorrect
	2	32	Password was not received within the required time interval
	3	33	Endpoint does not support passwords
Resource	0	40	System has insufficient TTRs to complete the call
	1	41	Resource (port) on the MMI circuit pack has failed
	2	42	Decoder or encoder port has failed on the VC circuit pack
	3	43	System has insufficient VC summers to complete the call
	4	44	System has tone generator problem
	5	45	System has insufficient MMI circuit pack resources to complete the call

Table 5-5. Drop Codes Associated with the Software Field — *Continued*

Drop Reason	Software Reason	Drop Code	Comment
	6	46	System has insufficient VC circuit pack ports to complete the call
	7	47	System has an internal exhaustion of timer buffers
	8	48	System has an internal exhaustion of queue buffers
	9	49	System has detected a DS1-to-MMI cable problem on an active connection
	A	4A	No TN744 ports are in service for UCC digit collection, or the TN744 port failed on an active UCC call
	B	4B	TN744 port failed while on an active UCC call.
	C	4C	System does not have sufficient time slot resources.
Bandwidth	5	55	Attempted to join a 64k conference at the wrong bandwidth
	6	56	Attempted to join a 56k conference at the wrong bandwidth
	7	57	Attempted to join a Wideband conference at the wrong bandwidth
	8	58	Bandwidth doesn't match previous channels
Conf End	0	60	Scheduled conference end occurred
Chair	1	71	Chair Controller used **drop endpoint** command
	2	72	Chair Controller used **drop conference** command
2-pri	1	81	Both sides of the cascade are administered as primary
2-sec	2	82	Both sides of the cascade are administered as secondary
Non-MCU	3	83	Far-end is not an MCU
Disconnect Notification	1	91	System detected a drop without notification from the endpoint.

Table 5-5. Drop Codes Associated with the Software Field — *Continued*

Drop Reason	Software Reason	Drop Code	Comment
	2	92	Endpoint notified the MCU before dropping. This should be considered a normal drop from the far end.
Unknown	0	A0	Internal software error
Agent	0	B0	CRCS or agent dropped the call
	1	B1	Endpoint was moved to another conference.
No-answer	1	C1	Call was answered and immediately dropped.
	2	C2	Dial-out call was dropped after the call rang
Busy	3	C3	Dial-out call was dropped; the far end was busy
No-trunks	4	C4	Dial-out call dropped on reorder (the network facilities are unavailable, or the endpoint is busy
Intercept	6	C6	Dial-out call dropped on intercept (an invalid number was dialed, or the far end was busy)
Far-end	7	C7	Dial-out call was dropped; the far end disconnected
No-ring	8	C8	Dial-out call dropped; the far end was left high and dry (no tones were received)
	9	C9	Dial-out call dropped; indicates cascading administration problem
	A	CA	Dial-out call dropped; the Call Classifier circuit pack is unavailable
Internal	B	CB	Dial-out call routing or trunk resource problem exists
PreAnsDrop	C	CC	Dial-out call dropped (by the network or the far end) before it was answered
Bondhshake	0	D0	BONDing handshake was lost
	1	D1	Information message had unacceptable parameters
	2	D2	Delay equalization timer expired

Table 5-5. Drop Codes Associated with the Software Field — *Continued*

Drop Reason	Software Reason	Drop Code	Comment
	3	D3	Additional channel setup timer expired while awaiting success for all channels
	4	D4	CID timer; Master channel negotiations failed
	5	D5	A-bit timer; multiframe alignment was not achieved
	6	D6	Incorrect BONDing framing was foun (the additional channel is not proper)
	7	D7	Setup sequence error; the event was received out of sequence
Ucc	1	E1	UCC dropped the endpoint
	2	E2	UCC ended the conference

For example, the value 0x60000000 and the drop reason **Resource** indicate that the system did not have enough VC resources to handle the call. The value 0x909c1513 and the drop reason of **Handshake** indicate the following: the MCU lost the endpoint's framing a total of 0x15 (21) times, the endpoint lost framing 0x13 (19) times while the endpoint was connected, and the MCU dropped the connection because it could not recover framing within a 40 second window. The value 0x5016008f and the drop reason **Bandwidth** indicate that a 56k trunk attempted to join a 64k conference.

Frame Alignment Signal Information (FAS) — Bytes 1 through 3

Frame Alignment information is divided into three parts. When the drop reason is not Bandwidth or BondHshake, Byte 1 gives far-end frame alignment loss. Byte 2 provides information on how many times framing was found and subsequently lost by the MCU. Byte 3 gives information on the FAS in a bitwise fashion as follows:

```
x--- ----

        1=>multiframe numbering present
        0=>multiframe numbering not present
-xxx ----
        xxx=>fas channel number
---- x---
        0=>mfa word not found
        1=>mfa word found
---- -x--       ( incoming A-bit )
        0=>endpoint found MCU framing
        1=>endpoint lost MCU framing
---- --x-       ( outgoing A-bit )
        0=>MCU found endpoint framing
        1=>MCU lost endpoint framing
---- ---x
        0=>Multichannel alignment achieved(2x56/64 calls only)
        1=>Multichannel alignment not achieved(2x56/64 calls only)
```

Figure 5-12. Frame Alignment Signal (FAS) Representation for Byte 3

The following table lists and describes the common FAS values.

Table 5-6. Common Frame Alignment Signal Values

Value	Description
0x98	Everything on the initial channel is okay.
0xa8	Everything on the first additional channel is okay.
0x9c	Far end lost H.221 framing
0x13	MCU lost H.221 framing
0x17	Far end and MCU lost H.221 framing
0x16	Never found FAS

Port Type—Byte 1

Whenever the drop reason code for the channel is **Bandwidth**, Byte 1 of the *Software* field contains information on the trunk port type. The port type will vary depending on the trunk type or, for ISDN trunks, the bearer capability associated with the call connected to the MCU. All voice port types can be valid for audio-only calls. Port types with values less than 0x8f are invalid for H.320 endpoints or cascade links. Port types less then 0x92 and greater than or equal to 0x8f are valid for Nx56 conferences. Port types greater than or equal to 0x92 are valid for Nx64 conferences.

Table 5-7. MCU Port Types and Associated Values

Type	Value	Description
voice	0x86	DS1 Tie Toll—DIU-T
	0x88	Voice DS1 trunk—DIU-DTT
	0x8a	Voice DS1 trunk—Diu-DTT with low tie/co loss
	0x8c	Voice Tie—DIU/AIU-TTS
56k	0x8f	ISDN-PRI restricted trunk—56k
	0x90	
	0x91	Robbed-bit 56k
64k	0x92	ISDN unrestricted trunk—64k
	0x93	Wideband > 64k per channel

Supplementary BONDing Drop Codes

Whenever the drop reason code for the channel is **BondHshake**, Byte 1 of the *Software* field contains the supplementary BONDing drop codes. These drop codes are associated with specific checks. The following codes are a result of checks made on the first Information Channel (IC) message received for a master channel when MCU is calling side (dial-out). (For dial-out, the MCU initializes the parameters desired for the call. The first IC received from the answering side must have parameters that match.)

Table 5-8. Supplementary BONDing Drop Codes for Dial-Out Initial Information Channel

Value	Description
0x01	Operating mode is not Mode 1.
0x02	Rate multiplier is not correct for the number of channels.
0x03	Bearer channel rate is not correct.
0x04	Revision level is not 0.
0x05	Channel ID is not 0.
0x06	Group ID is 0.
0x07	Transfer flag is not all binary 1s for the first IC.
0x08	Remote indicator cannot indicate call is delay equalized.
0x09	Remote loopback request indicates loopback.
0x0a	Sending endpoint is in loopback.

The following codes are a result of checks made on the first Information Channel (IC) message received for a master channel when MCU is answering side (dial-in). (In dial-in, the MCU receives the far end view of the call parameters. If they are not acceptable, the MCU tries to change them.)

Table 5-9. Supplementary BONDing Drop Codes for Dial-In Initial Information Channel

Value	Description
0x11	56k not allowed into an Nx64 conference
0x12	Bearer channel rate was 56k, but port type of call was 64k unrestricted
0x13	64k not allowed into an Nx56 conference
0x14	Bearer channel rate was 64k, but port type of call was not 64k unrestricted
0x15	Channel ID is not = 0
0x16	Group ID is not = 0
0x17	Transfer flag is not all binary 1s for the first IC
0x18	Remote indicator cannot indicate call is delay equalized
0x19	Remote loopback request indicates loopback
0x1a	Sending endpoint is in loopback

The following codes are a result of checks made on the first IC message received for an additional channel:

Table 5-10. Supplementary BONDing Drop Codes for Additional Information Channel

Value	Description
0x21	Operating mode is not MODE 1
0x22	Channel IDis not unique among other channels
0x23	Group ID does not match the master channel
0x24	Remote loopback request indicates loopback
0x25	Sending endpoint is in loopback
0x26	Revision level is not 0
0x27	Rate multiplier does not match the master channel
0x28	Bearer channel rate does not match the master channel

The following codes are a result of checks made against received framing indications:

Table 5-11. Supplementary BONDing Drop Codes for Received Framing Indications

Value	Description
0x31	Additional channel has IC alignment without multiframe alignment
0x32	Transparent endpoint has multiframe alignment on an additional channel

The following codes are a result of checks made against subsequent IC messages:

Table 5-12. Supplementary BONDing Drop Codes for Subsequent Information Channel Messages

Value	Description
0x41	Initial channel negotiations are completed, but the correct number of phone numbers were not exchanged
0x42	Transfer flag value changed when the channel ID was not 0
0x43	Transfer flag value and not the prior value is +1 while the channel ID is 0
0x44	Call is neither dial-out nor dial-in with digits = 1s
0x45	Operating mode is not Mode 1
0x46	Bearer channel rate cannot change.
0x47	Group ID cannot change
0x48	Rate multiplier cannot change
0x49	Remote loopback request cannot change
0x4a	Remote loopback indication cannot change

The following checks relate to timers:

Table 5-13. Supplementary Timer-Related BONDing Drop Codes

Value	Description
0x51	Initial channel negotiation procedures are not completed by a timeout

Monitor Conference Screen

The Monitor Conference screen allows you to monitor the status of a specified *active* conference. You can access this screen by issuing the command **monitor conference *conference-ID***, where ***conference-ID*** is a number identifying the conference. Issuing this command causes the system to display continually the status data of the conference with the indicated ID. Data on the screen is updated with new status every 30 seconds. If you enter the command and the conference is scheduled or has completed, the command terminates with the following message: “Conference status must be active or in-use.”

The “Update” key provides an on-demand status update. You can terminate the command by pressing the “Cancel” key. Whenever the command is canceled, the system returns to the command input mode.

The Monitor Conference screen appears as follows:

```
monitor conference                                              page 1 of 1


                          MONITOR CONFERENCE ___          Status: ________
   Conference Name: ____________________              Conf Bandwidth: ____
   Start Time/Date: _____ _____          Chair: __      Audio Mode: ________________
    Stop Time/Date: _____ _____      Fill Ept: __  Cont. Presence: ______ ____________
   Format (in/out): ________           Video Mode: ________________     Q1: __  Q2: __
  FPS 16C: __    4C: __ C: __   Q: __               Vid Mix: ______     Q3: __  Q4: __
          In  Drop                 Fr                  In   Drop                 Fr
      Ext Use Code  T120 Ts Vs Err       Ext    Use  Code   T120 Ts Vs Err
   1: _____ _   __    _  _  __ __    13: _____  _    __    _  _  __ __
   2: _____ _   __    _  _  __ __    14: _____  _    __    _  _  __ __
   3: _____ _   __    _  _  __ __    15: _____  _    __    _  _  __ __
   4: _____ _   __    _  _  __ __    16: _____  _    __    _  _  __ __
   5: _____ _   __    _  _  __ __    17: _____  _    __    _  _  __ __
   6: _____ _   __    _  _  __ __    18: _____  _    __    _  _  __ __
   7: _____ _   __    _  _  __ __    19: _____  _    __    _  _  __ __
   8: _____ _   __    _  _  __ __    20: _____  _    __    _  _  __ __
   9: _____ _   __    _  _  __ __    21: _____  _    __    _  _  __ __
  10: _____ _   __    _  _  __ __    22: _____  _    __    _  _  __ __
  11: _____ _   __    _  _  __ __    23: _____  _    __    _  _  __ __
  12: _____ _   __    _  _  __ __    24: _____  _    __    _  _  __ __
                                     25: _____  _    __    _  _  __ __
```

Figure 5-13. Sample Monitor Conference Screen

Fields in the top part of the screen contain conference-level information. The rest of the form displays endpoint-level fields, which can change after the endpoint joins the conference. All the fields in the screen are also contained in the “Status Conference” or “Status of Conference Endpoint” screens. Therefore, for a description of the fields, refer to the appropriate sections earlier in this chapter.

NOTE:
Drop codes that appear on the Monitor Conference screen are discussed in the “Software Field Description” section in this chapter.

Using the Screens for Troubleshooting

This section provides an example of how to interpret, for the purpose of troubleshooting, the information on the Status of Conference screens. The example involves a reserved conference that has been administered for nine conferees at the 64k rate. The assumption is that the **status conference *2*** command was entered to display the screens that follow. Another assumption is that the conference is currently active and was administered as indicated in the top half of the first screen. Here are the screens for our example.

```
status conference                                              page  1 of  6


   STATUS OF CONFERENCE: 2__       Status: in-use__         Conf Type: ____________
Conference Name: LuTech 12/11/98_____                 Conference Mode: voice-activated
       Password: 112358__________                                Chair: __
 Password Scope: audio_                Cascade Mode: primary__     Audio Mode: G.711-A_
          Class: reserved_                Data Mode: none_____       MLP Rate: _______
Start Time/Date: 12:00 DEC 11       Admin Bandwidth: 64k__       Lo/Hi Interworking? n
 Stop Time/Date: 14:00 DEC 11        Conf Bandwidth: 56k__          Rate Adaptation? y
Video Processor: ____ ________       Format(in/out): CIF______          SM Rate 1: _____
 VP Audio Delay: ____         FPS 16C: ___ 4C: ___ C: 30_ Q: 30_                2: _____
Q1: __  Q2: __  Fill Ept: __             Video Mode: H.261-down_only           3: _____
Q3: __  Q4: __   Vid Mix: full          H.263/H.261 Transcoding? n          FECC: __

                                Dial In ----Capability-----   SM Rt Bd
       Type  Ext  Meet-Me Number  Type Use Ch Au Vid T120 CC Gx Rt Ad Md Ts Vs
    1: P64_ 75000 (700)957-5939__  in_  y  y  y  y 1  _   _  y  _  y  _  t  b
    2: P64_ 75001 (700)957-5938__  in_  n  _  _  y 1  _   _  _  _  _  _  _  _
    3: AUD_ 75002 (700)957-5937__  in_  y  _  y  _ _  _   _  _  _  _  _  _  _
    4: CAS_ 75003 (700)957-5936__  out  y  y  y  y 1  _   _  y  _  y  _  t  r
    5: ____ _____ _______________  ___  _  _  _  _ _  _   _  _  _  _  _  _  _
    6: P64_ 75004 (700)957-5935__  out  c  y  c  y 1  _   _  e  _  y  _  _  _
    7: P64_ 75005 (700)957-5940__  out  c  y  y  y 1  _   _  y  _  c  _  t  _
    8: P64_ 75006 (700)957-5945__  in_  y  y  y  y 1  _   _  e  _  y  _  _  _
```

Figure 5-14. Sample Status Screen Used for Troubleshooting (1 of 4)

```
status conference                                              page 2 of 6

                    STATUS OF CONFERENCE ____    Status: ______

                                    Dial In  ----Capability----   SM Rt Bd
    Type  Ext  Meet-Me Number       Type Use Ch Au Vid T120 CC Gx Rt Ad Md Ts Vs
 9: ___ _____ ________________       ___  _   _  _  _ _  _    _  _  _  _  _  __
10: ___ _____ ________________       ___  _   _  _  _ _  _    _  _  _  _  _  __
11: ___ _____ ________________       ___  _   _  _  _ _  _    _  _  _  _  _  __
12: ___ _____ ________________       ___  _   _  _  _ _  _    _  _  _  _  _  __
13: ___ _____ ________________       ___  _   _  _  _ _  _    _  _  _  _  _  __
14: ___ _____ ________________       ___  _   _  _  _ _  _    _  _  _  _  _  __
15: ___ _____ ________________       ___  _   _  _  _ _  _    _  _  _  _  _  __
16: ___ _____ ________________       ___  _   _  _  _ _  _    _  _  _  _  _  __
17: ___ _____ ________________       ___  _   _  _  _ _  _    _  _  _  _  _  __
18: ___ _____ ________________       ___  _   _  _  _ _  _    _  _  _  _  _  __
19: ___ _____ ________________       ___  _   _  _  _ _  _    _  _  _  _  _  __
20: ___ _____ ________________       ___  _   _  _  _ _  _    _  _  _  _  _  __
21: ___ _____ ________________       ___  _   _  _  _ _  _    _  _  _  _  _  __
22: ___ _____ ________________       ___  _   _  _  _ _  _    _  _  _  _  _  __
23: ___ _____ ________________       ___  _   _  _  _ _  _    _  _  _  _  _  __
24: ___ _____ ________________       ___  _   _  _  _ _  _    _  _  _  _  _  __
25: ___ _____ ________________       ___  _   _  _  _ _  _    _  _  _  _  _  __
```

Figure 5-15. Sample Status Screen Used for Troubleshooting (2 of 4)

```
status conference                                              page 3 of 6

                          STATUS OF CONFERENCE 2__

                    Sum Grp: Group1  Group2  Group3  Group4
                         L1: 1a0716  1a0714  _______  _______
                         L2: 1a0715  1a0713  _______  _______

          Join  Drop      Drop     AC  -------Ports---------  Sum
    Ext   Time  Time      Reason   Num Trunk  Video  Aud/ESM  Grp Software
1: 75000 12:01 _____ ___________ ___ 1a1201 1a0801 1a0905    1  _________
         12:01 _____ ___________ ___ 1a1202 1a0802 _______       _________
2: 75001 12:06 12:07 Handshake_ ___ _______ _______ _______    _  _________
         _____ _____ ___________ ___ _______ _______ _______       _________
3: 75002 12:02 _____ ___________ ___ 1a1203 _______ 1a0907    _  _________
         _____ _____ ___________ ___ _______ _______ _______       _________
4: 75003 12:01 _____ ___________ _12 1a1215 1a0805 1a0908    1  _________
         12:01 _____ ___________ _13 1a1216 1a0806 _______       _________
5: _____ _____ _____ ___________ ___ _______ _______ _______    _  _________
         _____ _____ ___________ ___ _______ _______ _______       _________
6: 75004 12:02 _____ ___________ _14 1a1205 1a0803 1a0906    1  _________
         12:03 _____ ___________ _15 1a1206 1a0804 _______       _________
```

Figure 5-16. Sample Status Screen Used for Troubleshooting (3 of 4)

```
status conference                                              page 4 of 6

                              STATUS OF CONFERENCE 2__

          Join  Drop      Drop      AC  -------Ports--------  Sum
     Ext  Time  Time     Reason     Num Trunk   Video Aud/ESM  Grp Software
 7: 75005 12:01 _____ ___________ _16 1a1001 1a0807 1a1107    1  _________
          12:01 _____ ___________ _17 1a1002 1a0808 ______       _________
 8: 75006 12:01 _____ ___________ ___ 1a1005 1a0807 1a1108    1  _________
          12:01 _____ ___________ ___ 1a1006 1a0808 ______       _________
 9: 75007 12:01 _____ ___________ ___ 1a1007 1a0807 1a1105    1  _________
          12:01 _____ ___________ ___ 1a1008 1a0808 ______       _________
10: 75008 12:01 _____ ___________ _18 1a1202 1a0810 1a1109    2  _________
          12:01 _____ ___________ _19 1a1203 1a0811 ______       _________
11: _____ _____ _____ ___________ ___ ______ ______ ______    _  _________
          _____ _____ ___________ ___ ______ ______ ______       _________
12: _____ _____ _____ ___________ ___ ______ ______ ______    _  _________
          _____ _____ ___________ ___ ______ ______ ______       _________
13: _____ _____ _____ ___________ ___ ______ ______ ______    _  _________
          _____ _____ ___________ ___ ______ ______ ______       _________
14: _____ _____ _____ ___________ ___ ______ ______ ______    _  _________
          _____ _____ ___________ ___ ______ ______ ______       _________
```

Figure 5-17. Sample Status Screen Used for Troubleshooting (4 of 4)

The four screens indicate the following:

- Conference has been rate adapted from 64k to 56k.
- Conference is operating at G.711 audio mode.
- First administered endpoint is in use and fully connected. Also, this endpoint was a video source when the request was made. The endpoint has still graphics capability.
- User dialing (700) 957-5938 (the second administered user) failed to handshake correctly with the Lucent Technologies MCU.
- Third endpoint is an Audio Add-On endpoint that was required to enter the “112358” password in order to join the conference.
- Meet-me number (700)957-5936 (the fourth administered endpoint) has joined the conference as a dial-out cascade link.
- Fifth endpoint was either dynamically removed from the conference or never administered.
- Value **c** in the *In Use* field of the sixth administered endpoint indicates that this endpoint has changed the conference capability. A closer look at the *Capabilities* fields shows that the endpoint caused the audio operating mode to change from G.728 to G.711. Value **e** in the *Gx* field indicates that this endpoint does not have still graphics capability.
- Value **c** in the *In Use* field of the seventh administered endpoint indicates that this endpoint has changed the conference capability. A closer look at the *Capabilities* fields shows that this endpoint caused rate adaptation from 64k to 56k.
- The ninth administered endpoint never dialed in. (This endpoint might have dialed an incorrect Meet-Me number.)
- Value **f** in the *In Use* field of the tenth administered endpoint indicates that the endpoint failed to connect to all conference media. A closer look at the *Capabilities* fields shows the value **n** in the *Rate Adpt* field. This indicates that this endpoint had trouble rate adapting to the 56k rate. Value **i** in the *VS* field indicates that this endpoint is not considered a valid video source by the MCU.

Troubleshooting Feature-Related Problems

This section discusses how to troubleshoot feature-related problems in MCU conferences.

NOTE:
Endpoints are often easier to troubleshoot by moving them from their original conference to another conference. The **control conference CNF_ID-From move EPT_ID-From CNF_ID-To** command can be used to move an endpoint to another conference for this purpose. The command can also be used to move the endpoint back to its original conference. Refer to the "Moving an Endpoint to Another Conference" section in Chapter 4 of this document for details.

To track the movement of endpoints between conferences, be sure to check the *Related Conf ID* and *Related Ept* fields on Page 1 of the Status of Conference Endpoint screen, which is discussed earlier in this chapter.

Dial-In Conferee Cannot Join a Conference

If a dial-in conferee has tried unsuccessfully to join a conference (conferee may see a visual indication or hear a busy signal, or reorder tone), take the following actions:

1. Ask the conferee which conference was being joined (ask for the day, start time, convener's name, and purpose of the conference).
2. Log in as **agent** on the MCU-ST.
3. Find the Conference Appointment Form for the conference and note the *Conference ID*. You can also enter **list conference** for a listing of the day's conferences. Locate the *Conference ID* number for the conference in question.
4. Enter **display conference *ID*** where ID is the *Conference ID* number found in Step 3. The Conference Record screen is displayed (see the next figure).

```
change conference 1                                    Page  1 of  8

                              CONFERENCE RECORD

      Conference ID: 1                                  Status: inactive
    Conference Name: Exec 384k                Conference Mode: voice-activated
    Conference Type: _______________
                                                 Cascade Mode: _________
         Billing ID: _________________             Audio Mode: auto-w/upgrade
     Password Scope: ____________                   Bandwidth: 384k_
                                                  Speed Match? y
            Class: reserved___                 Entry/Exit Alert? tone
       Start Time: 13:00                          Warning Alert? tone
        Stop Time: 15:00
   OneNumber Access? n
Continuous Presence? n  Executive Conference? y     Video Mode: H.263/1-w/upgrade
 Site1 Ext: _____  Site2 Ext: _____
 Site3 Ext: _____                       H.263/H.261 Transcoding? n
  Starting Image: executive_
 Force Video Now to: ____________                  T.120 Data Mode: none
  Lip Sync  <= 128K: custom  Value: 200_                Far End Camera Control? n
  Lip Sync 224/256K: none
  Lip Sync  >= 336K: auto
```

Figure 5-18. Conference Record Screen (Page 1 of 8)

5. Check the following fields on the Conference Record screen:
   - On the first page of the record, check the *Start Time* and *Stop Time* to be sure the conferee has the correct time of the conference. If the conferee is dialing in even one minute early, the call will be rejected. Note also if there is a time zone difference between the scheduled conference time and the location of the endpoint.
   - On the second through eighth page of the record, check the *Name* field to see if the conferee has dialed the right Meet-Me number. Make sure the Meet-Me number is assigned to the conference in question.
   - If there is a **2** in the *No. of Channels* field, ask the conferee if the *same* Meet-Me number was dialed *twice.* A 2-channel 56k/64k conference requires dialing the same Meet-Me number twice to connect both channels.
6. Enter **display time** to see if there was a time change (for example, daylight savings time) or a power outage that caused the MCU system clock to be incorrect. If the MCU system clock is wrong, notify your telecommunications manager.
7. Enter **status conference *conference-ID*** where ***conference-ID*** is the *Conference-ID* number found in Step 3. The Status of Conference screen is displayed (see Figure 5-1).

8. *Check the Stop Time/Date field on the Status of Conference screen.* A **blank** means the conference has started and is still in progress.

   If there is a time recorded in the *Stop Time/Date* field, the conference is over. Make sure the conference did not end early and that the conferee is calling from the same time zone.

   A conference reserved for an earlier time (that is, reached its *Stop Time*) will show **complete** in the *Status* field. If this is the case, make sure the conferee is not calling from a different time zone. However, check the screen to make sure you are displaying data about the right conference.

   NOTE:
   A Status of Conference screen is not available for a conference that has not started. However, a Status of Conference screen will appear for any previous conference with a status of “complete” that used the same *Conference ID*. Be sure you are viewing the right conference. Conferences currently in progress have a status of “active” or “in-use.”

9. *Check the In Use* field.

   A **y** or **c** indicates the MCU-extension is being used by a conferee. The convener may have distributed the same number to two conferees. Give the conferee another MCU-extension assigned to the conference that has a blank in the *In Use* column.

   If there is a blank in the *In Use* column next to the dial type the conferee is using, the call never made it to the MCU. This indicates a problem with the endpoint or the network. Escalate the problem (see “When All Else Fails”).

   If there is an **n** in the *In Use* column, the endpoint call arrived at the MCU but was dropped. See “Conferee Dropped from Conference” for the drop reason.

10. If **Resource** appears in the *Drop Reason* field, see the “Conferee Dropped from Conference” section for an explanation of the drop reason.

Dial-Out Conferee Cannot Join a Conference

If a dial-out conferee has tried unsuccessfully to join a conference, take the following actions:

1. Ask the conferee which conference was being joined (ask for the day, start time, convener’s name, and purpose of the conference). Also, ask the conferee for the dial-out number or location identification information (for example, Location Name, Room Number, etc.). The dial-out numbers are needed to associate the endpoint with the administered data on the Status Conference forms.

2. Log in as **agent** on the MCU-ST.

3. Find the Conference Appointment Form for the conference and note the *Conference ID*. You can also enter **list conference** for a listing of the day's conferences. Locate the *Conference ID* number for the conference in question.
4. Enter **display conference *conference-ID,*** where ***conference-ID*** is the *Conference ID* number found in Step 3. The Conference Record screen is displayed (see the next figure).

```
change conference 1                                    Page  1 of  8

                              CONFERENCE RECORD

      Conference ID: 1                                    Status: inactive
    Conference Name: Exec 384k                  Conference Mode: voice-activated
    Conference Type: _______________
                                                   Cascade Mode: _________
         Billing ID: _________________               Audio Mode: auto-w/upgrade
     Password Scope: ____________                     Bandwidth: 384k_
                                                    Speed Match? y
              Class: reserved___                Entry/Exit Alert? tone
         Start Time: 13:00                         Warning Alert? tone
          Stop Time: 15:00
   OneNumber Access? n
Continuous Presence? n  Executive Conference? y    Video Mode: H.263/1-w/upgrade
 Site1 Ext: _____  Site2 Ext: _____
 Site3 Ext: _____                       H.263/H.261 Transcoding? n
  Starting Image: executive_
 Force Video Now to: ____________                T.120 Data Mode: none
  Lip Sync  <= 128K: custom  Value: 200_                 Far End Camera Control? n
  Lip Sync 224/256K: none
  Lip Sync  >= 336K: auto
```

Figure 5-19. Conference Record Screen (Page 1 of 8)

5. Check the following fields on the Conference Record screen:
   - On the first page of the record, check the *Start Time* and *Stop Time* to be sure the conferee has the correct time of the conference. Note also if there is a time zone difference between the scheduled conference time and the location of the endpoint.
   - Check the administered dial-out numbers against the information obtained in Step 1 to verify that the correct dial-out numbers were administered.
   - If there is a **2** in the *No. of Channels* field, ensure that both dial-out numbers are specified correctly.
6. Enter **display time** to see if there was a time change (for example, daylight savings time) or a power outage that caused the MCU system clock to be incorrect. If the MCU system clock is wrong, notify your telecommunications manager.
7. Enter **status conference *conference-ID*** where ***conference-ID*** is the *Conference ID* number found in Step 3. The Status of Conference screen is displayed (see Figure 5-1).
8. *Check the Stop Time/Date field on the Status of Conference screen.* A **blank** means the conference has started and is still in progress.

   If there is a time recorded in the *Stop Time/Date* field, the conference is over. Make sure the conference did not end early and that the conferee is in the same time zone.

   A conference reserved for an earlier time (that is, reached its *Stop Time*) will show **complete** in the *Status* field. If this is the case, make sure the conferee is not in a different time zone. However, check the screen to make sure you are displaying data about the right conference.

   NOTE:
   A Status of Conference screen is not available for a conference that has not started. However, a Status of Conference screen will appear for any previous conference with a status of “complete” that used the same *Conference ID*. Be sure you are viewing the right conference. Conferences currently in progress have a status of “active” or “in-use.”

9. *Check In Use* field.

   A **y** or **c** indicates the MCU-extension is being used and that the call is in the process of being connected. This may be true because the call failed to establish the first time and the MCU is attempting to reestablish the call. Check the join/drop times for both the channel and associated drop reasons.

   A **blank** indicates that the call never made it out of the MCU. This could indicate a problem with the dial-out number. If the administered dial-out numbers are correct, escalate the problem (see “When All Else Fails”).

An **n** indicates that the outgoing call was made on behalf of this MCU-extension but was dropped.

The following drop reasons on the endpoint channels indicate a dial-out failure:

- Busy

  The destination endpoint is busy. This drop reason is detected by Lucent Technologies MCU CPTR resources or via an ISDN cause value (for example, H0). Ensure that the destination endpoint is idle.

- Internal

  The MCU cannot allocate the trunk resources necessary to route the dial-out call for the specified dial-out numbers. This can indicate a problem with the routing pattern or the trunk-associated translation (for example, the Task Access Code (TAC) specified in the dial-out number or routing pattern points to a trunk group without members). This can also indicate a lack of trunk resources (for example, all trunk members are maintenance-busy, or all in-service members are busy on a call).

- No-ring

  The network has not returned ringing for more than 30 seconds. Verify the destination number.

- No-trunks

  The Lucent Technologies MCU has detected there are no available trunks in the network to place the call. This drop reason is detected by Lucent Technologies MCU CPTR resources. Escalate the problem.

- No-answer

  The destination's endpoint range and did not answer after more than 60 seconds. Check the equipment for this endpoint.

- Intercept

  The MCU has either detected an intercept tone from the network or received an intercept ISDN cause value. This could indicate a wrong destination number or a busy far end.

- Reorder

  The MCU has detected that there are no available trunks in the network to place the call. This drop reason is detected by the Lucent Technologies MCU CPTR resources.

- Wrong-num

  The MCU has detected that the wrong destination number was dialed. This drop reason is detected either by Lucent Technologies MCU CPTR resources, which detect the intercept tone from the network, or by an ISDN cause value.

If the drop reason itself does not identify the problem, see the “When All Else Fails” section.

Cascaded Call Deficient

It is possible that some conferees will join a cascaded conference but other conferees expected to join the conference will fail to do so. Whenever this happens, you should investigate the state of the cascaded link.

Whenever two MCUs are cascaded (joined) in a conference, one of the MCUs is set up to perform a dial-out and the other MCU is set up to receive the call. Usually, problems relevant to MCU call setup are the same as for other video endpoints. Therefore, you can use the same troubleshooting procedures to resolve these problems. However, in addition, the MCU provides three drop reasons (explained later) to identify problems relevant to cascading. The MCU performing the dial-out does not retry the call(s) for these drop reasons. Accordingly, you must modify the cascade mode or change the dial-out numbers on the Status Conference Record for one or both MCUs.

If the call is a dial-in call, refer to the steps in “Dial-In Conferee Cannot Join Conference” to track the problem. For a dial-out call, refer to the steps in “Dial-Out Conferee Cannot Join Conference.” In either case, you should investigate both the dial-in and dial-out MCUs.

If, in Step 9 of the procedure in either section, there is an **n** in the *In Use* column of the Status Conference forms for the CAS endpoint, the call was established on behalf of this MCU extension but was dropped.

Drop reasons specific to cascading include the following:

- 2-pri

  The cascading mode for both MCUs is specified as “primary” on the Status Conference forms. However, the cascading mode for one of the MCUs must be “secondary.” Change the cascading mode on Page 1 of the associated Status Conference record.

  If there is no preference that a specific MCU be the primary MCU, make the change on the dial-out MCU. This will allow the dial-out MCU to retry cascaded calls.

  If there is a preference that a dial-in MCU be the secondary MCU, you must change the administration of both MCUs. Specifically, you must change the dial-in MCU cascade mode designation to “secondary” and temporarily disable the dial-out MCU cascade mode designation. You must then submit the Status Conference Record form. Finally, you must change the mode from “disabled” to “primary.” This will cause the dial-out MCU to retry cascaded calls.

- 2-sec

  The cascading mode for both MCUs is specified as “secondary” on the Status Conference forms. However, the cascading mode for one of the MCUs must be “primary.” Change the cascading mode on Page 1 of the associated Status Conference record.

  If there is no preference that a specific MCU be the primary MCU, make the change on the dial-out MCU. This will allow the dial-out MCU to retry cascaded calls.

  If there is a preference that a dial-in MCU be the primary MCU, you must change the administration of both MCUs. Specifically, you must change the dial-in MCU cascade mode designation to “primary” and temporarily disable the dial-out MCU cascade mode designation. You must then submit the Status Conference Record form. Finally, you must change the mode from “disabled” to “secondary.” This will cause the dial-out MCU to retry cascaded calls.

- Not-MCU

  The destination on the dial-out MCU was not an MCU. Verify that the MCU-extension administered on both MCUs for the cascaded link is of type CAS or BCAS and that the dial-out number(s) on the dial-out MCU is (are) the correct number(s) to dial to reach the CAS MCU-extension on the dial-in MCU.

  The endpoint that called the CAS extension on the dial-in MCU was not an MCU. This could indicate that one of the H.320 conferees dialed the Meet-Me Number associated with the CAS MCU-extension. Verify that the dial-out MCU has the correct dial-out number(s) administered for the CAS MCU-extension.

  NOTE:
  For non-cascaded conference drop reasons, see “Conferee Dropped from Conference.”

Conferee Cannot Get Video

If a conferee has joined a conference but does not receive video, do the following:

1. Ask the conferee which conference was joined (ask for the start time, convener’s name and purpose of the conference).
2. Log in as **agent** on the MCU-ST.
3. Find the Conference Appointment Form for the conference and note the *Conference ID*. You can also enter **list conference** for a listing of the day’s conferences. Locate the *Conference ID* number for the conference in question.

4. Enter **display conference** ***conference-ID***, where ***conference-ID*** is the *Conference ID* number from Step 3. The Conference Record screen is displayed.
5. C*heck the Status field* on the Conference Record screen to see if the conference is **in-use**, **active** or **complete**. If the conference is in-progress, the status should be **active** or **in-use**. However, check the screen to make sure you are displaying data about the right conference.
6. Check the *No. of Channels* field. If there is a **2** in the field, ask the conferee if the *same* Meet-Me number was dialed *twice*. A 2B-channel 56k/64k conference requires dialing the same Meet-Me number twice to connect both channels.
7. Enter **status conference** ***conference-ID*** where ***conference-ID*** is the *Conference ID* number from Step 3. The Status of Conference screen is displayed (see Figure 5-1).

   NOTE:
   A Status of Conference screen is not available for a conference that has not started. A conference that has a class of “reserved” but has not begun displays **inactive** in the *Status* field of the Conference Record.

8. Check the *Has Capability...Ch field.*

   A **y** in this field means the endpoint has the required number of channels.

   An **e** in this field means the video endpoint has not declared support for the correct number of channels and cannot join the conference as a full video participant.

   An **n** means the video endpoint has declared the correct number of channels but all channels have not yet joined the conference. Ensure that the dial-in conferee dialed the *same* Meet-Me number *twice* (for a 2B 56k/64k conference) or just once (for any multirate or BONDed conference), or that the dial-out conferee’s conference record has the endpoint numbers specified the appropriate number of times (twice or once). If so, this may be a network or video endpoint problem; escalate the problem (see “When All Else Fails”).

9. Chech the *Data Mode* field. If this field contains **any-mlp**, check the *Capability...Mlp* field.

   An **e** in this field indicates that the endpoint does not support MLP data and can join the conference only as a secondary audio-only endpoint (this may indicate an endpoint problem).

   An **m** in this field indicates that the endpoint has declared the MLP data capability, but the MLP channel has not been opened to or from the endpoint. This may occur when the endpoint does not support the G.728 (LD-CELP) capability (that is, when the *Capability...Aud* field contains **e**). If this is true, MLP is not opened to the endpoint, and the endpoint can join the conference only as a secondary audio-only endpoint. Otherwise, for a

dial-out call, try rejoining the conferee to the conference via a redial. For a dial-in call, ask the conferee to drop and then dial in to the conference again. If the data call is again unsuccessful, escalate the problem (see the "When All Else Fails" section later in this chapter).

10. Check the administered *Audio Mode* field on the Conference Record by issuing the **display conference** command. If the field contains **G.728**, check the *Capabilities...Aud* field. The value **e** in this field indicates an endpoint capability problem because the endpoint does not have G.728 (LD-CELP) audio capability. Such an endpoint can join the conference only as a secondary audio-only endpoint.
11. Check the *Interworking* field. If the field contains **y**, check the *Rate Adpt* field. If this field contains **5** or **6**, all of these values together indicate that the endpoint is a low-speed (56k or 64k) endpoint interworking to a higher (112k or higher) bandwidth conference. A low-speed endpoint can join a higher bandwidth conference only as a secondary audio-only endpoint.

Conferee Cannot Get T.120 Data

If a conferee has joined a conference and is expected to receive T.120 data but does not receive it, do the following:

1. Ask the conferee which conference was joined (ask for the start time, convener's name and purpose of the conference).
2. Log in as **agent** on the MCU-ST.
3. Find the Conference Appointment Form for the conference and note the *Conference ID*. You can also enter **list conference** for a listing of the day's conferences. Locate the *Conference ID* number for the conference in question.
4. Enter **display conference *conference-ID***, where ***conference-ID*** is the *Conference ID* number from Step 3. The Conference Record screen is displayed.
5. C*heck the Status field* on the Conference Record screen to see if the conference is **in-use**, **active** or **complete**. If the conference is in-progress, the status should be **active** or **in-use**. However, check the screen to make sure you are displaying data about the right conference.
6. Check the *Data Mode* field on the Conference Record. If the field does not contain **any-mlp**, the conference is not a data conference.
7. Enter **status conference *conference-ID*** where ***conference-ID*** is the *Conference ID* number from Step 3. The Status of Conference screen is displayed (see Figure 5-1).

NOTE:
A Status of Conference screen is not available for a conference that has not started. A conference that has a class of "reserved" but has not begun displays **inactive** in the *Status* field of the Conference Record.

8. *Check the Capability...Ch1 field.* A **y** in this field means the endpoint has the required number of channels.

   An **e** in this field means the data endpoint has not declared support for the correct number of channels and cannot join the conference as a full data participant.

   An **n** means the data endpoint has declared the correct number of channels but all channels have not yet joined the conference. Ensure that the dial-in conferee dialed the *same* Meet-Me number *twice* (for a 2B 56k/64k conference) or just once (for any multirate or BONDed conference), or that the dial-out conferee's conference record has the endpoint numbers specified the appropriate number of times (twice or once). If so, this may be a network or data endpoint problem; escalate the problem (see "When All Else Fails").

9. Check the *Capability...Mlp* field. The following list indicates and explains each of the possible values for this field.

   - **y** — the data connection is fully established.
   - **e** — the endpoint has not declared the MLP data capability (this may be an endpoint problem.
   - **m** — the MLP channel was not opened to or from the endpoint (this may be an endpoint problem). Check the *Capability...Aud* field. If the field contains **e**, this is an endpoint capability problem. Check the *Aud/ESM* field on pages 3 through 6 of the Status Conference forms or on the Status Conference Endpoint form for the ESM resource port assignment. The ESM port is displayed in the second channel of the *Aud/ESM column*. If the ESM port is missing, escalate the problem (see the "When All Else Fails" section later in this chapter). Otherwise, for a dial-out call, try rejoining the conferee to the conference via a redial. For a dial-in call, ask the conferee to drop and then dial in to the conference again. If the data call is again unsuccessful, escalate the problem (see the "When All Else Fails" section later in this chapter).
   - **a** — the endpoint is specified on the Conference Record as an endpoint that runs non-compliant applications. This means that the full connection to MCS will not be established.
   - **t** — the endpoint has the MLP data capability and the MLP channel is opened, but the endpoint has no T.120 capabilities, or it did not confirm the **T.120 on** message. Since the MLP channel is open toward the endpoint, this endpoint is joined to the conference as an audio/video endpoint.
   - ***** — the link to the ESM is down. Since connection to MCS cannot be established, escalate the problem (see the "When All Else Fails" section later in this chapter).
   - **0** — the connection to the ESM cannot be established. Therefore, escalate the problem (see the "When All Else Fails" section later in this chapter).

- **1** or **2** — there are problems with the connectivity to the T.120 stack. Tell the conferee to close all windows relevant to data and to then reopen them. If this fails to clear the problem, escalate the problem (see the “When All Else Fails” section later in this chapter).

Conferee Dropped from Conference

The MCU records the reason why a video endpoint dropped from a conference on Pages 2 and 3 of the Status of Conference screen.

1. Ask the conferee which conference was joined (ask for the start time, convener’s name and/or purpose of the conference).
2. Log in as **agent** on the MCU-ST.
3. Find the Conference Appointment Form and note the *Conference ID* number. You can also enter **list conference** for a listing of the day’s conferences. Locate the *Conference ID* number for the conference in question.
4. Enter **status conference *conference-ID*** where ***conference-ID*** is the *Conference ID* number located in Step 3. The Status of Conference screen is displayed.
5. Confirm you are viewing the right conference. Then, on Page 1 of the Status of Conference screen note the MCU-extension of the conferee on the *Ext/Meet-Me Number* line. Check the MCU-Extension and Number to Dial Form to match the MCU-extension on the Status of Conference screen with the number the conferee dialed to join the conference.
6. If there is an **n** in the *In Use* column for the MCU-extension, the video endpoint call arrived at or was initiated by the MCU but was dropped. Make sure the conference *Stop Time* has not been reached.
7. Go to Pages 2 and 3 of the Status of Conference screen to find the drop reason.
8. Press PAGE to display the next page of the Status of Conference screen.
9. For the MCU-extension of the conferee, find the *Drop Reason* column. If there are more than six conferees, check Page 3 for the remaining MCU-extensions. The possible values are as follows:

   - 2-pri

     See “Cascaded Call Deficient.”

   - 2-sec

     See “Cascaded Call Deficient.”

   - Agent

     The reservations agent caused the call to drop (for example, the agent changed a connected dial-out destination number).

- Bandwidth

  Indicates a bandwidth mismatch between a call and a conference the call attempted to join. For example, a 56k call attempted to join a 64k conference that does not allow rate adaptation.

- BondHshake

  One of the following is true:

  — Information channel parameter is not supported or is invalid.

  — Parameter negotiation terminated out of sequence.

  — Timer expired because one of the secondary channels did not establish.

  — BONDing framing was not detected for one of the other channels.

- Busy

  See “Dial-Out Conferee Cannot Join a Conference.”

- Chair

  The Chair dropped the endpoint by using either the Chair Command Disconnect (CCD) or Chair Command Kill (CCK) signal.

- Conf End

  The conference ended because either the stop time for a reserved conference was reached or an active conference was converted to a file.

- Far-End

  Clearing was received from the DS1 circuit pack. The disconnect came from either the network or the endpoint.

- Handshake

  Framing was either never found (that is, the endpoint could not complete initialization, or there were problems finding FAS and MFA and getting a correctly coded cap-set) or was lost for more than 40 seconds and the endpoint was disconnected.

- Internal

  See “Dial-Out Conferee Cannot Join a Conference.”

- No-answer

  See “Dial-Out Conferee Cannot Join a Conference.”

- No-Ring

  See “Dial-Out Conferee Cannot Join a Conference.”

- No-Trunks

  See “Dial-Out Conferee Cannot Join a Conference.”

- Not-MCU

  See “Cascaded Call Deficient.”

- Password

  The user entered the wrong password, or the Audio Add-On user did not enter the password within 60 seconds (the waiting period). The Audio Add-On user gets one attempt to enter a correct password and inter-digit timing for each digit (about 10 seconds between each digit).

- PreAnsDrop

  The call disconnected before being answered by an endpoint due to either the network, an endpoint, or a terminal adaptor. This drop reason is different from “No-answer,” which indicates that a 60 second timeout occurred while alerting. For “PreAnsDrop,” the call drops before the 60 second timer expires. Some busy endpoints connected through terminal adaptors display this behavior.

- Reorder

  See “Dial-Out Conferee Cannot Join a Conference.”

- Resource

  One of the following reasons is relevant:

  — The VSP-MCU could not provide resources (VC or MMI) when the call arrived, or it lost the resources during the call. This happened because the resources were either out of service, busied out by the craft, or were being used by system maintenance.

  — The DS1/MMI cable was disconnected.

  If there was a resource problem when the call arrived, the call would get a reorder (fast busy), and it would not be dropped by the VSP-MCU.

- System

  An Lucent Technologies MCU restart (Level 2) dropped all calls.

- UCC

  The UCC controller intentionally dropped the endpoint.

- Unknown

  The system could not determine the cause for the disconnect (for example, a software glitch).

- Intercept

  See “Dial-Out Conferee Cannot Join a Conference.”

- Wrong-num

  See “Dial-Out Conferee Cannot Join a Conference.”

10. If you are unable to resolve the problem, see “When All Else Fails” to escalate the problem.

Video Quality Decreased

The MCU attempts to provide the best video quality for a conference. It is possible that as conferees join a conference, the first few video endpoints may support high video quality (G.728) and so those locations initially receive the best video available. However, as other endpoints join that do not support G.728, the MCU adjusts the audio mode in use so all the video endpoints can participate in the conference.

As a result, the original conferees may notice a degradation of the video quality. Even if the non-G.728 video endpoints disconnect from the conference, the conference remains at the lower-quality video.

If you receive a report from the conference convener or conferees about reduced video quality, perform the following steps:

1. Ask the conferee which conference was joined (ask for the start time, convener’s name and/or purpose of the conference).
2. Log in as **agent** to the MCU-ST.
3. Find the *Conference ID* number on the Conference Appointment Form or enter **list conference** and note the Conference ID**.**
4. Enter **status conference *conference-ID,*** where ***conference-ID*** is the *Conference ID* located in Step 3. The Status of Conference screen is displayed.
5. On Page 1 of the Status of Conference screen, check the *Vid* column. If there is a **c**, the conference video quality was changed when a video endpoint that did not support the higher-quality video joined. When this occurs, the entire conference is adjusted to accommodate the endpoint.
6. Check the *Aud* column. If there is a **c** in the field, the audio quality was changed when a video endpoint that did not support the audio capabilities of the other endpoints joined. When this occurs, the entire conference is adjusted to accommodate the endpoint.
7. Check the *Rate Adaptation* field. If the field contains entry **y**, check the *Admin Bandwidth* and *Conf Bandwidth* fields. If *Admin Bandwidth* contains **64k** and *Conf Bandwidth* contains **56k**, the rate of the 64k conference has changed to 56k because a 56k endpoint joined the conference.

Conferee Cannot be Seen by Other Participants

A conferee may not be seen by other conference participants even if these participants have full video priviliges (that is, even if the participants are *not* audio-only participants). To identify this problem, you must first obtain conference and conferee (Meet-Me number and MCU-extension) information. For a dial-in conferee, perform Steps 1 through 7 of the “Dial-In Conferee Cannot Join a Conference” section earlier in this chapter. For a dial-out conferee, perform Steps 1 through 7 of the “Dial-Out Conferee Cannot Join a Conference” section earlier in this chapter. Then enter the **status conference** command and do the following:

1. Check the administered endpoint type associated with the problem call. If the type is **BOND**, verify that the connected endpoint supports BONDing. If the endpoint supports only 2B capability, the endpoint will always be set up in audio-only (transparent) mode. In such a case, the *Bond Mode* field contains entry **t**, indicating transparent mode.
2. If the type is **H.320**, check the *Capability Ch* field. If this field has entry **y**, check the *Admin Bandwidth* and *Conf Bandwidth* fields. If the *Admin Bandwidth* field has entry **64k** and the *Conf Bandwidth* field has entry **56k**, check the *Rate Adpt* field for the problem endpoint. Entry **n** in the field indicates that the conference rate adapted from 64k down to 56k but the endpoint had problems adapting. Such an endpoint may still receive video and participate in the audio portion of the conference, but it cannot be a valid video source. Attempt to set up the call again. Ask the dial-out user to drop the call and dial in again. If these actions fail to correct the problem, check if the dial-in user can originate a 56k 2B call instead of a 64k 2B call to join the conference. For a dial-out user, change the endpoint to a dial-in endpoint (that is, remove the destination numbers for this endpoint on the Conference Record form); then change the endpoint back to a dial-out endpoint.
3. Entry **n** in the *Capability Ch* field indicates that the endpoint declared it can support 2B channels but one of the channels did not connect due to either a network or endpoint problem. Before escalating the problem, attempt to set up the call again. Ask the dial-out user to drop the call and dial in. Ensure that the dial-in user dialed the correct Meet-Me numbers for both channels. For a dial-out user, change the endpoint to a dial-in endpoint, then change it back to a dial-out endpoint to initiate the call. Ensure that the correct Meet-Me numbers for both channels are specified on the Conference Record form.
4. Entry **e** in the *Mlp* field indicates that the endpoint does not have MLP capability. Therefore, the endpoint joined to the conference as an audio-only endpoint.
5. Entry **e** in the *Aud* field indicates that the endpoint does not support G.728 (LD-CELP) audio mode. Therefore, the endpoint is joined to the conference as an audio-only endpoint.

Conferee Lost Audio or Video

For a dial-in call, ask the conferee to drop from the conference and attempt to rejoin. For a dial-out call, attempt to redial by entering **y** into the *Redial?* flag in the Conference Record. If the solution for either problem fails, see “When All Else Fails” to escalate the problem.

All Conferees Were Dropped

If all conferees are dropped from a conference that was in progress, use the following procedure:

NOTE:
Conferees are dropped vis-a-vis the MCU that services them. Therefore, in a cascaded conference, it is possible that some conferees will be dropped while others will not be dropped.

1. Ask the conferee which conference was joined (ask for the start time, convener’s name, and/or purpose of the conference).
2. Log in as **agent** on the MCU-ST.
3. Find the *Conference ID* number on the Conference Appointment Form or enter **list conference** and note the *Conference ID***.**

   NOTE:
   If you are trying to locate a reserved conference, you can enter **list conference class reserved** for a listing of reserved conferences only. Likewise, if you are trying to locate a dedicated conference, you can enter **list conference class dedicated**.

4. Enter **display conference *conference-ID*** where ***conference-ID*** is the *Conference ID* number located in Step 3. The Conference Record screen is displayed.
5. Check the *Stop Time* field on the Conference Record screen to make sure the conference has not ended.
6. Check the *Status* field. If the *Status* is **complete**, the *Stop Time* of the conference has been reached and all conferees were dropped.
7. If the *Status* is **inactive**, check the *Class* field. If **file** appears in the *Class* field, the conference was canceled when it was changed from **reserved** or **dedicated** to **file**.
8. If you cannot diagnose the problem, see “When All Else Fails” to escalate the problem.

H.320 Party Dials Audio Add-On Number

If a H.320 party tries to enter a conference by dialing a number intended for an audio-only endpoint, the caller hears a fluttering sound and is then dropped. The drop reason “Far-End” appears on the Status Conference report. If a H.320 party calls the reservations agent to complain about not being able to enter the conference, the agent can diagnose the problem by issuing the **status conference** command and then checking which multimedia parties have not yet joined the conference. The agent can then give the correct number to the multimedia party.

Audio Add-On Party Dials H.320 Number

If an Audio Add-On party tries to enter a conference by dialing a number intended for a H.320 endpoint, the call is dropped. The drop reason “Handshake” appears on the Status Conference report. If an Audio Add-On party calls the reservations agent to complain about not being able to enter the conference, the agent can diagnose the problem by issuing the **status conference** command and then determining the correct number for the Audio Add-On party. The agent can then give the correct number to the Audio Add-On party.

Calls Involving BONDing Deficient

A conference with BONDing numbers or participants may encounter problems. The following sections discuss these problems and how you should deal with them.

NOTE:
For problems vis-a-vis cascade BONDing, refer to the “Cascaded Call Deficient” section earlier in this chapter.

NOTE:
Refer to the Status Conference Endpoint screen for debugging purposes.

H0 Participant Dials a BONDing Number

If an H0 participant tries to enter a conference by dialing a number intended for a BONDing endpoint, the call is dropped. The drop reason *Bandwidth* appears on the Status Conference report. For this scenario, use the **status conference** command and provide the correct H.320 Meet-Me number to the H0 endpoint.

2B Participant Dials a BONDing Number

If a 2B participant tries to enter a conference by dialing a number intended for a BONDing endpoint, the call is cut through to the conference and joined only as a H.320 audio-only endpoint via BONDing transparent mode. For this scenario, use the **status conference** command. Entry **t** appears in the *Bond Mode* field. This indicates that BONDing transparent mode has been invoked.

BONDing Participant Dials an H0 Number

If a BONDing participant tries to enter an H0 conference by dialing a number intended for a H.320 H0 endpoint, the call is dropped. The drop reason *Bandwidth* appears on the Status Conference report. For this scenario, use the **status conference** command. If the BONDed participant can communicate at a 384k bandwidth, provide the correct H.320 Meet-Me number to join the BONDed conference.

BONDing Participant Dials a 2B Number

If a BONDing participant tries to enter a conference by dialing a number intended for a 2B endpoint, the call is dropped. The message “Handshake” appears in the report. For this scenario, use the **status conference** command and provide an appropriate H.320 Meet-Me number to the participant to join the BONDed conference.

BONDing Participant Dials a Multirate Number

If a BONDing participant tries to enter a conference by dialing a number intended for a H.320 multirate endpoint, the call is dropped. The drop reason *Bandwidth* appears on the Status Conference report. If the BONDed participant can communicate at a reserved bandwidth, provide a correct H.320 Meet-Me number to join the BONDed conference.

When All Else Fails

If you have performed the preceding procedures and you do not have the solution to the problem (or the recommended action is to escalate the problem), the Lucent Technologies Video Technical Center (VTC)/Customer Care Center or International Technical Assistance Center (ITAC) engineers are available to assist you. To contact the VTC or ITAC, follow these procedures:

1. Make sure you have both the Conference Appointment Form and the Port and Extension Usage Chart available.
2. Log in as **agent** to the MCU-ST.
3. Enter **status conference *conference-ID*** to display the Status of Conference screen.
4. Call either the Video Technical Center at **800 242-2121** or the International Technical Assistance Center at **303 804-3777**.
5. From the voice menu, select Video. A VTC or ITAC engineer will assist you.

Conference Command Summary

A

Overview

This appendix provides a summary of the following:

- Conference scheduling and troubleshooting commands.
- Conference control commands

Conference Scheduling and Troubleshooting Commands

The tables in this section include the relevant commands and actions available from the MCU Scheduling Terminal (MCU-ST).

Table A-1. Conference Scheduling and Troubleshooting Commands

Command	Description
list mcu-extension	Lists all the MCU-extensions that have been administered.
add mcu-extension xxxxx	Adds an MCU-extension. Used by telecommunications managers who should choose MCU-extensions that are consistent with the MCU's dial plan and trunks.
display mcu-extension xxxxx	Displays an MCU-extension.
change mcu-extension xxxxx	Changes the characteristics of an MCU-extension.
remove mcu-extension xxxxx	Deletes an MCU-extension.
add conference x	Adds a dedicated, reserved, or file conference.
change conference x	Changes the characteristics of a conference.
display conference x	Displays the conference record.
remove conference x	Removes a conference (file or reserved/completed only).
list conference	Lists the conferences administered on the MCU.
display available-ports 2B	Displays the number of ports available for 2B-channel 56k or 64k conferences over the next 24 hours.
display available-ports xxx	Displays the number of ports available for various ISDN multirate or BONDing conferences over the next 24 hours (for example, *display available-ports 384k, display available-ports B280/320,* etc.).
display available-ports T.120	Displays the number of data ports available for MCU conferences over the next 24 hours.
status conference x	Displays details about a conference that is active, in-use, or completed.

Table A-1. Conference Scheduling and Troubleshooting Commands — *Continued*

Command	Description
status conference x endpoint y	Displays details about a specific endpoint in a conference.
status to-conference x	Generates a conference termination warning tone and indicates to the endpoints how much time is left in the conference.
monitor conference x	Displays the status of a specified active conference in 30-second intervals. You can also display status on demand by pressing the “update” button.

Table A-2. Conference Class

Class	Description
dedicated	Conference is available continuously until it is changed to file.
reserved	Conference is scheduled to be available for a specified time period within the next 24 hours.
file	Conference is ignored by the MCU. Use this for recurring conferences and for recordkeeping.

Table A-3. Conference Status

Status	Description
active	Conference is available to receive calls.
inactive	Conference is not available to receive calls.
in-use	One or more conferees has joined the conference.
complete	A reserved conference has passed its stop time.

Table A-4. Conference Record Procedures

Action	On the Conference Record Screen
Stopping a dedicated or reserved conference	Change the *Class* to **file**. This stops the conference immediately.
Extending a reserved conference	Change the *Stop Time*.
Removing a dedicated or reserved conference	Change the *Class* to **file** and then remove it.

Conference Control Commands

Conference control commands can be issued from the MCU-MT to control MCU conferences. These commands are intended for troubleshooting purposes, and they are discussed in chapter 4.

Feature Descriptions

B

Overview

This appendix describes the primary Lucent Technologies MultiPoint Conferencing Unit (MCU) features and the availability of each feature.

The features in this section are arranged in alphabetical order, regardless of the functional area to which they apply. The information for each feature is presented under six headings: Description, Considerations, Interactions, Administration, and Hardware and Software Requirements.

- Description

  Defines the feature, tells what it does for the user or how it serves the system, and briefly describes how it is used.

- Considerations

  Discusses the applications and benefits of the feature, followed by the feature parameters and any other factors to be considered when the feature is used.

- Interactions

  Lists and briefly discusses other features that may significantly affect the feature being described. Interacting features are those that:

  — Depend on each other — one of the features must be provided if another particular feature is being used.

  — Cannot coexist — one of the features cannot be provided if the other feature is provided.

— Affect each other — the normal operation of one feature modifies, or is modified by, the normal operation of the other feature.

— Enhance each other — the features, in combination, provide improved service to the user.

- Administration

  States whether or not administration is required, how the feature is administered, who administers the feature, and lists items requiring administration.

- Hardware and Software Requirements

  Lists any additional hardware and/or software requirements needed for the feature.

NOTE:
Many of the Lucent Technologies MCU features and options are closely tied to administration and reports generation. This chapter is intended to provide an overview of the Lucent Technologies MCU features and options. For information concerning administration and reports, see Chapters 2 and 3.

Feature Summary

The following table provides a brief summary of the system features.

Table B-1. System Feature Summary

Feature	Category/Function
Audio Add-On (Integrated Audio Conferencing with Dynamic Echo Cancellation)	Multimedia Conferencing
Audio Modes	Multimedia Conferencing
Automatic Alternate Routing	Routing
Automatic Circuit Assurance	Identifying trunk problems (currently not used on the MCU)
Automatic Route Selection	Routing
Bandwidth On Demand Interoperability Group (BONDing)	Routing
Basic/Enhanced Service Flag	Multimedia Conferencing
BRI/DCP Direct Connect Interface	BRI or DCP connection directly to the MCU
Call Detail Recording	Detailed call information
Call-by-Call (CBC) Service Selection	Trunk service selection on a call-by-call basis (ISDN service)
Cascading	Multimedia Conferencing
Class of Restriction (COR)	Assigning restrictions to facilities
ClearPresence @716	Multimedia Conferencing
Conference Redial Flag	Automatic retry on conference dial failure
Conference Reservation and Control System (CRCS)	Multimedia Conferencing
Continuous Presence *Plus*	Enables a multipoint video conference to compose a picture consisting of the video from a maximum of 16 sites participating in the conference
DS1 Trunk Service	DS1 trunk interface
Dedicated Access	Allows non-signalling T1/E1 as well as EIA-530, RS449, and V.35 endpoints to join a conference
Dial-Out	Automatically calls all conference participants

Table B-1. System Feature Summary — *Continued*

Feature	Category/Function
Dial Plan	Enables trunk, station, and video port administration
Dynamic Resizing	Multimedia Conferencing
Executive Conferencing	Enables a conference to compose a picture of two or three “other” sites for each conference participant with no “self-view” (that is, no site ever sees itself).
Facility Test Calls (with Security Measures)	Provides for test calls from the maintenance alarm terminal
Facility and Non-Facility Associated Signaling	ISDN-PRI signaling
Far-End Camera Control (FECC)	Allows a site that requests far-end camera control to control the camera of the far-end site it is currently viewing
H.263/H.261 Transcoding	Video mode interworking capabilities
High-/Low-Speed Interworking	Allows lower speed endpoints to join higher speed conferences as audio-only endpoints
Integrated Services Digital Network (ISDN) — Primary Rate Interface (PRI)	Used for connecting the Lucent Technologies MCU to an ISDN-PRI trunk
ISDN-BRI Trunk Interface	Supports ISDN-BRI connections to the public network
Lip Synchronization	Conference audio delay option
Multirate Bandwidths	Larger bandwidths with ISDN interface
Networking/Call Handling	Connection of network calls to the appropriate video port
Notification Package	Includes the following: ■ Terminal names feature ■ Conference tones ■ Video-switching mode notification (tones and text) ■ Play tone command
On-Screen Display	Text and ANI code capabilities

Table B-1. System Feature Summary — *Continued*

Feature	Category/Function
OneNumber Access	Common external extension for multiple MCU endpoints
Passwords	Provides additional conference security with per-conference passwords.
Rate Adaptation	Adapts for mixed 56k/64k conferences
Recent Change History	History report generation
Report Scheduler and System Printer	Reports scheduling
RS366 Dialing	Dial string processing for endpoints connected to the MCU via data interfaces
Security Violation Notification	Notification of unauthorized attempt at system management access
Selected Communications Mode (SCM) Upgrades	Configures conferences as endpoints join or leave the conference
Service Flags	Provides for identification of interoperability concerns
Speed Match	Interworking of low-speed and high-speed endpoints
Still-Frame (H.261 Annex D)	Still-image transfer in the video channel
System Measurements	Traffic and efficiency report generation
System Status Report	Report generation
T.120 Data Conferencing	Multipoint data conferencing
Terminal Names	Provides endpoint identification during a conference
Uniform Dial Plan (UDP)	Shared dial plan among a group of Lucent Technologies MCUs or PBXs
Universal Conference Control (UCC)	Limited chair control for non-standard endpoints.

Table B-1. System Feature Summary — *Continued*

Feature	Category/Function
Video Conference Control Functions	Multimedia Conferencing
Video Modes	Video compression and transcoding
Web.Res.Now	Web-based conference reservation system for dedicated MCU conferences

Audio Add-On (Integrated Audio Conferencing with Dynamic Echo Cancellation)

Description

Audio Add-On (Integrated Audio Conferencing with Dynamic Echo Cancellation) enables a reservations agent to add up to 25 audio-only endpoints to a conference. In this way, an endpoint that is not H.320-compliant can join the audio portion of a conference.

Considerations

The maximum number of Audio Add-On ports per MCU is 32. The maximum number of Audio Add-On ports per conference is 25.

Interactions

Dynamic Resizing allows Audio Add-On only as an available audio port.

Administration

Audio Add-On is administered per system by the system administrator. For the Audio Add-On capability, the number of Audio Add-On ports must be administered. Audio Add-On ports are administered in increments of four.

Hardware and Software Requirements

The TN788B Voice Conditioner circuit pack is required.

Audio Modes

Description

For each conference, a preferred audio mode must be selected. A conference can be set up to require each endpoint to call the Lucent Technologies MCU (known as *dial-in* or *meet-me* setup), for the Lucent Technologies MCU to call each endpoint (known as *dial-out* setup) in turn, or for a combination of dial in and dial out. As each endpoint connects with the Lucent Technologies MCU for the conference, it automatically provides a capabilities list. The Lucent Technologies MCU, in turn, is administered on a per-conference basis for one of three preferred audio modes or for auto selection. In most cases, conferences are set for automatic selection of audio mode.

The possible audio modes are:

- G.722
- G.728 (LD-CELP) with data sharing
- G.711 (PCM)
- ClearPresence @716

Considerations

The following considerations are provided based on whether or not the ClearPresence @716 RTU is optioned.

ClearPresence @716 RTU Not Optioned

In auto selection, the Lucent Technologies MCU refers to the capabilities list of each endpoint in a highest common denominator (HCD) algorithm to determine the audio mode to use. First, the Lucent Technologies MCU attempts to use the audio mode requiring the smallest bandwidth G.722 or G.728 (LD-CELP) if the conference is also administered for data sharing) to provide greater video quality. If a connecting endpoint is not capable of G.722 or G.728 (LD-CELP) mode, G.711 pulse-code modulation (PCM) is selected and data sharing is disabled.

NOTE:
G.728 is the preferred mode in all 2-channel conferences. G.722 is the audio mode that allows the best video quality and is preferred in all other conference bandwidths.

If the endpoint capabilities list does not contain an audio mode common to the endpoints already connected and it is a single-channel endpoint attempting to connect to a 2-channel conference, the connecting endpoint is allowed an audio-only PCM connection to the conference. Except in the case of High-Speed/Low-Speed Interworking, any other mismatch (such as, a 112k

endpoint attempting to connect to a 384K (H0) conference) the endpoint is not allowed to join.

In conferences involving endpoints at higher rates than 64k, endpoints that cannot join with the administered bandwidth are connected as audio-only PCM endpoints.

ClearPresence @716 RTU Optioned

For details, refer to "ClearPresence @716 Audio Coder" on page B-63.

Interactions

The following considerations are provided based on whether or not the ClearPresence @716 RTU is optioned.

ClearPresence @716 RTU Not Optioned

The following features interact with the audio mode selection:

- High-Speed/Low-Speed Interworking.

  This feature allows mismatched bandwidths to join conferences as audio-only endpoints.

- T.120 Data Conferencing

  See "T.120 Data Conferencing" on page B-169 for details.

ClearPresence @716 RTU Optioned

For details, refer to "ClearPresence @716 Audio Coder" on page B-63.

Administration

Audio modes are administered via the Customer Options form and the Conference Record form. For general information, refer to the “Audio Mode” bullet item on page 1-12. For information based on the ClearPresence @716 RTU optioned, refer to "Administering ClearPresence @716 Audio Coder" on page 2-255.

Hardware and Software Requirements

Consult your Lucent Technologies service representative.

Automatic Alternate Routing

Description

Automatic Alternate Routing (AAR) provides alternative routing choices for private on-network calls. With AAR, the system automatically selects the most desirable (normally the least expensive) route over various trunking facilities for private network calls. AAR also provides digit modification to allow on-network calls to route through the public network when an on-network route is not available, and to convert incoming digits to MCU extensions in the Lucent Technologies MCU.

The private network of PBXs that utilizes the AAR feature is called an Electronic Tandem Network (ETN). An ETN is a hierarchical network of privately owned trunk and switching facilities that can provide a cost-effective alternative to toll calling between locations. An ETN consists of tandem switches, the inter-tandem tie trunks that interconnect them, the access or bypass tie trunks from a tandem switch to a main switch, and the capability to control call routing over these facilities.

Within an ETN, each switching facility is identified by a unique private network office code. Private network office codes may be one to eight digits in length. Throughout the rest of this description, the private network office code will simply be referred to as the “office code.”

ETN addresses for Uniform Dial Plan (UDP) destinations are limited to a seven-digit format. This means that the location code part for UDP is a three-digit code of the form RNX and the extension number is a four-digit number in the XXXX format (along with limitations that the UDP number cannot start with a 0). Note that five-digit UDP extensions are supported. For other destinations, ETN addresses are not limited to the seven-digit RNX format.

The principal use of AAR is to provide routing of private network calls. Private network calls originate and terminate at a customer location without accessing the public network. The normal scenario is as follows: the dial-out feature is used to dial the AAR access code followed by an on-network number. AAR then selects the route for the call and performs any necessary digit manipulation. AAR selects the most desirable route for the call. If the first choice route is not available, another route is chosen automatically. AAR provides up to six routes for each office code.

Feature operation is completely transparent to the user. The AAR access code is normally the digit 8. Normally, the called number is a private network number.

Private network (on-network) numbers are handled by the AAR feature. An on-network number can be changed into a public network direct-distance dialing number, a CDOS number, or an IDDD number by administering the “ars” call-type for such numbers.

The private network location codes may match public network central office codes. Therefore, the only way to determine the intended network for seven-digit calls is by the AAR or ARS access codes administered in the dial-out number(s) or by specific administration of the “ars” call-type on the AAR Analysis form.

AAR and Subnet Trunking provide a convenient means to place IDDD calls to a frequently called foreign city. Such calls route as far as possible over the private network before exiting the network. The office code is, of course, reserved to represent a particular country and city. At the final on-network switch, the office code is deleted. The international prefix code (011 in the US, 00 in most of Europe, and so on), the country code, and the city code are inserted. The inserted digits plus the last four digits of the originally dialed number constitute the IDDD number.

Similar to the IDDD case, certain domestic calls may reach a point on the network where they can route no further because tie trunks to the next switch are busy or none are provided. In this case, the office code can be deleted and the appropriate public network code inserted. Calls of this type route off-network via a central office. The central office may be connected to either an ETN tandem or main switch. Toll charges, if any, are from the final ETN switch to the destination.

Each office code can point to any one of several Routing Patterns, numbered 1 through a maximum limit of 254 for your system. More than one office code can point to the same pattern. A blank pattern provides intercept treatment and pattern 254 is the default for all office codes. Routing Patterns are shared with ARS. Access to a route within the pattern is controlled by Facilities Restriction Level (FRL) assignments. FRLs are fully described elsewhere in this chapter. For outgoing ISDN calls, route selection is dependent on Bearer Capability Class (BCC), FRL, and type of facility.

Digit Conversion

Once the AAR access code and the called number are dialed, the dialed number is compared to entries in the Matching Pattern fields of the AAR Digit Conversion Table screen. An example of this is shown in the following figure. If all or part of the dialed number matches one of the Matching Patterns on the screen, the dialed number is replaced by a new number from the Replacement String field on the screen. This new number is then used to route the call, the call becomes an ARS call, and is routed using the ARS Analysis Table. This function may be used to route specific dialed number strings to a different number, intercept, and so on. The Digit Conversion Table is only used once per call.

```
change aar digit-conversion 0                                Page  1 of  2

                          AAR DIGIT CONVERSION TABLE
                                                        Percent Full: 2

Matching Pattern     Min   Max   Del    Replacement String     Net  Conv ANI Req
_________________    __    __    __     _________________      __   ___  ___ n_
_________________    __    __    __     _________________      __   ___  ___ n_
_________________    __    __    __     _________________      __   ___  ___ n_
_________________    __    __    __     _________________      __   ___  ___ n_
_________________    __    __    __     _________________      __   ___  ___ n_
_________________    __    __    __     _________________      __   ___  ___ n_
_________________    __    __    __     _________________      __   ___  ___ n_
_________________    __    __    __     _________________      __   ___  ___ n_
_________________    __    __    __     _________________      __   ___  ___ n_
_________________    __    __    __     _________________      __   ___  ___ n_
_________________    __    __    __     _________________      __   ___  ___ n_
_________________    __    __    __     _________________      __   ___  ___ n_
_________________    __    __    __     _________________      __   ___  ___ n_
_________________    __    __    __     _________________      __   ___  ___ n_
_________________    __    __    __     _________________      __   ___  ___ n_
_________________    __    __    __     _________________      __   ___  ___ n_
_________________    __    __    __     _________________      __   ___  ___ n_
```

Figure B-1. AAR Digit Conversion Table

The purpose of AAR digit conversion is simply to convert private network numbers to public network numbers. This allows the system to change specific dialed digits to a public network number and route some calls via ARS. Also, unauthorized private network calls can be routed to receive intercept treatment.

AAR Analysis

After an AAR call passes through AAR Digit Conversion, AAR Analysis is performed.

The system uses AAR Analysis to compare the dialed number with entries in an AAR Analysis Table. When the system finds a dialed string entry in the table that matches the dialed number, the AAR Analysis Table maps the dialed number to a specific Routing Pattern (discussed later) and Call Type. The selected Routing Pattern and Call Type are then used to route the call. If the Call Type field on the AAR Analysis Table for a digit string is “ars,” call processing crosses over to ARS and the call is processed as an ARS call. The AAR Analysis Table screen form also shows the minimum and maximum number of digits required for digit analysis of each dialed number. An example of the AAR Analysis Table screen form is shown in following screen.

Dialed string entries may contain the letter ‘‘x’’ or ‘‘X,’’ which is used as a ‘‘wildcard’’ character. This wildcard character matches any of the digits 0 through 9. For example, a dialed string entry of ‘‘3x’’ applies to all calls beginning with 30 through 39. This ‘‘wildcard’’ makes it possible for traditional three-digit RNXs to be represented in several ways in the AAR Analysis Table. For example, RNXs 200 through 299 can be assigned to the AAR Analysis Table in either of the following ways:

Dialed String	Minium Number of Digits	Maximum Number of Digits
2	7	7
	or	
20	7	7
21	7	7
22	7	7
...	...	...
29	7	7
	or	
2xx	7	7
	or	
20x	7	7
21x	7	7
22x	7	7
...	...	...
29x	7	7

It is possible that some numbers may overlap other numbers. For example, the AAR Analysis Table may have dialed string entries of ‘‘645’’ and ‘‘6452.’’ In this case, for example, the number 645-2045 will be routed according to the 6452 entry (the longest dialed string).

When the UDP is used, the three-digit RNX dial string representation must be used on the AAR Analysis Table to match the administration in the UDP table.

Possible Call Types in the AAR Analysis Table are as follows:

- **aar** — Regular AAR call
- **ars** — Crossover to ARS call
- **haar** — Home ETN address (indicates that the call should be terminated locally on the home switch instead of routing to another ETN node. If the UDP is administered, then that dial plan is used to convert the number to a local extension number. Otherwise, the location code is deleted from the dialed number, and the remaining digits are used to route the call to a local extension).

The following figure shows the AAR Digit Analysis Table.

```
change aar analysis 1                                    Page  1 of  1


                          AAR DIGIT ANALYSIS TABLE
                        Partitioned Group Number: 1    Percent Full:6

     Dialed    Total Rte Call Nd ANI       Dialed    Total Rte Call Nd ANI
     String    Mn Mx Pat Type Num Rq       String    Mn Mx Pat Type NUM Rq
 2_____________ 7_ 7_ 254 aar   _  n     _____________ __ __ ___ ___   _   n
 3_____________ 7_ 7_ 254 aar   _  n     _____________ __ __ ___ ___   _   n
 4_____________ 7_ 7_ 254 aar   _  n     _____________ __ __ ___ ___   _   n
 5_____________ 7_ 7_ 254 aar   _  n     _____________ __ __ ___ ___   _   n
 6_____________ 7_ 7_ 254 aar   _  n     _____________ __ __ ___ ___   _   n
 7_____________ 7_ 7_ 254 aar   _  n     _____________ __ __ ___ ___   _   n
 8_____________ 7_ 7_ 254 aar   _  n     _____________ __ __ ___ ___   _   n
 9_____________ 7_ 7_ 254 aar   _  n     _____________ __ __ ___ ___   _   n
 _____________ __ __ ___ ___   _  n     _____________ __ __ ___ ___   _   n
 _____________ __ __ ___ ___   _  n     _____________ __ __ ___ ___   _   n
 _____________ __ __ ___ ___   _  n     _____________ __ __ ___ ___   _   n
 _____________ __ __ ___ ___   _  n     _____________ __ __ ___ ___   _   n
```

Figure B-2. AAR Digit Analysis Table

Some special dialing patterns are automatically mapped to a specific Routing Pattern and Call Type. These special dialing patterns, shown in the following table, are preassigned and require no administration. However, they can be changed by the system administrator.

Table B-2. AAR Analysis Default Translations

Dialed String	Number of Digits		Route Pattern	Call Type
	Minimum	Maximum		
0	1	23	—	ars
1	4	23	—	ars
2	7	7	254	aar
3	7	7	254	aar
4	7	7	254	aar
5	7	7	254	aar
6	7	7	254	aar
7	7	7	254	aar
8	7	7	254	aar
9	7	7	254	aar
X11	3	3	—	ars

Normally, the **Route Pat** (routing pattern) field on the Analysis Table screen contains a routing pattern number (1 through 254). However, this field may instead contain a Remote Home Numbering Plan Area (RHNPA) table number (r1 through r32). When an AAR Analysis table points to an RHNPA table, the next three dialed digits are compared with codes in the selected RHNPA table. Each code on the table is then mapped to a specific routing pattern number (1 through 254).

Routing Patterns

The digit translations performed on an AAR call by the AAR Analysis and RHNPA tables cause a specific Routing Pattern to be selected for the call. The Routing Patterns are numbered 1 through 254. More than one combination of dialed digits can point to the same pattern. A blank entry instead of a Routing Pattern number provides intercept treatment. However, with AAR, digit translation should always point to a Routing Pattern. If calls to some numbers are to be denied, this should be handled by FRL assignment, not by intercept on the codes. FRLs are discussed elsewhere in this chapter.

The Routing Pattern applicable for a given call contains a list of up to six trunk groups that can be used for the call. Trunk group access is controlled by FRLs. The digit manipulation necessary to route the call is controlled by the Subnet Trunking feature. Otherwise, the digit string to be outpulsed is as dialed by the user or as converted by AAR Digit Conversion.

Considerations

AAR provides efficient use of private network facilities.

ARS and AAR Analysis tables together can have up to 2,000 entries.

If a customer changes AAR routing assignments, it is the customer's responsibility to notify the RSC network designer and the SCO technician of the changes to receive their continued support.

Digit deletion/insertion can be used to route calls to the MCU.

Internal memory resources used for AAR Digit Analysis are shared by ARS, AAR, Digit Conversion, and Toll Analysis features. A Percent Full field on the ARS and AAR Digit Analysis screens indicates how many of these resources have been used.

Interactions

The following features interact with the AAR Feature.

- ARS

  ARS and AAR can access the same trunk groups and share the same Routing Patterns and RHNPAs. Also, AAR calls can be administered to cross over to ARS via digit analysis and digit conversion.

- CDR

  An AAR call using a trunk group marked for CDR is indicated by the dialed access code and by a Condition Code. The dialed number is recorded as the called number.

  Subnet Trunking does not affect CDR. The dialed digits are recorded, not the outpulsed digits.

  The originating FRL associated with the call is recorded. However, if 15-digit CDR account codes are used, the FRL value is overwritten.

  If CDR generation is administered for a trunk group assigned to a Routing Pattern, data will be collected for all calls routed through the trunk group. If an CDR account code is to be dialed with an AAR call, it must be dialed before the AAR access code is dialed.

- UDP

  When a UDP number is dialed (four or five digits), the routing software initially converts the dialed number to a seven-digit format. The location code in the UDP table equates to a three-digit RNX plus four digits. In a four-digit UDP, the number created is the RNX plus the four extension digits originally dialed. In a five-digit UDP, the number created is the RNX plus the last four extension digits dialed.

UDP destinations are limited to a seven-digit format. This means that the location code part for UDP is a three-digit code of the form RNX and the extension number is a four-digit number in the XXXX format (along with limitations that the UDP number cannot start with a zero).

When the UDP is used, the three-digit RNX dial string representation must be used on the AAR Analysis Table to match the administration in the UDP table.

Administration

AAR is initially assigned on a per-system basis by a Lucent Technologies service technician. After the feature is activated, the following items are administered by either the system administrator or the service technician:

- AAR Access Code (one to three digits)
- AAR Analysis Table
- AAR digit conversion table Up to 32 RHNPA Tables
- Up to 254 Routing Patterns
- FRLs — Assigned via Class of Restriction to each originating facility, authorization codes, and barrier codes
- Trunk Groups to be used with AAR

Hardware and Software Requirements

AAR may require additional tie trunks. These additions are, however, cost effective when compared to the other alternatives for call routing.

Automatic Circuit Assurance

Description

Automatic Circuit Assurance (ACA) assists users in identifying possible trunk malfunctions. The system maintains a record of the performance of individual trunks relative to short and long holding time calls. The system automatically initiates a referral call to a maintenance alarm terminal when a possible failure is detected.

This feature is currently not used on the MCU.

Automatic Route Selection

Description

The Automatic Route Selection (ARS) feature routes calls over the public network based on the preferred (normally the least expensive) route available at the time the call is placed.

ARS provides a choice of up to six routes for any given public network call. The following types of trunk groups can be accessed by ARS:

- **Tie trunks** — Used to provide access to an ETN or public network. In some cases, it is preferable to allow a private network to handle the routing of calls destined for the public network.
- **ISDN-PRI** — Used for calls over an ISDN and provides users access to a variety of switched nodal services such as the ACCUNET. digital service and allows access to other inter-exchange carriers or private networks.

A variety of numbers can be called using ARS, including seven-digit numbers, 10-digit numbers, International Direct Distance Dialing (IDDD) numbers, service codes, Customer-Dialed Operator-Serviced (CDOS) numbers (for example, in the United States, 0+ or 01+), and Inter-Exchange Carrier (IXC) numbers.

ARS Dialing

The dial-out feature uses ARS for outgoing calls. ARS is initiated if the ARS access code (normally the digit 9) is included in the first digit of the dial-out number(s) administered by the reservations agent.

Special Dialing Patterns

The system recognizes certain dialing patterns on outgoing calls and routes these calls accordingly. The descriptions of these dialing patterns reflect the system defaults as used in the United States. Other countries may require different administration of these values in the dial-out numbers. The following dialing patterns are recognized:

- **DDD Calls With Prefix Digit 1 Required**

  The dial-out number may or may not be required to contain a 1 before a seven- or 10-digit dial out string, depending on the system's dial plan administration. There are two cases where the digit 1 must be included:

  — Some metropolitan areas are so densely populated that there simply are not enough traditional central office codes. Therefore, it is possible that some NPA codes, also called ''area codes'' may also serve as CO codes. In this case, the digit 1 must be included if a 10-digit call is intended. The first digit tells the system whether to route the call as a seven-digit call within the home NPA (1 not included) or as a 10-digit call to another NPA (1 included). In this case, the dial plan should be administered so that the 1 is included for 10-digit calls.

  — Digit 1 may also be required in areas near an NPA boundary. In these areas, certain calls to the adjacent NPA may be local calls rather than toll calls. However, central office codes may be duplicated in the home and adjacent NPAs. Also a CO code in the home NPA may be a toll call. Therefore, if the digit 1 is not required on certain adjacent NPA local calls, then it must be included on the home NPA seven-digit toll calls so the system can differentiate between the intended destinations.

- **DDD Calls with Prefix Digit 1 Not Required**

  The first digit following the ARS access code may or may not be a 1. In systems where the 1 prefix is used, but not required (as administered on the Dial Plan form), using the 1 prefix before a 10-digit call is optional and the prefix is ignored.

- **IDDD Calls**

  IDDD numbers consist of a Country Code and a National Number. The National Number is simply the number used when calling within the country. The Country Code can be from one to three digits in length. In the NANP the National Number is 10 digits in length. The Country Code and National Number together cannot exceed 12 digits. In the NANP, international numbers are recognized by special prefix codes:

  — **011** — Indicates that dial out is making a station paid direct international call. The Country Code and National Number follow the 011 prefix.

- **Calls Dialed with Inter-Exchange Carrier (IXC) Access**

  The first digits following the ARS access code are an IXC Access Code. The access code may be followed by a DDD or an IDDD number. If included in the dial-out number(s), it selects the carrier or facilities used for routing the call.

Digit Conversion

Once dial out uses the ARS access code and the called number, the dialed number is compared to entries in the Matching Pattern fields of the ARS Digit Conversion Table screen. An example of this screen is shown in the following figure. If all or part of the dialed number matches one of the Matching Patterns on the form, the matching part of the dialed number is replaced by a new number from the **Replace** field on the form. This new number routes the call via AAR over a private network. If no corresponding entry is found in the AAR Digit Analysis table, the ARS Digit Analysis table is searched for a match with the new modified number and routed accordingly. If the call fails and a match is not found, an intercept tone is sounded.

```
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                        ARS DIGIT CONVERSION TABLE
                                                    Percent Full: 2

Matching Pattern     Min   Max   Del    Replacement String     Net  Conv ANI Req

_________________     __    __    __    _________________      _   ____  ___ n_
_________________     __    __    __    _________________      _   ____  ___ n_
_________________     __    __    __    _________________      _   ____  ___ n_
_________________     __    __    __    _________________      _   ____  ___ n_
_________________     __    __    __    _________________      _   ____  ___ n_
_________________     __    __    __    _________________      _   ____  ___ n_
_________________     __    __    __    _________________      _   ____  ___ n_
_________________     __    __    __    _________________      _   ____  ___ n_
_________________     __    __    __    _________________      _   ____  ___ n_
_________________     __    __    __    _________________      _   ____  ___ n_
_________________     __    __    __    _________________      _   ____  ___ n_
_________________     __    __    __    _________________      _   ____  ___ n_
_________________     __    __    __    _________________      _   ____  ___ n_
_________________     __    __    __    _________________      _   ____  ___ n_
_________________     __    __    __    _________________      _   ____  ___ n_
_________________     __    __    __    _________________      _   ____  ___ n_
_________________     __    __    __    _________________      _   ____  ___ n_
```

Figure B-3. ARS Digit Conversion Table

Digit analysis implies both route and conversion. The algorithm for deciding how to analyze a call is as follows:

- number of digits
- most digits matched
- explicit digits over wildcards (from left to right)
- conversion nodes over route nodes

The primary purpose of ARS digit conversion is to convert public network numbers (to be routed with ARS) to private network numbers (to be routed with AAR).

NOTE:
You can convert back to another AAR number or an extension, where the extension most likely is a UDP.

This can save toll charges and allow the reservations agent to route calls on the customer's private network facilities. Also, unauthorized public network calls can be blocked and routed to a receive intercept treatment via digit conversion.

The following table shows several ARS Digit Conversion examples. The following conditions are assumed for the examples: ARS Access Code = 9, AAR Access Code = 8, Home RNX (Private Network Office Code) = 222, Prefix 1 is required on all long-distance DDD calls, Dashes "-" shown in the table are for readability only.

Table B-3. ARS Digit Conversion Examples

Operation	Actual Digits Dialed	Matching Pattern	Replace-ment String	Modified Address	Notes
DDD call to ETN	9-1-303-538-1345	1-303-538	362	362-1345	The call will be routed via AAR on the route selected for RNX 362.
Unauthorized call to intercept treatment after 976 are ignored by ARS. The call receives intercept treatment.	9-1-212-976-1616	1-XXX-976	#	(blank)	The "#" signifies the end of dialing. Any digits dialed after 976 are ignored by ARS. The call receives intercept treatment.

NOTE:
The dial-out number digits are matched to the Matching Pattern that most closely matches them. For example, if the dial-out number is 957-1234 and matching patterns 957-1 and 957-123 are in the table, the match is on pattern 957-123. The call will be routed as dialed.

ARS Digit Analysis

ARS calls that pass through ARS Digit Conversion and Toll Analysis are analyzed based on the dial-out PGN.

The system uses ARS Digit Analysis to compare the dialed number with entries in an ARS Digit Analysis Table. When the system finds a Dialed String entry in the table that matches the dialed number, the ARS Digit Analysis Table maps the dialed number to a specific Routing Pattern, discussed later in this chapter, and Call Type. The selected Routing Pattern will then be used to route the call. The ARS Digit Analysis Table screen also shows the minimum and maximum number of trailing digits required for digit analysis of each dialed number. An example of the ARS Digit Analysis Table follows.

Possible Call Types in the ARS Digit Analysis Table are as follows:

- **fnpa** — 10-digit call within North America
- **hnpa** — 7-digit call within North America
- **int** — International call (in the United States, the international prefix is 011)
- **natl** — National numbers within a country (used outside North America)

```
change ars analysis 0                                    Page  1 of  1


                              ARS DIGIT ANALYSIS TABLE
                          Partitioned Group Number: 1    Percent Full: 6

     Dialed      Total Rte Call Nd ANI     Dialed      Total Rte Call Nd ANI
     String      Mn Mn Pat Type Num Rq     String      Mn Mn Pat Type Num Rq
 0_____________ 1  1  den op     _  n  _____________ _  _  ___ __     _  n
 0_____________ 8  8  den op     _  n  _____________ _  _  ___ __     _  n
 0_____________ 11 11 den op     _  n  _____________ _  _  ___ __     _  n
 00____________ 2  2  den op     _  n  _____________ _  _  ___ __     _  n
 01____________ 9  17 den iop    _  n  _____________ _  _  ___ __     _  n
 011___________ 10 18 den intl   _  n  _____________ _  _  ___ __     _  n
 101xxxx0______ 8  8  den op     _  n  _____________ _  _  ___ __     _  n
 101xxxx0______ 18 18 den op     _  n  _____________ _  _  ___ __     _  n
 101xxxx01_____ 16 24 den iop    _  n  _____________ _  _  ___ __     _  n
 101xxxx011____ 17 25 den intl   _  n  _____________ _  _  ___ __     _  n
 10xxxx1_______ 18 18 den fnpa   _  n  _____________ _  _  ___ __     _  n
 10xxx0________ 6  6  den op     _  n  _____________ _  _  ___ __     _  n
 10xxx0________ 16 16 den op     _  n  _____________ _  _  ___ __     _  n
 10xxx01_______ 14 22 den iop    _  n  _____________ _  _  ___ __     _  n
 10xxx011______ 15 23 den intl   _  n  _____________ _  _  ___ __     _  n
```

Figure B-4. ARS Digit Analysis Table

Some special dialing patterns are automatically mapped to a specific Routing Pattern and Call Type. These special dialing patterns, shown in the following table, are preassigned and require no administration. They can, however, be changed by the system administrator and probably will be changed for PBXs used outside North America.

Table B-4. ARS Digit Analysis Default Translations

	Trailing Digits			
Dialed String	**Minimum**	**Maximum**	**Pattern**	**Route Type**
011	10	23		int
1	11	11		fnpa
10XXX011	15	23		int
2	7	7	1	hnpa
3	7	7	1	hnpa
4	7	7	1	hnpa
5	7	7	1	hnpa
6	7	7	1	hnpa
7	7	7	1	hnpa
8	7	7	1	hnpa
9	7	7	1	hnpa

Legend:
fnpa — foreign number plan area (10-digit call)
hnpa — home number plan area (7-digit call)
int — international
X — any digit (0-9)

Normally, the Route Pat (routing pattern) field on the ARS Digit Analysis Table screen contains a routing pattern number of 1 through 254. However, this field may instead contain a Remote Home Numbering Plan Area (RHNPA) table number (r1 through r32). An RHNPA is simply a concentrator for up to 1,000 calls. Calls are routed to these tables by ARS/AAR Digit Analysis when an ARS Digit Analysis table points to an RHNPA table. The next three dialed digits (the code) are compared with the codes in the selected RHNPA table. Each code on the table is then mapped to a specific routing pattern number (one through 254).

The RHNPA tables allow up to 1000 codes to be handled by one entry in the Digit Analysis table.

In summary, Digit Analysis is merely a method of selecting a routing pattern. The routing pattern may be selected in two ways:

- It may be selected directly from the Digit Analysis table.
- The Digit Analysis table may first have to select an RHNPA table that will, in turn, select the routing pattern.

Routing Patterns

The digit translations performed on an ARS call by the Digit Analysis and RHNPA tables cause a specific Routing Pattern to be selected for the call. The Routing Patterns are numbered 1 through 254. More than one combination of dialed digits can point to the same pattern. A blank entry instead of a Routing Pattern number provides intercept treatment. However, with ARS, digit translation should always point to a Routing Pattern. This way, calls to unassigned office codes will be intercepted by the central office, not by the system. By allowing the unassigned codes to be intercepted by the central office, the system administrator does not have to keep track of which office codes are in service. If calls to some codes are to be denied, this should be handled by FRL assignment, not by intercept on the codes. FRLs are discussed elsewhere in this chapter.

The Routing Pattern applicable for a given call contains a list of up to six trunk groups that can be used for the call. Trunk group access is controlled by FRLs. If access to the public network is through a main switch (an Access trunk group is selected for the call), then the call will route through the main switch to one of the public network offices serving the main switch. The digit manipulation necessary to route the call is controlled by the Subnet Trunking feature. Otherwise, the digit string to be outpulsed is controlled by ARS. ARS digit manipulation is called code conversion. Code conversion is used to determine whether or not to outpulse the digit 1 on toll calls and whether to insert, keep, or delete the NPA on toll calls.

The following paragraphs describe how the switch decides what digits to outpulse in specific situations.

Digit 1 Outpulsing

The digit 1 may or may not be required at the public network office to which the call will be routing. If 1 is dialed on 7-digit calls at a stand-alone system (non-ETN), the 1 is outpulsed by the system. In the other cases, the 1 outpulsing requirements are indicated in the system. Since any given call may have a choice of up to six routes, some of which may require a 1 and some of which may not, this indication is associated with each route. Five choices are available and are identified in translations by a Prefix Mark. Digit 1 outpulsing only applies to calls administered as ‘‘fnpa’’ or ‘‘hnpa’’ in the ARS Digit Analysis table. The values and meanings of the Prefix Marks are as follows:

- Prefix Mark 0 — Suppress a dialed Prefix digit 1 for 10-digit FNPA calls, but leave a Prefix digit 1 for the following types of calls:
  - — 10-digit calls that are not administered as FNPA or HNPA types in the ARS Routing Table.
  - — 7-digit HNPA calls

- Prefix Mark 1 — Send a 1 on 10-digit calls, but not on 7-digit calls.
- Prefix Mark 2 — Send a 1 on all toll calls (for example, all 10-digit calls and 7-digit toll calls).
- Prefix Mark 3 — Send a 1 on all toll calls and keep or insert the NPA to ensure that all toll calls are 10-digit calls. Note that a dialed Prefix digit 1 for a 7-digit call makes it a toll call and, hence, NPA is also inserted in this case.
- Prefix Mark 4 — Always suppress a Prefix digit 1.

NOTE:
This capability is required, for example, when routing ISDN-PRI calls to a Lucent Technologies 4ESS. If the prefix digit 1 were not suppressed, then the 4ESS would reach calls.

Which of the five possible treatments of the 1 prefix digit should be administered on a given route is based on the characteristics of the distant office. Prefix Mark 0 prevents the system from sending a 1 prefix digit for 10-digit FNPA calls. However, the system leaves a user-dialed prefix digit 1 for 7-digit HNPA calls and 10-digit calls that are not administered as FNPA or HNPA types in the ARS Routing Table.

Prefix Mark 1 causes the system to send a 1 prefix on all 10-digit FNPA calls.

With Prefix Marks 2 and 3, the decision is based on whether the call is a toll call. Toll Lists are provided in the system to furnish this information. A Toll List simply indicates if the office code associated with the call constitutes a toll call from the interconnecting office (not from the local system). Up to 32 Toll Lists are provided. The applicable list number, if any, for the call is given in the Routing Pattern.

Prefix Marks are only applicable on 7- or 10-digit DDD public network calls. Requirements for outpulsing a 1 are specified via Prefix Marks and go into effect when the call accesses is outpulsed. Digit 1 outpulsing only applies to calls administered as ‘‘fnpa’’ or ‘‘hnpa’’ in the ARS Digit Analysis table.

NPA Deletion and Insertion

Each public network route in the ARS Routing Pattern contains an indication of the NPA of the distant end of the trunk group. If this NPA is the same as the NPA associated with the call, the NPA is deleted prior to outpulsing unless the Prefix Mark is 3 and the call is a toll call in the associated Toll List.

NPA deletion and insertion only applies to calls administered as ‘‘fnpa’’ or ‘‘hnpa’’ in the ARS Digit Analysis table.

The NPA is inserted on 7-digit calls if the distant NPA is different from the home NPA or if the Prefix Mark is 3 and the call is a toll call in the associated Toll List.

The preceding paragraphs describe NPA deletion or insertion when the call is an ARS 7- or 10-digit DDD call. An ARS call accessing a tandem trunk is another example of NPA insertion. If the call is a 7-digit ARS call, the system inserts the home NPA before sending the call to the tandem trunk. Therefore, all ARS calls accessing a tandem trunk are 10-digit calls. Whether or not the digit 1 is sent on a tandem call is determined by the prefix rules. This enables the system to distinguish between ARS calls and the 7-digit on-network calls.

IDDD and Service Code Dialing

ARS can provide individual Routing Patterns for each type of call. An ARS call can be processed via the RHNPA table. This is particularly useful on international calls, since the RHNPA table can be used on the country code. Thus, call routing can be determined according to the called country, rather than handling all international calls alike.

Considerations

ARS provides the most-preferred usage of public network facilities available at a system.

Up to 254 Routing Patterns, shared with AAR, can be provided.

Up to 32 Toll Lists can be provided.

Up to 32 RHNPA tables, shared with AAR, can be provided.

ARS and AAR Digit Analysis tables together with the Toll Analysis table can have up to 2,000 entries.

Internal memory resources used for ARS Digit Analysis are shared by ARS, AAR, Digit Conversion, and Toll Analysis features. A Percent Full field on the ARS and AAR Digit Analysis screens indicate how many of these resources have been used.

If a customer changes ARS routing assignments, it is the customer’s responsibility to notify the Regional Support Center (RSC) network designer and the System Control Office (SCO) technician of the changes to receive their continued support.

Interactions

The following features interact with the ARS feature.

- AAR

  ARS and AAR can access the same trunk groups and share the same Routing Patterns, toll lists, and RHNPA tables. ARS calls may be converted to AAR calls.

- GRS

  Generalized Route Selection (GRS) works with ARS to provide call routing over the appropriate trunking facilities. Routing is determined by the type of call being made. With GRS, calls may be routed differently than they would with justARS. For details on GRS, see the Generalized Route Selection feature description elsewhere in this chapter.

- CDR

  An ARS call using a trunk group marked for CDR is indicated by the dialed access code and by a Condition Code. The dialed number is recorded as the called number. Subnet Trunking does not affect CDR.

  If CDR generation is administered for a trunk group assigned to a Routing Pattern, data will be collected for all calls routed through the trunk group.

Administration

ARS is initially assigned on a per-system basis by a Lucent Technologies service technician. After the feature is activated, the following items are administered by the system administrator:

- ARS Access Code 1 (one to three digits)
- ARS Digit Analysis Table
- Up to 254 ARS Routing Patterns
- Trunk Groups to be used with ARS
- Whether or not the system returns dial tone after the ARSFAC is dialed on trunk calls

Hardware and Software Requirements

ARS may be used on a stand-alone system or may be an integral part of a private network. No additional hardware is required for a stand-alone system. A private network may require additional tie trunks and TN744 Tone Detector circuit packs. These additions are, however, cost effective when compared to the alternatives for call routing.

BONDing

Bandwidth on demand interoperability group (BONDing) is a standardized form of inverse multiplexing that is an alternative for gaining higher bandwidths between 112k and 768k without the use of ISDN-PRI multirate or H0 channels.

Description

As an alternative to the special hardware and software required for ISDN-PRI for higher bandwidths, an ANSI BONDing inverse multiplexing standard has been established. There are three BONDing modes outlined in the BONDing standard:

- Mode 1

  Provides user data rates in multiples of the bearer rate (56k/64k). This is the BONDing mode currently supported by the Lucent Technologies MCU.

- Transparent mode

  The default mode used when BONDing framing is not detected by a specific timeout period.

- Mode 3

  Includes an in-band monitoring function. Uses 7 B-channels instead of 6 B-channels for 384k. The Lucent Technologies MCU does not currently support this mode.

Considerations

To ensure proper operation of the BONDing link, the first call must be successfully connected before the Lucent Technologies MCU connects the remaining aggregated calls.

H0 calls can be placed to a 384k BONDed conference; however, the endpoints placing the calls must be translated within the conference as Px64 type endpoints.

One-channel calls placed to a BONDed MCU conference enabled with Interworking are added as audio-only calls.

Within a fixed BONDed environment only, all channels in a BONDed call are required to terminate on the same TN787K circuit pack; this may affect the system port capacity. This not true within a flexible (non-fixed) environment, which is typical. The type of environment in effect depends on the BONDed rate in effect and the number of BONDing ports used. For details, refer to Chapter 6 of the Installation and Test document.

Interactions

The following features interact with the BONDing feature.

- Low-/High-Speed Interworking

  BONDed ports cannot coexist with interworking ports in the same conference.

- Audio Add-On

  A maximum of 25 Audio Add-On endpoints are allowed for each BONDed conference.

- Cascading

  The inter-Lucent Technologies MCU link must be specified on the Conference Record as a specific BONDed bandwidth cascade. If transparent mode is invoked in the cascade link, the Lucent Technologies MCUs connected by the link do not share video, and the participants on one Lucent Technologies MCU are added to the conference as audio-only endpoints. If a bandwidth between 112k and 336k is specified for the link, the cascade must join the conference at a BONDed mode1 bandwidth of the same bandwidth or as a BONDed transparent mode link. If 384k is specified for the link, the cascade must join the conference at an H0 bandwidth, a BONDed mode 1 384k bandwidth, or as a BONDed transparent mode link.

- Dial-Out

  The Dial-Out feature dials one channel first and then all remaining channels simultaneously when an endpoint is designated as a BONDed endpoint. The calls can all be the same number or all different numbers. Different numbers are supplied from the endpoint equipment through the BONDing protocol.

- Dynamic Resizing

  To join an active H0 conference, a BONDed endpoint may only join if it is running at the same transfer rate.

Administration

The following administration is required for BONDing.

- Conference Record form

  Identify extensions that correspond to BONDing conferees.

- Availability of Ports form

  Use the BONDing option to identify the available BONDing ports.

- Optional Features form

  Enter the number for maximum BONDing port capacity.

Hardware and Software Requirements

Release 2.0 or later of the Lucent Technologies MCU is required for BONDing. The TN787K, vintage 5 or greater Multimedia Interface circuit pack is required for BONDed calls.

Basic/Enhanced Service Flag

Description

There are two user-selectable flags you can use for identifying a Lucent Technologies MCU to endpoint interface.

- Basic/enhanced service
- Application compliance

The basic/enhanced service flag is set to identify endpoints that can use the full set of conferencing messages. Some endpoints do not operate properly when other endpoints (or the MCU) attempt to use the full set of messages. If an endpoint cannot operate with the full message set, set the basic/enhanced service flag to basic.

The application compliance flag identifies whether endpoints in a conference can operate properly with T.120 Data Conferencing applications.

Considerations

None.

Interactions

Refer to the individual feature sections for details.

Administration

Translations must be performed to use this feature.

Hardware and Software Requirements

Only release 3.0 or later versions of the Lucent Technologies MCU can provide the service and compliance flags.

BRI/DCP Direct Connect Interface

Description

This feature allows the user to connect Basic Rate Interface (BRI) or Digital Communications Protocol (DCP) endpoints directly to the MCU without involving a public or private network, PBX, or MUX. A maximum of 36 Direct Connect endpoints can be administered.

NOTE:
In addition to BRI/DCP, several data interfaces allow endpoints to connect to the Lucent Technologies MCU. For details, refer to the “Data Interfaces for Access Endpoints” section in this appendix.

Considerations

BRI Direct Connect endpoints support data only. Voice capabilities are not provided via the BRI interface.

The endpoints directly connected to the MCU via BRI or DCP can participate in point-to-point conferences as well as multipoint conferences through the MCU.

The feature reduces the system port capacity. Therefore, for example, a 16-port BONDed system, which is a maxed-out system, cannot support BRI/DCP Direct Connect endpoints. Also, CRCS supports the feature but deducts resources from its calculated port capacities.

Interactions

All H.320 and T.120 features supported on the Lucent Technologies MCU are supported by this feature. However, refer to the last paragraph in the previous section for some important notes.

Administration

DCP Direct Connect endpoints are administered in increments of four with a maximum capacity of four and a default value of zero. BRI Direct Connect endpoints are administered in increments of 12 with a maximum capacity of 36 and a default value of 0. *A maximum of a combination of 36 of these two endpoint types can be active at one time*.

The following list indicates the fields in the Customer-Options form that you must populate along with the possible values up to a combined maximum total of 36.

- *Max. BRI Direct Connect Port Capacity:* **0**, **12**, **24**, or **36**
- *Max. DCP Direct Connect Port Capacity:* **0** or **4**

Also, the *Type* field in the Processor Interface Data Module form must be populated. To provide ISDN-BRI connectivity via this field, enter value **7500**. Otherwise, enter value **procr-intf**.

NOTE:
The feature can be administered as blocked or non-blocked. For assistance, contact your Lucent Technologies representative.

Hardware and Software Requirements

This feature is supported between an MCU running R3.0 (or greater) software and H.320 endpoints.

The TN556 ISDN Line circuit pack and the TN778 Packet Control circuit pack are required in every system that contains BRI Direct Connect endpoints. A maximum of three of each of these circuit pack types is supported. Also, a maximum of two TN754 Digital Line circuit packs is supported; the second circuit pack is added to allow for DCP Direct Connect capability.

Endpoints that are directly connected to the MCU via BRI terminate on the TN556 circuit pack. Endpoints that are directly connected to the MCU via DCP terminate on the TN754 circuit pack. The High-Speed Link (HSL) is the data module used to connect endpoints to the MCU directly via DCP. For 112k and 128k connections, two HSLs connect to the MCU per endpoint.

Call Detail Recording

NOTE:
Call Detail Recording (CDR) is also referred to as Station Message Detail Recording (SMDR).

Description

The CDR feature records detailed call information on all incoming and outgoing calls on specified trunk groups and sends this information to a CDR output device. The CDR output device provides a detailed printout that can be used by the telecommunications manager or system administrator to compute call costs, allocate charges, analyze calling patterns, detect unauthorized calls, and keep track of unnecessary calls.

For two-channel conferences, two records with identical Conference Billing ID fields are stored. For multirate conferences, a single record is stored on a per-call basis. For BONDed conferences, a record for each B-channel with identical Conference Billing ID fields is stored.

Call detail information is provided on trunk groups that are administered for CDR. CDR provides detailed call information for the following types of calls:

- Outgoing Calls — calls originated by Administered Connection/MCU extension, going out on a trunk group.
- Incoming Calls — calls incoming on a trunk group and terminating at an MCU extension.
- Ineffective Call Attempt — (a) unavailable incoming or outgoing trunks due to trunk usage allocation for ISDN Call-By-Call Service Selection trunks and (b) incoming calls rejected by the MCU due to network specific feature (NSF) mismatch.

You have the option of turning off CDR generation for specific trunk group(s).

NOTE:
Some call accounting systems do not support all the call information offered by CDR. See your account executive for details.

Resource Limitation on CDR Records

If new calls come in when the CDR link is down and the buffer is full (the MCU has a 180 record maximum), the earliest records are overwritten.

Set Time and Date

The system clock must be set for daylight savings time (if appropriate) when the time changes. Changing the time and date ensures that CDR records have the correct time and date for the records being kept. The time and date can be changed using the MCU management terminal (MCU-MT).

NOTE:
If the time is changed while calls are in progress, the actual call durations for these calls are not reflected in the CDR record. A ‘‘9999’’ gets output in the CDR duration field. We recommend that you not make date and time changes when there are active calls.

CDR Data Formats

This part covers two types of formats sent to the CDR output device: date record and call detail formats.

Date Record Format

The date record format is shown in the following table.

Table B-5. Date Record Format to Printer and Expanded Version

ASCII Character Position	Data Field Description
01-02	Month*/Day* or Day*/Month*
03	Space
04-05	Day*/Month* or Month*/Day*
06	Carriage Return
07	Line Feed
08-10	null

* Leading zero added if needed. Also, depending on the CDR format, the month and day as they appear in the Feature-Related System Parameters form may be reversed.

Call Detail Record Format

The call detail record format provides detailed information concerning a call. The first table that follows summarizes the DS1 robbed bit CDR direct output from the system to the printer. The second table summarizes the 24-word ISDN expanded CDR record format. Call detail records are generated during processing of the call and are sent to the CDR output device in ASCII format when the call is disconnected.

Even though the MCU outputs the record in ASCII character representation, the terminology “word’’ is used here to mean the ASCII characters.

Table B-6. DS1 RB CDR Direct Output Format

ASCII Character Position	Data Field Description
01	Time Hour (tens)
02	Time Hour (units)
03	Time Minute (tens)
04	Time Minute (units)
05	Space
06	Duration Hour
07	Duration Minute (tens)
08	Duration Minute (units)
09	Duration Minute (tenths)
10	Space
11	Condition Code
12	Space
13-15	Access Code Dialed*
16	Space
17-19	Access Code Used*
20	Space
21-35	Dialed Number*
36	Space
37-41	Calling Number*
42	Space
43-57	Conference Billing Identification information*
58	Space
59-65	Authorization Code
66-69	Space
70	FRL
71	Space
72	IXC
73	Space
74-76	Incoming Circuit† (hundreds, tens, units)
77	Space

Table B-6. DS1 RB CDR Direct Output Format — *Continued*

ASCII Character Position	Data Field Description
78-80	Outgoing Circuit† (hundreds, tens, units)
81	Space
82	Feature Flag
83	Carriage Return
84	Line Feed

* Data is right-justified and padded with blanks.

† Data is right-justified and padded iwht 0s.

Table B-7. 24-Word ISDN Expanded CDR Record Format

ASCII Position	Data Field Description
01	Time Hours (tens)
02	Time Hours (units)
03	Time Minutes (tens)
04	Time Minutes (units)
05	Space
06	Duration Hours (units)
07	Duration Minutes (tens)
08	Duration Minutes (units)
09	Duration Minutes (tenths)
10	Space
11	Condition Code
12	Space
13-16	Access Code Dialed*
17	Space
18-21	Access Code Used*
22	Space
23-37	Dialed Number* (The MCU extension in the MCU)
38	Space
39-48	Calling Number* (TAC of the trunk group for DS1 robbed bit signaling; CPN/BN, if available, for ISDN)
49	Space
50-64	Conference Billing Identification information*
65	Space
66-72	Authorization Code*
73-76	Space
77	FRL
78	Space
79-81	Incoming Circuit ID* (hundreds, tens, units)
82	Space
83-85	Outgoing Circuit ID* (hundreds, tens, units)
86	Space

Table B-7. 24-Word ISDN Expanded CDR Record Format — *Continued*

ASCII Position	Data Field Description
87	Feature Flag
88	Space
91	Space
92-95	Incoming Trunk Group Access Code*
96-99	Space
100-102	INS*
103-107	Space
108	BCC
109-120	Space
121-122	Bandwidth
123-130	Space
131	Carriage Return
132	Line Feed
133-135	Null

* Data is right-justified and padded with blanks

Call Detail Record Fields

The following list describes the CDR data collected for each call and the number of digits in each field. All information is right-justified in the respective field, unless otherwise indicated.

- Access Code Dialed (four digits on 24-word formats)

  NOTE:
  This field is used only for outgoing calls.

  This field can be the ARS access code, AAR access code, or the access code of a specific trunk group.

- Access Code Used (four digits on 24-word formats)

  NOTE:
  This field is used only for outgoing calls.

  This field contains the access code of the actual trunk group that the call was routed over (often the trunk group used is different from the trunk group access code dialed). This field always shows the access code of the used trunk group, even if it is the same as the dialed access code.

- Billing Identification (up to 15 digits)

  NOTE:
  This field contains Account Code.

  This is a numeric field, and contains a number that uniquely identifies a conference. Information in this field is right justified. The billing ID is optional and is administered on the conference form. It allows the system administrator to associate conferences with projects or account numbers.

  If the MCU cannot allocate resources for a conference, but the trunk call is answered, a CDR report for that call is generated. However, the billing identification field is left blank.

- Authorization Code (seven digits)

  This field contains the four- to seven-digit authorization code used to make the call. This applies to outgoing calls only.

- Bearer Capability Class (BCC) (one digit)

  This field contains the BCC for ISDN calls, identifying the type of an ISDN call. It distinguishes between different types of data. Either of the following digits may appear in this field.

  — 1 = Mode 1 (112k synchronous data)

  — 4 = Mode 0 (128k data clear)

  — w = Multirate bandwidth (112k, 128k, 168k, 192k, 224k, 256k, 280k, 320k, 336k, 384k, 448k, 512k, 672k, 768k, 1008k, 1152k, 1288k, 1344k, 1472k, 1536k, 1680k, or 1920k)

- Bandwidth (two digits)(24-word Record only)

  Used to capture the bandwidth of the multirate bandwidth calls. Bandwidth is expressed as the number of DS0s or 128k channels comprising a call, and is stored in the two-digit bandwidth field (1 or 6).

- Calling Number (up to five, and 10 digits in 24-word format)

  — For *outgoing calls*, this field contains the MCU extension.

— For *incoming calls*, this field contains the TAC of the trunk group used for the call (in the standard 18-word formats). In the 24-word format, the calling number field is 10 digits and contains the CPN/BN information, if provided, on incoming ISDN calls. If the CPN information is not output, the field is blank.

- Carriage Return (one character)

  The ASCII carriage return character followed by a line feed is used to terminate CDR records.

- Condition Code (one character)

  The codes in this field reflect special events relating to the call. Table 4-8 describes the condition code mapping.

Table B-8. Condition Code Mapping

Condition Codes	Description
4	Identifies an extremely long call (10 hours or more) or an extremely high message count for TSC (9999 messages or more). On a call exceeding 10 hours, a call record with this condition code and a duration entry of 9 hours, 59 minutes, and 1 to 9 tenths of a minute is produced after the first period. A similar call record with this condition code is produced after each succeeding 10-hour period. When the call does terminate, a final call record with a different condition code identifying the call type is produced.
7	Identifies calls served by the AAR or ARS Selection feature.
9	Identifies an incoming call.
A	Identifies an outgoing call.
E	Identifies an ineffective call attempt due to trunks not being available. This also identifies an ISDN Call-By-Call Service Selection call that is unsuccessful because of an administered trunk usage allocation plan.
F	Identifies an ineffective call attempt because of either (a) insufficient calling privileges of the originator (assigned per FRL), (b) ISDN calls rejected by the switch due to an NSF mismatch, or (c) an authorization mismatch which prevents the completion of a data call.

NOTE:
When more than one condition applies to a call, the overriding code is shown in the following table.

When two condition codes apply on the same call, one will override the other. The following table matrix defines the overrides. To illustrate how to use this matrix, assume that condition codes 4 and 9 apply to the same call. The matrix contains four horizontal rows (4, 9, E, and F) and four vertical columns (4, 9, E, and F). To find the condition code that overrides, look at the point of intersection between row 4 and column 9. In this case, condition code 4 overrides. This can also be found by looking at the point where row 9 and column 4 intersect.

Table B-9. Condition Code Override Matrix

	Condition Code			
	4	**9**	**E**	**F**
4	NA	4	NA	NA
9	4	NA	E	F
E	NA	E	NA	NA
F	NA	F	NA	NA

- Dialed Number (up to 15 digits)

  For incoming calls, this field contains the MCU extension that corresponds to the dial-out number. If more than 15 digits are dialed, the least significant digits are truncated.

  NOTE:
  Dial out allows 22 digits in the dialed number field. The same truncation rules apply so that, as an example, for the dial-out number 90114567890123456, the dialed number field of the CDR record would show 901145678901234.

  The # sign (“E”) may be printed in this field:

  — When the user dials a feature access code that starts with a #

  — When the user dials # at the end of digit dialing (for example, for WATS and IDDD calls)

  — When the inter-digit timeout occurs before the answer supervision timeout, even if the user has not dialed the # sign.

- Duration (four digits)

  All calls are timed. The timing is recorded in hours (0 through 9), minutes (00 through 59), and to the nearest tenth of a minute (0 through 9).

- FRL (one digit)

  FRLs, numbered zero through seven, are associated with the AAR and ARS features and define calling privileges. The information contained in this field is as follows:

— If the call is an outgoing call, this field contains the FRL of the originator.

— If the call is an incoming call, this field contains the FRL assigned to the incoming trunk group.

— If the call is an incoming tandem tie trunk call, this field contains either the FRL assigned to the tandem tie trunk or the Traveling Class Mark (TCM) sent with the tandem tie trunk call, depending on which was used to complete the call. On ISDN calls, this field always contains the TCM, if it was received.

— The CDR System Parameters can be administered to have ‘‘Disconnect Information in Place of FRL.’’ For trunk CDR, the following call disconnect data is printed in this field in place of the FRL data:

Data	Meaning (for calls)
0	Don’t know who dropped first
1	We dropped first
2	The CO dropped first
3	Maintenance got the trunk

- Feature Flag (one digit)

  The digit in this field indicates whether or not the switch has received answer supervision from the network and whether the call was a voice or data call.

  — A **4** in this field indicates a voice call with network answer supervision.

  — A **0** in this field indicates a voice call without network answer supervision.

  — A **5** in this field indicates a data call with network answer supervision.

  — A **1** in this field indicates a data call without network answer supervision.

  Answer Supervision is indicated for non-interworked ISDN calls.

  The answer supervision flag is interpreted as follows:

  — For ISDN trunks, if the answer supervision field contains a **5**, the call interworked with non-ISDN trunks and the duration was calculated but does not have the degree of accuracy of a strictly ISDN call. Thus, this field shows whether the call was interworked or went through a strictly ISDN network.

- Incoming TAC (four digits) (24-Word Records Only)

  This field contains the access code of the incoming trunk group.

- ISDN Network Services (INS) (three digits)

  This field specifies the INS requested for a call. This field applies only to ISDN calls. Each Network Specific Facility is translated into an INS according to the following table.

Table B-10. Network Specific Facility to INS Mapping

Network Specific Facility	INS Value
Network Operator	324
Presubscribed Common Carrier Operator	325
Software Defined Data Network (SDDN)	352
INWATS	355
Maximum Banded WATS	356
Long Distance Service	358
ACCUNET Digital Service	357
International 800	359
Multiquest	367

- IXC Code (three or four digits with an ISDN format)

  — Non-ISDN Formats

  IXC codes, numbered 1 through 15 (1 through F hexadecimal), are associated with the AAR and ARS features and depict the carrier used on the call. This information is sent to the CDR output device in ASCII code as a hexadecimal representation (for example, ASCII ‘‘F’’ equals ‘‘15’’).

  An IXC access number is used to access a specific common carrier for a call. In the US, this number is of the form 10XXX, 950 — 1XXX, or NXX — XXXX, where N is any digit 2 through 9 and X is any digit 0 through 9. The IXC access numbers applicable at a given location are associated with an IXC code on the IXC form. When ARS is used, and a routing pattern inserts one of the administered IXC codes, the associated IXC code is recorded. If no IXC access number is used, a 0 is recorded. In this case, either an IXC carrier is not used on the call or the carrier is selected at the CO. A one-character index (1 through 15 and through F hexadecimal) corresponding to the administered IXC code’s index in the first page of the IXC form is generated when that matching IXC code is used on an ARS call. If none of the administered IXC codes is used, a zero is recorded.

IXC codes on the first page of the IXC form can be any one of the following :

- Any number matching the second page of the IXC form
- A 7-digit number of the form 950-XXXX, where ‘‘X’’ is any digit 0 through 9
- Any 8 to 11-digit number

— ISDN Formats

With an ISDN record format, this field is a three-digit field that identifies the actual IXC used on an ISDN call. This information is determined from the routing pattern administration. On AAR and ARS calls, the three-digit IXC value is administered in the routing pattern for all ISDN calls. If a user dials an IXC code with a 10 XXX format as administered on the IXC Codes form, the CDR device will put only the last three digits in the CDR record. If a user dials a seven-digit IXC code, this field will contain a zero.

- Incoming Circuit Identification (three digits)

  This field contains the member number of a trunk within a trunk group used for an incoming call.

- Line Feed (one character)

  The ASCII line feed character followed by a carriage return is used to terminate CDR records.

- Null (one character)

  The NULL is used to terminate and divide CDR Records (usually in triplets) when needed by the receiving adjunct.

- Outgoing Circuit Identification (three digits)

  For outgoing calls, this field contains the member number of the trunk within a trunk group used. This field is blank for incoming calls.

- Space (one to forty characters)

  The ASCII blank character is used to separate other CDR fields or to blank fill unused record locations.

CDR Output Devices

A standard 232C interface is provided by the system's processor circuit pack. This allows for direct connection of the CDR output device to the system. If this port is not used, additional interface equipment is required as described in the Hardware and Software Requirements part of this feature description.

If the CDR link is down for more than a minute, some data may be lost. However, the most recent 180 records are stored by the MCU if the link goes down. When the link comes back up, these records are output on a ''first-in, first-out'' basis.

If the CDR buffer is full, the MCU overwrites the earliest CDR records.

The time stamp on calls recorded by CDR is normally applied at the end of the call.

The following paragraphs give a brief description of the printer as CDR output device.

Printer

An 80- or 132-column (character) printer can be connected as an CDR output device. The printer prints CDR records in a two-line format. No data processing or reports are provided. The 18-word CDR records sent to the printer are 84 bytes or 672 bits long. The 24-word CDR records sent to the printer are 135 bytes or 1,080 bits long.

CDR can be used as a polling device or can be buffered to the following equipment:

- Host computer
- Tape drive
- ASCII printer

Considerations

The system can store up to 180 CDR records (when the link is down), which are sent to the output devices when the link comes back up.

CDR records of DS-1 calls are generated only if the answer supervision timeout is exceeded or if the call is answered at the far end. Therefore, more accurate CDR records for DS-1 facilities can be obtained by setting the answer supervision timeout field on the DS-1 tie trunk form to the highest possible value (250 seconds).

Endpoints that are moved to another conference do not change the CDR data. This data appears as if the endpoints were never moved.

Interactions

For two-channel conferences, two records with identical Conference Billing ID fields are stored. For 1x56k, 1x64k, 112k, 128k, 168k, 192k, 224k, 256k, 280k, 320k, 336k, 384k, 448k, 512k, 672k, 768k, 1008k, 1152k, 1288k, 1344k, 1472k, 1536k, 1680k, and 1920k multirate conferences, a single record is stored on a per-call basis. For BONDed 336K or BONDed 384K (H0) calls, six records with identical conference billing ID fields are stored. The following interaction discussions assume CDR is activated.

- AAR and ARS

  CDR records the following information for ARS:

  - Fact that an ARS call was made
  - Calling extension number
  - FRL of the calling extension
  - Called number
  - Type of trunk group used for the ARS call
  - Time of call completion
  - Call duration (how long the parties talked)
  - IXC code, if any

  If CDR is suppressed for the trunk group actually used on an ARS call, an CDR record is not generated; otherwise, Condition Code 7 applies. The ARS access code is recorded in the Access Code Dialed field and the trunk access code for the trunk group actually used is recorded in the Access Code Used field.

  If an AAR call is placed to a busy trunk group and CDR is suppressed for that trunk group, the user hears reorder tone and the CDR output shows an ineffective call attempt.

- Call-By-Call Service Selection

  When a successful call is made on a Call-By-Call Service Selection trunk, the network specific facility used on the call is translated into an INS number and recorded in the INS field of the CDR record. If a Call-By-Call Service Selection call is unsuccessful because of an administered trunk usage allocation plan, the INS number is recorded in the INS field of the report with a condition code of ‘‘E.’’

- Intercept Treatment

  If an outgoing or tandem call is routed to Intercept Treatment, the number dialed by the calling party is recorded as the dialed number, and Condition Code F is recorded.

- ISDN

  When specific answer supervision is received from the network, an indication is sent to the CDR device to this effect. If an ISDN call has been interworked, the CDR record will not record the call as having answer supervision.

- Private Network Access

  Private Network Access calls will be recorded if CDR is administered for this trunk group. Private Network calls will be recorded if either an incoming and/or outgoing tie trunk is assigned CDR.

  UDP

  If one user calls another user via a Uniform Dial Plan extension number, and the trunk group used has CDR assigned, CDR records the following information:

  — Condition Code — 7

  — Access Code Dialed — blank

  — Access Code Used — trunk access code of trunk used

  — Dialed Digits — Uniform Dial Plan extension

Administration

CDR is administered via the CDR System Parameters form. For details, refer to "Administering the Call Detail Recording (CDR) Link" on page 2-39.

The following sections summarize what can be administered on the form.

System Parameters

- Whether standard or ISDN formats are used.
- The speed at which the CDR device connected to the direct RS-232C interface on the processor circuit pack will operate (300, 1200, 2400, 4800, or 9600 baud rate).
- Whether the reason for disconnect is recorded instead of the FRL.
- CDR can be suppressed for Ineffective Call Attempts. Ineffective call attempts include calls to incoming or outgoing trunks that are unavailable due to trunk usage allocation for ISDN Call-By-Call Service Selection trunks and incoming calls rejected by the switch due to NSF mismatch.
- Primary output layout: expanded or printer.
- Primary output destination: EIA or the assigned extension.
- ISDN layout?: A yes or no field.
- Data Format: either day/month or month/day format.

Date and Time

The date and time should always be updated for events such as a leap year, daylight savings time, or a system restart after a power failure. If a time of day is not administered, CDR records will not be generated.

Data Modules and Modems

The CDR output devices can be connected to a processor data module (PDM), Trunk Data Module, or a modem. The following items must be administered:

- A netcon channel must be assigned using a Data Module form and entering data-channel or netcon channel for the type. This channel provides a path for CDR data from the Switch Processing Element to the time-division bus.
- If the CDR output device is connected to an PDM, administer a PDM form.
- If the CDR output device is connected to a Trunk Data Module, administer a Trunk Data Module form.

If the EIA port on the Processor Interface circuit pack is used by the output device, the CDR output device extension should be administered as “eia.”

Hardware and Software Requirements

Hardware requirements depend on the type of output device used for CDR. The CDR output device can be connected directly to the processor circuit pack which provides a standard RS-232C interface. If the output device is connected via a 7400B data module (as is the case for a remote CDR printer), the 7400B must be administered so that 7400B result codes do not interfere with CDR data output.

Call-By-Call Service Selection

Description

Call-By-Call Service Selection allows a single ISDN-PRI trunk group to carry calls to more than one service or facility.

NOTE:
The Call-by-Call Service is only applicable for Country Protocol option 1 (US).

Without Call-By-Call Service Selection, each trunk group must be dedicated to a specific service or facility. Call-By-Call Service Selection eliminates this requirement by allowing a variety of services to use a single trunk group. These services are specified on a call-by-call basis. Trunking efficiency is immediately obtained with Call-By-Call Service Selection by distributing traffic over the total number of available trunks.

Services Used with Call-By-Call Service Selection

The services used on Call-By-Call Service Selection calls are assigned after an ISDN-PRI trunk group is assigned a service type of Call-By-Call Service Selection. A Call-By-Call Service Selection trunk group can be administered to carry calls to many services. The services relevant to the MCU are as follows:

- ACCUNET Digital Service — Digital network services for various high-volume, high-speed data transmission requirements.
- SDDN — An offering that provides a virtual private network using the public-switched network. SDDN can carry voice and data between customer locations as well as off-net locations.
- Other user-defined services — New service types can be assigned as they are developed and defined.

A Call-By-Call service selection example is shown in the following figure.

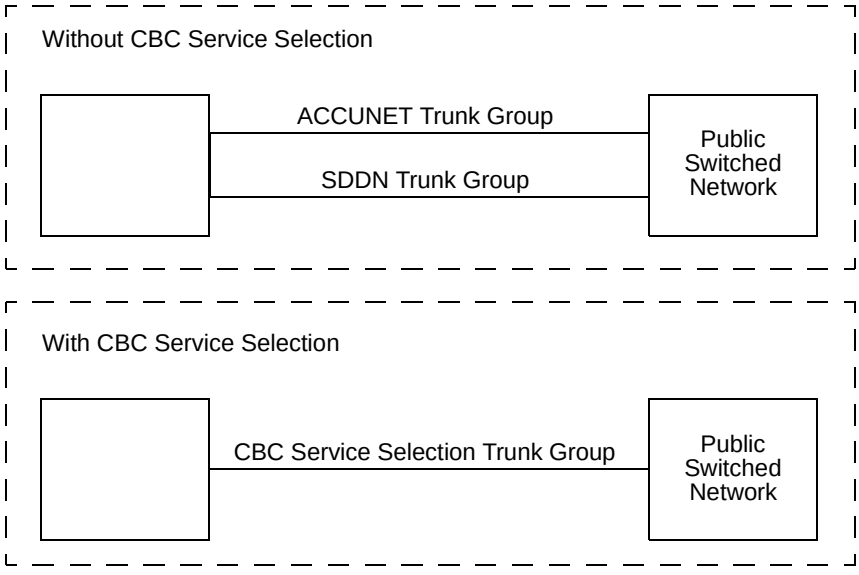

Figure B-5. Call-By-Call Service Selection Example

ISDN-PRI Messages and Information Elements used for Call-By-Call Service Selection

Although the technical details of ISDN-PRI messages and information elements are not critical to implementing the ISDN-PRI application, the following details may aid in the understanding of some readers and are, therefore, included in this description.

Call-By-Call Service Selection allows the system to specify one of the preceding service types on a call-by-call basis. This is done via a SETUP message that indicates the intent of the originating system to initiate a call using the specified service or facility. The SETUP message contains units called information elements that specify call-related information. The information elements used with Call-By-Call Service Selection are as follows:

- Network Specific Facility — indicates which facilities or services are to be used to complete the call.

  The system also checks all incoming ISDN-PRI calls for the presence of a Network Specific Facility information element. If this information element is present, the system makes sure that the requested service is compatible with the administration of the trunk. If the requested service is not compatible with administration, the switch rejects the call.

Considerations

Call-By-Call Service Selection provides the following benefits:

- Cost Reduction — since many services share the same trunks, the total number of trunks can be reduced.
- Improved Service — Call-By-Call Service Selection trunks can reduce the probability of features and services being blocked.
- Simplified Networking — network engineering is simplified because analysis of trunking needs can be done based on total traffic instead of on a per-service basis.
- The ability to respond to changes in a more timely fashion. The network does not have to be consulted because of the flexibility provided by the usage allocation plans.
- Measurement of Call-By-Call Service Selection calls.

Interactions

The following features interact with the Call-By-Call Service Selection feature.

- AAR

  Call-By-Call Service Selection uses the same routing tables and routing preferences that are used by AAR.

- ARS

  Call-By-Call Service Selection uses the same routing tables and routing preferences that are used by ARS.

- GRS

  Call-By-Call Service Selection uses the same routing tables and routing preferences that are used by GRS.

- CDR

  On successful incoming Call-By-Call Service Selection calls, the Network Specific Facility specified by the call's Network Specific Facility Information Element is recorded by CDR. CDR refers to this information as the INS (ISDN Network Service).

  When a Call-By-Call Service Selection call is rejected because of a trunk group usage allocation plan, CDR records the cause as an ineffective call attempt.

Administration

Call-By-Call Service Selection is administered by the system administrator on a per trunk group basis. The following items require administration:

- ISDN-PRI Trunk Group — must be administered with a Service Type of Call-By-Call Service Selection. The trunk group administration also includes the following:

  — Incoming Call Handling Treatment

- — Whether or not UAPs are required
- — UAPs
- — UAP Assignment Schedule
- — Group Member Assignments

- ARS or AAR Routing Patterns — Routing Patterns can be administered to include a Network Specific Facility and/or IXC.
- Network Specific Facilities Encoding — new Network Specific Facilities can be added as needed by the system technician.

Hardware and Software Requirements

A TN767D DS1 circuit pack is required for assignment of a signaling link and up to 23 ISDN-PRI Trunk Group members. A TN2207 circuit pack is required for E1 interfaces for assignment of a signaling link and up to 31 trunk group members.

The DS1 provides 24 ports, and the E1 provides 32 ports. A TN768 Tone-Clock circuit pack is required to provide synchronization for the DS1 circuit pack. A TN765 Processor Interface circuit pack is required in conjunction with the DS1 circuit pack for ISDN Link Administration.

Cascading

Description

Cascading provides the means to connect two MCUs via a switched H.320 or BONDing link to form a *cascaded* conference. For all practical purposes, each MCU treats the cascade connection as if it were simply another endpoint in the conference. All the features and services available for non-cascaded conferences are available in cascaded conferences as well. Cascading provides some particular benefits:

- Through Cascading, network charges can be greatly reduced for a conference between geographically distant endpoints. Without Cascading, all endpoints must call into one MCU that may be distant from some of the endpoints. This can mean many endpoints, each paying a network toll for each connection to the MCU. With Cascading, the endpoints can connect on an MCU that is geographically close to them and then request a single line for a cascade call to another MCU being used by other participants. This can be particularly useful for cross-continent conferences.
- Cascading allows larger conference sizes since the cascaded MCUs pool resources for as many as 50 multimedia endpoints (25 per MCU) and as many as 50 audio-only endpoints (25 per MCU) for a maximum of 25 endpoints per MCU per conference.

In addition, Star Cascading allows a maximum of 25 cascaded endpoints per MCU, as opposed to one such endpoint per MCU in some previous releases.

Considerations

In a cascaded conference, one MCU is referred to as primary and the other secondary. Any given MCU can be primary in one cascaded conference yet secondary in another simultaneous cascaded conference.

Additionally, each MCU in a cascaded conference honors its own stop time. The first MCU to reach its individual stop time is dropped from the conference despite the stop time of the second MCU.

The times for the cascaded MCUs must be within one minute of each other.

Interactions

The cascade feature interacts with the following features.

- CDR

  Each MCU in the cascade outputs its own CDR records for calls in the conference.

- Dial-Out

  One MCU in the cascade uses the Dial-Out feature to call the second MCU to provide the cascading connection.

Administration

It is recommended that the following fields in the Conference Record form be the same for both MCUs in a prearranged conference.

- *Conference Number*
- *Conference Name*
- *Billing ID*
- *Entry/Exit Tones*

Ideally, these fields should be (but need not be) identical for dynamically joined cascades. However, for two independent conferences to be dynamically joined, the following fields must be identical for each.

- *Password*

  All MCUs in the cascade must have the same password or no password.

- *Number of Channels and Bandwidth*

  The effective rate of the conference (1x56k, 1x64k, 2x56k, 2x64k, 112k, 128k, 168k, 192k, 224k, 256k, 280k, 320k, 336k, 384k, 448k, 512k, 672k, 768k, 1008k, 1152k, 1288k, 1344k, 1472k, 1536k, 1680k, or 1920k)

- *Conference Mode*

  The selected conference control mode must be the same for all MCUs involved.

To ensure that the dial-in MCU is ready to receive the connection that is initiated by the dial-out MCU, the dial-in MCU should be administered such that the conference is active before the dial-out MCU.

Star Cascading

This feature merely requires the administration of additional cascaded endpoints via the Conference Record form. A maximum of 25 cascaded endpoints per MCU is allowed. All other administration is identical to conventional Cascading.

Hardware and Software Requirements

Consult your Lucent Technologies service representative.

Class of Restriction

Description

Class of Restriction (COR) defines up to 64 different classes of call origination and termination privileges. Systems may have only a single COR, one with no restrictions, or many CORs up to 64 as necessary to effect the desired restrictions.

A COR is assigned to each of the following:

- Maintenance alarm terminals
- Data module
- Trunk group

Use of CORs can be categorized as follows:

- Calling party restrictions (that is, the extension for the maintenance alarm terminal)
- Called party restrictions
- Facility Access Trunk Test
- Fully Restricted Service
- Restriction Override
- Restricted Call List
- Unrestricted Call List
- Miscellaneous restriction groups
- Selective denial of public network calling through a CCSA or EPSCS network
- An ARS or AAR FRL for control of call routing

Features assignable as calling party restrictions are as follows:

- Origination Restriction
- Outward Restriction
- All-Toll Restriction
- TAC-Toll Restriction

Features assignable as called party restrictions are as follows:

- Inward Restriction
- Manual Terminating Line Restriction
- Termination Restriction
- Public Restricted

Use of CORs

CORs can be used to assign a variety of restrictions to a variety of facilities. The types of restrictions that can be assigned are discussed in the following paragraphs. The Unrestricted Call List field is displayed in the Class of Restriction form only if the calling party restriction is administered as TAC-Toll or All-Toll.

Calling Party and Called Party Restrictions

Calling party restrictions prevent specified users from placing certain calls or accessing certain features. Features assignable as calling party restrictions are Code Restriction, Origination Restriction, Outward Restriction, and Toll Restriction. These individual features are fully described elsewhere in this chapter. A brief description is given here:

- Outward Restriction — prevents callers at specified voice terminals from accessing the Public Network Access feature. Calls can be placed to other voice terminal users and to tie trunks.
- Origination Restriction — prevents callers at specified voice terminals from originating calls. Voice terminal users can, however, receive calls.
- TAC-Toll Restriction — prevents callers at specified voice terminals from making trunk access calls to certain toll areas as defined on the system's administered Toll Analysis form, unless the number is on an Unrestricted Call List associated with the caller's COR. This restriction applies to calls made using trunk access codes of CO or FX trunk groups. See the Restriction — Toll feature description elsewhere in this document for more details.
- All-Toll Restriction — identical to the TAC-Toll Restriction described above, except this restriction also applies to ARS calls. See the Restriction-Toll feature description elsewhere in this manual for more details.

Called party restrictions prevent specified users from receiving certain calls. Features assignable as called party restrictions are Inward Restriction, Manual Terminating Line Restriction, Termination Restriction, and Public Restriction. These individual features are fully described elsewhere in this chapter. A brief description is given here:

- Inward Restriction — restricts users at specified voice terminals from receiving public network, access endpoint-originated, and access endpoint-extended calls.
- Manual Terminating Line Restriction — restricts users at specified voice terminals from receiving calls other than those from an access endpoint.
- Termination Restriction — restricts users of specified terminals from receiving any calls.
- Public Restriction — restricts users of specified voice terminals from receiving direct public network calls.

The field labeled `Calling Party Restriction` and the field labeled `Called Party Restriction` are both administered as none. However, the field `Calling Party Restriction` could be administered as any of the other previously described calling party restrictions. Likewise, the field for `Called Party Restriction` could be administered as any of the other previously described called party restrictions. Including ‘‘none’’ as a choice of restrictions, as many as 20 combinations of calling and called party restrictions are possible.

However, it is unlikely that all 20 combinations are needed in any one situation. Therefore, only the required ones should be established.

Calling and called party restrictions are the basis for all CORs. In cases where no restrictions are needed, a single COR could be assigned with calling and called party restrictions of ‘‘none.’’ This same COR could be used for unrestricted trunk groups, terminating extension groups, UCD groups, DDC groups, and data modules.

The following are typical examples of calling and called party restrictions that may be assigned to a COR:

- Long-distance calling is limited by All-Toll Restriction, but there are no restrictions on incoming calls.
  - Calling party restriction=All-Toll
  - Called party restriction=None

The called party restriction is checked only at the called terminal, module, attendant console, zone, or group.

Each COR is established as needed and is arbitrarily identified by a number.

Facility Access Trunk Test

This field on the COR form is used to grant a user permission to make Facility Test Calls to access trunks. A y in this field allows users with this COR to make these calls. An n in this field causes users with this COR to receive intercept treatment when they attempt to make these calls. For more information on Facility Test Calls, see the Facility Test Calls feature elsewhere in this chapter.

Fully Restricted Service

Denies the specified voice terminal access to public network trunks for either incoming or outgoing completion. Note that Restriction Override should be set to No.

Restricted Call List

A Restricted Call List is assigned to the system by the system administrator. This call list is made up of specific digit strings that cannot be dialed from facilities that are restricted by the call list. The `Restricted Call List` field on the COR form is used to determine which facilities are restricted by this call list. If a COR has this field administered as y, facilities with that COR cannot be used to dial a number that matches one of the digit strings in the Restricted Call List.

Unrestricted Call List

Ten Unrestricted Call Lists are assigned to the system by the system administrator. Each `Unrestricted Call List` is made up of specific digit strings that can be dialed from facilities associated with the call list. The Unrestricted Call List fields on the COR form are used to determine which facilities have access to each of the 10 call lists. If a COR has any of these 10 fields administered with an Unrestricted Call List number, facilities with that COR can call any numbers contained in the assigned list(s) (unless the number is included in the Restricted Call List, which overrides the Unrestricted Call List).

Selective Denial of Public Network Calling Through a CCSA or EPSCS Network (APLT)

Public network calling via the private CCSA or EPSCS network (commonly referred to as off-network calling) is optional on a per-private network basis. If off-network calling is not provided, then the APLT field can be ignored. If off-network calling is provided, then permission or denial to access the off-network capability is set via the APLT field. Users assigned a COR that has APLT set to **n** (no) can use off-network calling. Users assigned a COR that has APLT set to **y** (yes) cannot. If there is a need for both yes and no choices in a system, separate CORs must be assigned to reflect this.

The field labeled APLT is preset as y. This means that a facility with this COR is not allowed to access CCSA or EPSCS off-network capabilities for public network calling. An n in this field would indicate that the facility can access CCSA or EPSCS off-network capabilities.

ARS/AAR FRL for Control of Call Routing

If the system does not use AAR or ARS to determine the most preferred routing of calls, then the FRL field can be ignored. If AAR or ARS is used, then an FRL is used to either allow or deny access to certain routes. The FRL for the outgoing (trunk) side of the call is provided in the AAR or ARS Routing Pattern. Although each outgoing trunk group has a COR and each COR has an FRL, this FRL is not used unless the trunk group is the originator of the call. Call routing is determined by a comparison of the FRLs in the AAR or ARS Routing Pattern and the FRL in the COR of the call originator.

The **FRL** field is preset to **7**. However, this field can have a value of 0 through 7. An originating FRL of 0 has the least calling privileges, whereas an originating FRL of 7 has the most calling privileges. Each route in each of the AAR or ARS Routing Patterns also has an FRL. These route FRLs can also have a value of 0 through 7. A route FRL of 0 is the least restrictive, whereas a route FRL of 7 is the most restrictive. An FRL of 0 is checked before the other routes in a given ARS routing pattern. To access a route, the originating FRL must be greater than or equal to the route FRL. Determination of appropriate FRL values must be made with respect to the outgoing routes from a specific system and the desired levels of calling privileges. This is part of AAR or ARS customization. The FRL of the call originator is contained in the COR assigned. The **FRL** field in a COR assigned to an outgoing trunk group is never checked and should be ignored.

Assuming AAR and/or ARS has been customized for a system, the system administrator must establish unique CORs for each of the up to eight levels of ARS calling privileges that is used in the system. However, these CORs must maintain the desired restrictions dictated by the other fields on the screen form. The simplest case is a COR specifying no restriction. Ordinarily, this COR can be assigned to all unrestricted users. However, if some subset(s) of these users requires different FRLs, separate CORs must be established for each different FRL required.

For a detailed description of AAR, ARS, and FRLs, refer to the individual feature descriptions given elsewhere in this chapter.

COR Example

CORs could be used to prevent certain uses of the maintenance alarm terminal for anything other than its intended use.

Considerations

COR provides the means to consolidate assignment and administration of the various restriction features available with the system.

All items associated with a COR are distinct and separate. A unique COR must exist for each needed combination of FRLs, CCSA/EPSCS off-network restrictions, calling party restrictions, called party restrictions, and miscellaneous restrictions. Up to 64 CORs can be established, as required, to provide the needed combinations.

Interactions

The following features interact with the Class of Restriction feature.

- AAR or ARS

  Originating FRLs are assigned via a COR. Termination and Miscellaneous Restrictions do not apply to ARS or AAR calls.

Administration

COR is administered by the system administrator. For each COR that is assigned, the following items must be administered:

- Access to Malicious Call Trace
- COR Number
- FRL
- Permission to access EPSCS or CCSA off-net facilities
- Calling Party Restriction

- Called Party Restriction
- Permission to call other CORs
- Forced Entry of account codes for CDR (yes or no)
- Partitioned Group Number
- Priority Queuing (yes or no)
- Service Observing (yes or no)
- Time of Day Plan Number
- Direct Agent Calling
- Facility Access Trunk Test
- Fully Restricted Service
- Restricted Call List
- Unrestricted Call List

Assignment of Restrictions

A COR is assigned to each of the following:

Voice Terminals

All voice terminals must be assigned a COR. The same COR may be assigned to all voice terminals or a unique COR may be assigned to a particular voice terminal or group of voice terminals. This COR applies individually to each voice terminal and is independent of all other COR applications, such as Miscellaneous Restriction groups or UCD groups.

The main items of concern for individual voice terminals are calling party restrictions and called party restrictions (discussed previously under ‘‘Use of CORs’’). If no restrictions are needed for a certain group of voice terminals, ‘‘none’’ can be specified for both calling party and called party restrictions. If it is desired to restrict a group of voice terminals from making outside calls, a COR specifying a calling party restriction of ‘‘outward’’ should be established.

Additionally, miscellaneous restrictions, restrictions to CCSA and EPSCS off-network calling capabilities, and FRLs also apply. A separate COR must be established for each unique set of restrictions.

Trunk Groups

Each trunk group is assigned a COR. Trunk groups are assigned CORs mainly for the use of miscellaneous restrictions. Calling party and called party restrictions should be ‘‘none.’’ Whether or not a CO or FX trunk group is restricted is specified on the trunk group form used during implementation. CO and FX trunk groups default to being toll restricted for TAC calls. Toll Restriction for TAC calls can be disabled for certain CO or FX trunk groups on the trunk form.

Data Module

Each data module is assigned a COR.

Hardware and Software Requirements

No additional hardware or software is required.

ClearPresence @716 Audio Coder

Description

The ClearPresence @716 feature is an audio mode that provides 7 kHz of voice quality while utilizing only 16 kbs of H.221 bandwidth. As such, this audio mode provides Wideband-quality benefits when compared to G.728 and reduced H.221 bandwidth usage when compared to G.711 or G.722, therefore preserving more bandwidth for video. ClearPresence @716 is based on an algorithm developed by Lucent Technologies Bell Laboratories.

Considerations

ClearPresence @716 complements the other audio modes currently supported by the MCU, including G.722, G.728, and G.711. ClearPresence @716 is possible in conferences and point-to-point calls for endpoints supporting ClearPresence @716.

ClearPresence @716 interworks with G.722, G.728, and G.711. Furthermore, ClearPresence @716 provides interworking between G.722 and G.728 in Speed Match conferences.

The feature affects MCU endpoints as illustrated in the following table.

Table B-11. Per-Conference Audio and Video Capabilities of Endpoints Per ClearPresence @716 RTU and Other Factors

Audio Mode Field Value in Conference Record Form	Audio/Video Capabilities of Endpoint(s)	Transfer Rate of Endpoint(s)	Capabilites Received By Endpoints	
			Audio	Video
G.711	Video-capable	Any	G.711	Yes
G.722	G.722		G.722	
	Non-G.722		G.711	
G.728	G.728		G.728	
	Non-G.728		G.711 (downgraded to secondary endpoint)	No
auto-down_only, **auto-w/upgrade**	Originally all ClearPresence @716 or G.728; subsequently joined by G.722 or G.711 endpoint(s)	less than or equal to 2B	G.711 or G.722 (downgraded)	Yes
	All ClearPresence @716; subsequently joined by endpoints not supporting ClearPresence @716	greater than 2B	Original audio mode for all endpoints supporting ClearPresence @716; if endpoints not supporting ClearPresence @716 join, downgraded to G.711 or G.722	
ClearPres716	ClearPresence @716	Any	ClearPresence @716	Yes
	Non-ClearPresence @716; audio bit rate that matches video with ClearPresence @716		Audio that matches video with ClearPresence @716	Yes
	Non-ClearPresence @716; audio bit rate that does not match video with ClearPresence @716		G.711 (downgraded to secondary endpoint)	No

To obtain the benefits of the feature, customers must pay an RTU fee.

Interactions

ClearPresence @716 interworks with G.722, G.728, and G.711. Furthermore, ClearPresence @716 provides interworking between G.722 and G.728 in Speed Match conferences.

The interworking of G.722 and G.728 is not allowed in homogeneous conferences (that is, in conferences where all endpoints are administered at the same transfer rate).

ClearPresence @716 supports the following features: Audio Add-On, Cascading, and Lip Synchronization.

ClearPresence @716 supports the following features with the considerations as noted.

- Dynamic Resizing

  ClearPresence @716 is allowed as an option for changing the audio mode of a conference depending on the administered audio mode and transfer rate of the conference. The following tables show the allowable audio modes for Dynamic Resizing when the ClearPresence @716 RTU is optioned and when it is not optioned.

  NOTE:
  The allowable audio modes for the ClearPresence @716 RTU not optioned are identical to those for the R5.0 MCU. This applies to both Speed Match and non-Speed Match conferences.

  The values in the following tables are for non-T.120 Data Conferencing conferences. For T.120 Data Conferencing conferences, the MLP rate may block some of the allowable audio modes (as is true for the R5.0 MCU). For ClearPresence @716 optioned, the capabilities in the corresponding table apply to both non-Speed Match and Speed Match conferences.

Table B-12. Audio Mode Eligible for Changing in Dynamic Resizing Conferences (ClearPresence @716 RTU Not Optioned)

Audio Mode Field Value Administered in Conference Record Form	Transfer Rate	Audio Modes Eligible for Changing
G.722	less than or equal to 2B	G.722, G.711
	greater than 2B	G.722, G.711, auto-down_only, auto-w/upgrade
G.728	less than or equal to 2B	G.728, G.711, auto-down_only, auto-w/upgrade
	greater than 2B	G.728, G.711
G.711	less than or equal to 2B	G.728, G.711, auto-down_only, auto-w/upgrade
	greater than 2B	G.728, G.711
auto-down_only or **auto-w/upgrade**	less than or equal to 2B	G.728, G.711, auto-down_only, auto-w/upgrade
	greater than 2B	G.728, G.711, auto-down_only, auto-w/upgrade

Table B-13. Audio Modes Eligible for Changing in Dynamic Resizing Conferences (ClearPresence @716 RTU Optioned)

Audio Mode Field Value Administered in Conference Record Form	Transfer Rate	Audio Modes Eligible for Changing
ClearPresence @716	any	ClearPresence @716, G.722, G.728, G.711, auto-down_only, auto-w/upgrade
G.722	any	ClearPresence @716, G.722, G.728, G.711, auto-down_only, auto-w/upgrade
G.728	any	G.728, G.711
G.711	any	G.728, G.711
auto-down_only or **auto-w/upgrade**	any	ClearPresence @716, G.722, G.728, G.711, auto-down_only, auto-w/upgrade

- Speed Match

  In Speed Match conferences, each endpoint experiences audio based on its capabilities and the audio mode of the conference as specified in Table B-11 on page B-64.

  G.728 and G.722 can interwork in Speed Match conferences. Endpoints joining at a transfer rate less than or equal to 2B can experience G.728 audio; endpoints joining at a transfer rate greater than 2B transfer can experience G.722 audio.

- T.120 Data Conferencing

  Whenever the audio mode is “auto-w/upgrade” or “auto-down_only,” only MLP rate “6.4k” is permitted.

  NOTE:
  This also applies if ClearPresence @716 is not optioned.

  SM T.120 Data Conferening conferences are allowed with audio mode “ClearPresence @716” and MLP rate “fixed” (and not “var_MLP”). Also, these conferences are allowed with audio mode “auto” and MLP rate “6.4k.”

Administration

When administering a customer’s system, the VTC should use the list configuration all command to verify that no TN788A or TN788B circuit packs are in the system.

You must administer the Customer Options Form and the Conference Record form to administer the feature. For details, refer to "Administering ClearPresence @716 Audio Coder" on page 2-255.

Hardware and Software Requirements

None.

Conference Redial Flag

Description

This feature allows a user to ask the reservations agent to redial the call. This can occur for any of the following reasons:

- The user failed password validation three times and wants to try again.
- The user missed the password entry window.
- The user disconnected from the conference and wants to rejoin.

There may be other legitimate reasons as well. With the redial flag set, a failed call receives the same redial treatment as any other dial-out call.

Considerations

For an endpoint that has one connection established successfully, the redial applies only to the second connection.

Interactions

None.

Administration

Administer the *(Re)dial* field in the Conference Record form, as appropriate. Values include **y**, **n** and blank. Enter **y** to enable redial capabilities. Enter **n** to indicate that the MCU will not dial out at the start of the conference.

The field value can be changed if Dynamic Resizing is optioned. The field should be blank whenever the conference is not active or not in use, or if *Class* is set to **file**.

Hardware and Software Requirements

Consult your Lucent Technologies service representative.

Conference Reservation and Control System

Description

The CRCS feature is an optional software package used on a personal computer or personal computer network to view and add MCU information. With this feature, one or more reservations agents can perform conference reservation and system administration functions.

Considerations

There are single-user and multiuser versions available for the CRCS software. The multiuser version allows a company such as a service provider to subdivide (or *partition*) logically the MCUs to which it is attached so that several reservations agents and administrators can work within their partition as if on a separate MCU.

Conference reservation functions are performed via the ResCenter module. This module provides a graphical user interface (GUI) for scheduling and managing conference rooms, point-to-point conferences, and multipoint conferences on the MCU. ResCenter is available in single-user and multiuser client/server models.

The *RESOLVE* option is available within single-user and multiuser ResCenter. This option provides Web-browsing capabilities, and it is intended to “steer” some of the conference reservation tasks away from the CRCS agents and toward the end-users.

Administration functions are performed via the OpCenter module. This module is a PC-based real-time GUI that enables conference management. OpCenter is available in two modes. “Basic” OpCenter is integrated with the ResCenter database. Standalone OpCenter, or “OpCenterSP,” is not intergrated with ResCenter. OpCenter can support multiple simultaneous agents in a multipoint conference. Also, Multi-User OpCenterSP can support more than two MCUs.

OpCenter allows you to move active endpoints between conferences via the **move** command. The conference operator can use the **move** command to provide traditional “meet and greet” services, troubleshoot a customer site in isolation, or provide subconferencing services.

UCC *Web* provides an extension of UCC P&C-like capabilities to conference participants via a standard web browser interface. UCC *Web* provides bi-directional communication between the web server (and therefore the end client’s browser) and OpCenter.

Starting with R6.0, the “One-Release Back” CRCS option provides CRCS control of multiple MCUs running either of two software releases. For the R6.0 version of this feature, CRCS control is provided for the R5.0 and R6.0 MCUs.

To use the CRCS software, you must supply the personal computer or network system as described in the appropriate online CRCS documentation.

NOTE:
For more information on CRCS, refer to the MultiPoint System guide and also to the appropriate online CRCS documentation.

Interactions

CRCS supports all MCU conferencing features. Refer to the appropriate online CRCS documentation for details.

Administration

An MCU-ST is administered as an additional MCU-MT. CRCS-specific administration, including MCU partitioning, is described in the appropriate online CRCS documentation.

Hardware and Software Requirements

CRCS remote access via modem pooling requires the TN746B and TN754B circuit packs, the 7400A data module, and a modem. See the appropriate online CRCS documentation for details.

Continuous Presence *Plus*

Description

This feature is a customer option that enables a multipoint video conference to compose a picture consisting of the video from multiple (up to 16) sites participating in the conference. Each Continuous Presence *Plus* conference shows a Continuous Presence *Plus*-mixed (CPP-mixed) video to all “eligible” sites from the beginning of the conference. The user can transition from one video layout to another (including quad-image, dual-panoramic, lower-panoramic, upper-panoramic, as well as full screen) via administration, conference control commands, UCC, and CRCS.

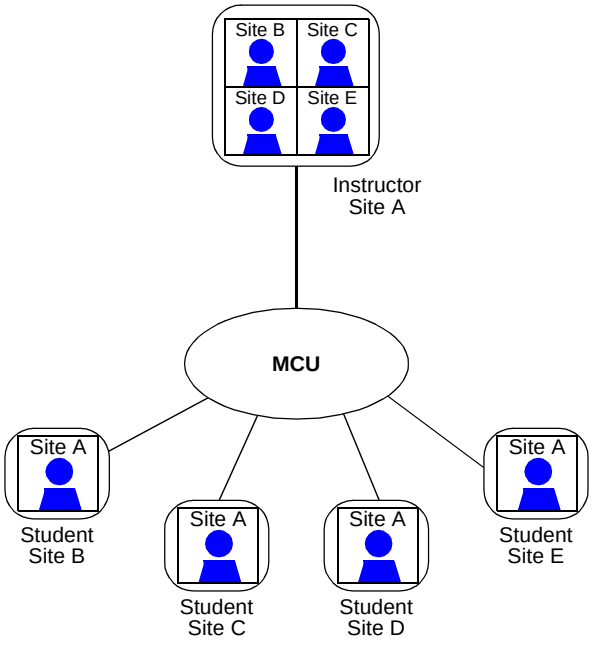

Figure B-6. Continuous Presence *Plus* Quad-Image Scenario

Considerations

Continuous Presence *Plus* supports multipoint video conferences at the following rates: 56/64k, 112/128k, 224/256k, 336/384k, 448/512k, and 672/768k. At the 448/512k and 672/768k rates, the maximum number of simultaneous CPP conferences that the system can support is reduced. (Refer to the “Administration” section for details.)

The “basic” CPP conference can support either two or three images (also known as “panoramic”) or four images (also known as “quad-images”). The user can select the type of image to be displayed. The panoramic composition continuously shows video from two or three endpoints. For upper-panoramic mode, the lower half of the screen is segmented into two quadrants, and each quadrant displays video from an endpoint. The upper half of the screen is not segmented, and it shows the video from still another endpoint. For lower-panoramic mode, the upper half of the screen is segmented into two quadrants, and each quadrant displays video from an endpoint. The lower half of the screen is not segmented, and it shows the video from still another endpoint. For dual-panoramic mode, neither the top half nor the bottom half of the screen is segmented and, therefore, video from a total of two endpoints is shown. The quad-image (4-image) configuration continuously shows video from four endpoints. The entire screen is segmented into four quadrants of identical dimensions, and each quadrant displays video from one endpoint.

The user can select the CPP picture composition on a per-conference basis. The default is set to quad-image. All the CPP quad-related features, such as fixing quadrants, UCC control, etc, are available in CPP panoramic.

The user can transition among the panoramic, quad-screen, and full-screen images. For transitioning between panoramic and quad-screen, the previous composition of the two quad images is retained whenever possible.

Even if the CPP-image is set to a panoramic mode, fixing the fourth quadrant via conference control or administration is still allowed; however, it is ignored in such a case. Fixing an endpoint to the fourth quadrant takes place only whenever quad-screen mode is in effect. Within panoramic CPP mode, whenever the endpoint fixed in the fourth quad speaks, this endpoint is fixed into one of the VAS quads. On transition to quad-screen mode, the endpoint fixed in the fourth quad appears in the fourth quad.

The MCU allows users to view seven, 10, 13, or 16 CPP participants simultaneously on the same screen. For this capability, several requirements must be in place, and the appropriate administration must be done. (Refer to the “Hardware and Software Requirements” and “Administration” sections for details.)

NOTE:
As the number of CPP images increases, the number of CPP conferences supported decreases.

UCC can be used in conventional CPP fashion for each individual conference. There can still be up to 24 participants in any of the individual conferences. Also, any *one* or *four* of these participants can contribute to the composed image at any given time.

CPP non-SM conferences support the 30 fps frame rate.

The composition and arrangement of the sites to be included in the mixed video image can be controlled via administration and control interfaces provided by the MT and CRCS. Control over the image can be exercised before the conference begins as well as during an active conference.

Voice-Activated Switching determines the composition of the image for any quadrants that have not been fixed via the administration and control interfaces. Universal Conference Control (UCC) provides another means to control the image.

Effect of Adding and Dropping Endpoints

Parties joining or dropping from a Continuous Presence *Plus* conference do not cause the conference to transition from full screen to CPP-mixed video, or vice versa.

As new conferees join the conference, they enter one of the following: their designated fixed quadrant or half-screen, an MCU-selected Voice-Activated Switching (VAS) quadrant, or no quadrant at all.

When a site drops from a Continuous Presence *Plus* conference, and if the site is currently part of the Continuous Presence *Plus* image, the effect on the CPP-image depends on whether the dropping site is occupying a VAS quadrant or a fixed quadrant. (A fixed quadrant is established via conference administration, UCC, or CRCS.)

If the dropping site was occupying a VAS quadrant or half-screen, the MCU attempts to select another source to insert in place of the dropping site. If no such source is available, the MCU follows the “empty quadrant (half-screen)” behavior (described in the next section). If the dropping site was occupying a fixed quadrant (half-screen), no alternative site is automatically selected to occupy the quadrant (half screen).

Empty Quadrants/Half-Screens and “Fill” Patterns

The *Empty Quadrant Extension* field in the Conference Record form is used to provide a fill pattern in the CPP display whenever there are fewer than four parties (for a quad-screen conference) or three parties (for a panoramic conference) in the CPP conference. By default, the Lucent Technologies MCU provides a solid blue fill pattern. Howerver, the user can designate a specific site in lieu of the fill pattern to display, for example, information pertinent to the user’s company (for example, the company logo).

Interfaces Affecting the CPP Image

The composition of the Continuous Presence *Plus* image can be changed via UCC, administration at the MT, administration at the CRCS ResCenter, or CRCS

OpCenter. For example, these interfaces can be used to toggle a conference between CPP image and single image. Commands in the BAS channel, such as MCV/Cancel-MCV and VCB/Cancel-VCB, can also toggle from one image to the other.

Whenever a Continuous Presence *Plus* conference is currently in CPP-mixed video mode, any change to the composition of the CPP image via any interface causes the image viewed by the user to change immediately.

NOTE:
There is no hierarchy of interface control. The last change to the CPP image from any interface becomes the current Continuous Presence *Plus* composition.

The control interfaces can change the stored composition of the CPP-image while the conference participants are viewing full-screen video. A change to the composition of the CPP image does not by itself force the conference to exit full-screen mode and return to CPP-mixed video mode. The change takes effect once the conference returns to CPP-mixed video mode.

NOTE:
The return to CPP-mixed video can always be forced, if desired, via the separate full-quad toggle provided at each control interface. For example, a user might send MCV, thereby forcing the conference to full screen in order to send a still frame. While the conference is in full screen, administration changes the extension for quadrants 1 through 4. This change is stored. Once the conference returns to CPP-mixed video (via Cancel-MCV, for example), the newly administered composition is shown.

Whenever a Continuous Presence *Plus* conference transitions from single image to CPP image, the current composition of the quadrants stored in the MCU is used. The composition of the CPP image can be changed any number of times by any control interface while the conference is in full-screen mode. The MCU simply updates its stored information regarding the content of each quadrant as each change is made from each interface. The MCU refers to this information to decide which sites to display when the conference toggles back to CPP-mixed video.

Whenever a Continuous Presence *Plus* conference transitions from CPP image to single-image for any reason other than MCV (see-me) or VCB (chair), the MCU reverts to VAS mode for the single image. In VAS mode, the broadcast video is selected via VAS. In Presentation mode, the return video seen by the presenter is selected via VAS. For MCV, the site sending MCV is shown, per usual. For VCB <M> <T>, the broadcaster identified by MCU number 'M' and terminal number 'T' is shown. If UCC chooses a single-image broadcaster, and if the conference is then transitioned to CPP-mixed video and later back to full-screen video, VAS is used to select the full-screen broadcaster (that is, the UCC-selected single image broadcaster is not "remembered").

Whenever a Continuous Presence *Plus* conference is in single-image due to MCV (see-me), any of the control interfaces can break the MCV/VCB "locks" and return the conference to CPP-mixed video mode. If a control interface requests a return to CPP-image video, and if there are multiple MCV requests outstanding, the MCU returns the video to the stored CPP-image composition.

Whenever a Continuous Presence *Plus* conference is currently using a UCC single-image option (such as Select Broadcaster, Rollcall, or Browse), any of the other control interfaces can break the associated "locks" and return the conference to CPP-mixed video mode.

Changes to other conference parameters using MT/CRCS change/edit

Whenever the Conference Record form for an in-use conference is edited for reasons other than to make a change to the composition of the CPP image, the CPP image composed by the MCU is unaffected by the edit.

Video Switching Mode and Broadcaster Notification

The MCU cannot send broadcaster notification ns-command information to sites eligible to view a composed quad image. Such an action would be unwarranted because this feature announces the current video switching mode and broadcaster to all endpoints supporting the Multi-Byte Extension (MBE) capability. Also, the MCU sends a nonstandard command (ns-cmd) with information that enables the endpoint to determine the conference mode (for example, VAS, Presentation Mode, etc.) that is in effect. The current broadcaster is also included in the message.

Self-broadcast/Multipoint Command Visualize (MCV)

A terminal may send MCV to the MCU to force its video to be broadcast to other terminals in the conference. For example, MCV can be sent to lock the broadcaster prior to transmitting a still-frame graphic over the video channel. Whenever a Continuous Presence *Plus* VAS conference is in full-screen mode, MCV requests from a terminal are honored, just as in non-Continuous Presence *Plus* conferences. The MCV “locking” effect on the full-screen broadcast video is broken and, if appropriate, the BAS code “VCR” is sent to the previous MCV broadcaster whenever the following are true:

- Any unlocking action applicable to non-Continuous Presence *Plus* conferences occurs.
- Conference is transitioned back to CPP-mixed video operation via the MCU-MT, CRCS, or UCC interfaces (that is, if the conference is subsequently returned to full screen, the MCU need not “remember” and display the previously locked full-screen broadcaster selected via VCB).

The MCU automatically transitions a Continuous Presence *Plus* conference to full-screen MCV operation whenever MCV is received while the conference is in CPP-mixed video mode.

NOTE:
Some terminals may send MCV and then fail to send Cancel-MCV.

Since MCV causes an automatic shift to full screen, a conference could transition to full screen and remain in full screen. The control interfaces provide an override back to CP video that can be used in such a case.

If one endpoint sends MCV to the MCU and is the MCV single-image broadcaster, and if other endpoints also send MCV, the requests by these other endpoints are remembered. If the current MCV broadcaster sends Cancel-MCV or drops, the MCU selects one of the other MCV endpoints to display in single-image mode. Once the last site cancels MCV, the conference returns to the state in effect before the first site sent MCV. For example, if the conference was in CPP-image mode when MCV took effect, the conference returns to CPP-image mode. Unless the first MCV was in quad mode, if the conference was already in single-image VAS mode when MCV took effect, the conference returns to single-image VAS mode.

The **status conference** command indicates whether the broadcaster is self-designated via MCV. Therefore, the reason why a Continuous Presence *Plus* conference transitions to full-screen video is traceable.

Interactions

The following interactions are in effect.

Audio Add-On

Conferences using Continuous Presence *Plus* support as many Audio Add-On ports per conference as conferences that do not use Continuous Presence *Plus*. Audio Add-On port capacity is shared among Continuous Presence *Plus* and non-Continuous Presence *Plus* conferences. There is one systemwide Audio Add-On port capacity. Therefore, Audio Add-On ports may be assigned freely to conferences that use Continuous Presence *Plus* and those that do not use this feature.

Speech detected from an Audio Add-On party (or an audio-only H.221 party) has no effect on the content of the quad image for single MCU conferences.

Audio Modes

Continuous Presence *Plus* conferences support the audio modes (including G.711, G.722, G.728) supported for non-Continuous Presence *Plus* conferences running at the same transfer rate.

BONDing

Endpoints in conferences with Continuous Presence *Plus* can use BONDing (Mode 1).

2 *64 and 4*64 BONDing and also 2*64 and 4*64 Multirate can be mixed in the same conference (Each endpoint receives video service.) Similarly, 6*64 BONDing and 6*64 Multirate/H0 can be mixed in the same conference with each endpoint receiving video service.

Cascading

CPP conferences, including Presentation CPP conferences, are supported on both primary and secondary MCUs.

Presentation mode is configured at the secondary MCUs to force the inter-MCU link to be the full-screen broadcaster at each secondary MCU. The return video at the secondary MCU(s) is therefore subject to VAS. This means that the primary MCU, which composes the CPP image, receives the video from the last or current speaking site connected to each secondary MCU.

The primary MCU does not include CIF capability in the capability set sent to the secondary MCU on the inter-MCU link. Therefore, the secondary MCU removes CIF capability from the capability set that it sends toward all its locally connected

endpoints. The secondary MCU also removes CIF capability from the inter-MCU link back to the primary MCU. Under normal circumstances, the primary MCU stops transmitting CIF video toward an endpoint or inter-MCU link whose advertised capability set does not include CIF. However, in a cascaded Continuous Presence *Plus* conference, the primary MCU sends CIF video to the secondary MCU over the inter-MCU link even if CIF capability is not included. The result is that all compliant endpoints connected to all secondary MCUs:

- Transmit QCIF, and can be a source for the CPP image composed by the primary MCU
- Receive the CIF CPP image from the primary MCU whenever the primary MCU is using VAS mode and Continuous Presence *Plus*.
- Receive the CIF full-screen image from the primary MCU whenever the primary MCU uses presentation mode and Continuous Presence *Plus*. (This assumes that the presenter is CIF-capable.) In presentation mode, the presenter (that is, the controlling extension) can transmit CIF, and the presenter's video is not subject to video processing by the MCU. This means that the presenter is viewed by all non-presenters at all MCUs in "full screen."

NOTE:
Secondary MCUs are not required to be current release MCUs running Continuous Presence *Plus* hardware and software to participate in a Continuous Presence *Plus* conference. The video mixed and distributed by the primary MCU is broadcast to all capable endpoints by the secondary MCU.

In cascaded Continuous Presence *Plus* conferences, only one site connected to a given secondary MCU at a time can be included into the composed Continuous Presence *Plus* image . At any point in time, the primary MCU is receiving exactly one QCIF source from each secondary MCU. Usually, this source is the current (or last) speaker connected to the secondary MCU. At the primary MCU, the four new quadrant administration fields may be populated with the MCU extension of a cascaded link to a secondary MCU.

In cascaded Continuous Presence *Plus* conferences, whenever an Audio Add-On party begins speaking at a secondary MCU, the content of the quad image changes if all of the following are true:

- VAS quadrant exists at the primary MCU.
- No one at the secondary MCU was speaking previously (the inter-MCU link was previously viewed by the primary MCU as silent).
- Video received from the inter-MCU link is not currently part of the quad image.

Conference Modes

Interactions involving specific conference modes are provided in the following sections.

Broadcast with Auto-Scan Mode

This mode is not supported in Continuous Presence *Plus* conferences.

H.243 Chair Control

All the existing H.243 Chair Control operations are supported in Continuous Presence *Plus* conferences, as follows:

- Requesting (CCA) or relinquishing (CIT) the chair token can occur regardless of whether the conference is currently in full-screen or CPP-mixed video mode.
- Requests for the Floor (TIF) are forwarded to the Chair Terminal regardless of whether the conference is currently in full-screen or CPP-mixed video mode.
- The specific site can be dropped by the chair (CCD <MCU> <Terminal>) regardless of whether the conference is currently in full-screen or CPP-mixed video mode. If the site dropped by the chair is “on-air,” the effect on the conference video is the same as for the endpoint voluntarily dropping.
- The entire conference (CCK) can be dropped.

  NOTE:
  The chair's dropping the entire conference does not release the reserved ports or Continuous Presence *Plus* resources from a scheduling perspective.

- The chair terminal may send VCB <MCU> <Terminal> to the MCU to force the video from terminal number *T* connected to MCU number *M* to be the full-screen broadcast video source. Whenever a Continuous Presence *Plus* conference is in full-screen mode, the chair terminal's VCB requests are honored, just as in non-Continuous Presence *Plus* conferences. The VCB “locking” effect on the full-screen broadcast video is broken if the following are true:
  - — Any unlocking action applicable to non-Continuous Presence *Plus* conferences occurs.
  - — Conference is transitioned back to CPP-mixed video operation via the MCU-MT, CRCS, or UCC interfaces (that is, if the conference is subsequently returned to full screen, the MCU need not “remember” and display the previously locked full-screen broadcaster selected via VCB).

The MCU automatically transitions a Continuous Presence *Plus* conference to full-screen operation whenever VCB is received while the conference is in CPP-mixed video mode. The **status conference** command output shows why the conference has transitioned to full-screen.

Presentation Mode

In presentation mode conferences with Continuous Presence *Plus*, the viewing experience at the non-presenter sites is the same as it would be without Continuous Presence *Plus* (that is, the controlling extension remains the video broadcaster for the duration of the conference). All the non-presenter sites see the full-screen image of the presenter at the CIF frame rate. This means that the presenter can be managed via capability exchange to transmit CIF video (assuming it is capable); Therefore, there is neither a loss of video resolution nor a lowering of the frame rate introduced by using Continuous Presence *Plus*.

The viewing experience at the presenter site is subject to the same video mixing as for a VAS conference supporting Continuous Presence *Plus*. The composition of the mixed image transmitted toward the presenter can be fixed via administration or UCC, or it can be left to voice-activated MCU control as in a VAS conference.

Similarly, UCC allows the presenter to transition to full-screen VAS switching of the return video.

Since the presenter must be CIF-capable, the presenter both transmits and receives CIF video as long as the presenter is viewing the Continuous Presence *Plus* image. Therefore, the endpoint interoperability concerns associated with QCIF/CIF asymmetry do not apply to the presenter as long as the presenter supports CIF video and continues to view the Continuous Presence *Plus* image. However, if the conference is toggled to full screen so that the presenter views the full-screen switched video from one other site, the presenter could be transmitting CIF while receiving QCIF.

NOTE:
If the conference is translated to upgrade capabilities on transition to full screen, it is possible that the presenter will again be able to send and receive CIF, depending on the capabilities of other endpoints in the conference.

Since the presenter is transmitting CIF video, the presenter site does not appear in the mixed quad image transmitted by the MCU to the presenter. The non-presenter sites are managed via capability exchange to transmit QCIF video and receive CIF video.

Endpoints that join the conference before the presenter receive the BAS code Video Indicate Suppress (VIS) from the MCU.

Voice-Activated Switching (VAS)

Whenever the conference mode is VAS, VAS can be used to determine the Continuous Presence *Plus* image seen by all sites. Whenever the conference mode is Presentation, VAS can be used to determine the Continuous Presence *Plus* image seen by the presenter.

The user may fix any or all of the quadrants via administration. The quadrants may or may not be subject to VAS. Whenever there is at least one VAS quadrant and normal human interaction, the following statements are true:

- If a site speaks, and if that site is already part of the mixed quad image, there is no change to the quad image. In other words, a site that is already on screen is not “rotated around the screen.”
- If a site speaks, and if that site is not part of the mixed quad image, the MCU replaces a nonspeaking site currently occupying a VAS quadrant with the new speaking site. However, given that the MCU will tend to have more than one quadrant from which to choose, the MCU can decide to do one of the following:
  - — Return the speaking party to the quadrant it had last previously occupied.

    NOTE:
    However, if two sites become engaged in a dialog, and if each site last occupied the same quadrant, the MCU allows both sites to appear on screen simultaneously rather than alternating them on screen in the same quadrant.

  - — Select for removal the site that has been quiet the longest

The methods in the previous list help minimize the likelihood that video switching via Continuous Presence *Plus* will be perceived to be disruptive or chaotic. Another factor to this end is the video holdover timer. This timer is applied to the entire quad image. As soon as any VAS-triggered change to the content of the quad image is made, the MCU sets the holdover timer. While the holdover timer is running, the MCU refrains from making any other VAS-triggered changes to the content of the quad image. Once the holdover timer expires, the queue of talkers is reexamined to determine if action is warranted.

Conference Reservation and Control System (CRCS)

CRCS supports conferences with Continuous Presence *Plus*. However, the following actions are required or suggested.

- CRCS scheduler must be updated to account for the greater resource consumption of Continuous Presence *Plus* conferences.
- CRCS must ensure that the number of conferences scheduled to use Continuous Presence *Plus* does not exceed the value administered on the System-Parameters Customer-Options screen. It is understood that CRCS is unaware of the status of the actual hardware resources supporting Continuous Presence *Plus*.
- CRCS user interface for reporting and resolving conflicts must be updated. The user interface for reporting resource exhaustion of Continuous Presence *Plus* conference capacity can follow the existing model (for example, using a red bar for a given time period along with the existing "detail" buttons to find out which resources are lacking).

The CRCS supports Dynamic Conference Resizing within conferences with Continuous Presence *Plus* via the real-time agent monitoring and control interfaces (such as OpCenter). Also, CRCS can use Dynamic Conference Resizing and OpCenter to allow a conference to toggle between full-screen switched video and Continuous Presence *Plus* mixed video. CRCS (along with the MCU) also prevents an ongoing conference without Continuous Presence *Plus* from being transitioned to a conference with Continuous Presence *Plus*.

CRCS supports the cascading of Continuous Presence *Plus* conferences to both MCUs controlled by CRCS as well as by external MCUs. (See the "Cascading" section.) For such conferences, CRCS is designed to present one conference mode choice to the agent. Whenever the agent enters **VAS** for a cascaded conference with Continuous Presence *Plus*, CRCS downloads the secondary MCU(s) as presentation mode regardless of whether the overall conference mode is VAS or presentation. The primary MCU is downloaded with the conference mode selected by the agent

Dedicated Access

Parties in conferences with Continuous Presence *Plus* may be connected via dedicated, nonswitched access.

Dial-In

This feature functions the same way in conferences with Continuous Presence *Plus* as it does in conferences without Continuous Presence *Plus*.

Dial-Out

This feature functions the same way in conferences with Continuous Presence *Plus* as it does in conferences without Continuous Presence *Plus*.

Direct Connect

Endpoints in conferences with Continuous Presence *Plus* can be directly connected to the MCU.

Dynamic Conference Resizing/Reconfiguration

The following fields on the Conference Record form can be used during an active conference to enable this feature:

- Each of the four conference record form fields controlling the Continuous Presence *Plus* picture composition
- Field controlling whether an existing Continuous Presence *Plus* conference is currently using full-screen switched video or CPP-mixed video.
- Field controlling whether a conference will schedule and configure Continuous Presence *Plus* resources. (This field can be changed from **yes** to **no** but not from **no** to **yes**.)
- Fields controlling the video resolution (QCIF/CIF) and frame rate (i.e., the video caps)
- Field specifying the video source to be shown in “empty quadrants.”

H.CTX/H.CTX+

Conferences with Continuous Presence *Plus* and SG4 are blocked via administration from supporting CLI's H.CTX/H.CTX+ video. Continuous Presence *Plus* is supported for conferences using H.261 or H.263 video only.

Low-/High-Speed Interworking

This feature is supported in conferences with Continuous Presence *Plus*.

Mixed Dial-in and Dial-out

Conferences with Continuous Presence *Plus* allow any mixture of dial-in and dial-out parties.

MultiRate Video (MRV)

This feature is not supported by Continuous Presence *Plus.*

Passwords

The procedures for password collection and authentication are unchanged for Continuous Presence *Plus* conferences.

SG4

Conferences with Continuous Presence *Plus* are blocked via administration from supporting PictureTel's SG4 video. Continuous Presence *Plus* is supported for conferences using H.261 or H.263 video only.

Speed Match

Continuous Presence *Plus* can be disabled within a CPP Speed Match conference. If this happens, the conference changes to a full screen Speed Match conference. Also, if CPP Speed Match ports are administered in the System Parameters Customer Options form, and if there are no CPP Speed Match conferences active at a given time, the appropriate number of MCU resources are freed up to allow one or more additional CPP-only or full screen Speed Match conferences.

Still-Frame Graphics

A Continuous Presence *Plus* conference currently operating using full-screen switched video allows high resolution still-frame graphics (H.261 Annex D) to be transmitted in full-screen. CLI Token-Controlled Still Frame Graphics (TC-SFG) can also be transmitted.

T.120 Data Conferencing

This feature is supported within non-SM CPP conferences at all fixed MLP rates between 6.4k and 46.4k.

Universal Conference Control (UCC)

The UCC controller can toggle a conference between CPP-mixed video operation and full-screen switched video. To accommodate scenarios where one site is the preferred (or required) viewed site (broadcaster), UCC can override VAS to ensure that this site remains the broadcaster despite speech detected from other sites.

If the UCC control port dials the access code for going full screen, all sites previously receiving the quad image begin to view the switched full-screen video from one other site.

The UCC commands for dropping a site or ending a conference operate within conferences currently viewing CPP-mixed video. Whenever a site that was previously part of the quad image is dropped via UCC, the effect on the conference video is the same as when an endpoint voluntarily drops.

Access to the commands for the Select Broadcaster, Browse, and Rollcall options of UCC are available in a Continuous Presence *Plus* Conference.

The UCC Set Quadrant option allows the UCC controller to place specific sites into specific quadrants. The sites to be included in the quad image are identified according to the number assigned via conference administration and appearing on the CRCS confirmation report.

NOTE:
UCC is not permitted in Chair conferences.

Like conference administration, UCC denies attempts to make the controlling extension in a presentation mode Continuous Presence *Plus* conference one of the four sites specified for the quad image. (The controlling extension is transmitting CIF video.)

In Continuous Presence *Plus* mode, the UCC Unlock Video option returns the conference to its previous behavior (that is, the conference returns to its effective operation mode before UCC was used). The meaning of the option is context-sensitive; it depends on whether the conference is currently viewing full-screen video or CPP-mixed video. Therefore, for example, if the UCC controller uses the Set Quadrant option to designate the sites appearing in the quad image, the Unlock Video option returns the conference to the quad composition specified by conference administration.

If the UCC port drops from a conference after specifying the Continuous Presence *Plus* composition, the specified composition is remembered and not automatically cleared. Other control interfaces can subsequently change the composition.

Administration

Continuous Presence *Plus* is administered via the Customer-Options form and the Conference Record form. Refer to "Administering Continuous Presence Plus" on page 2-221 for details.

More than four images are obtained by using existing standard CPP conference administration techniques. Conferences can be combined via Cascading or by initiating a dial-out to the central conference. The central conference encodes the final composition of seven, 10, 13, or 16 images. The Presentation mode conference ensures that the encoded video image from the central conference is seen by all participants.

The following table lists the maximum number of simultaneous conferences for the higher CPP transfer rates based on 3-party BONDed configuration. For CPP/SM, the basis is two parties at a higher rate and one party at the 2B rate.

Table B-14. CPP Conference Maximums Based on Conference Types and Transfer Rates

Conference Type (Option)	Transfer Rate	Maximum Number of Conferences
CPP	448/512k	7
	672/768k	5
CPP/SM	448/512k	8
	672/768k	6

The following table lists the maximum number of simultaneous CPP conferences and the maximum number of parties per CPP conference for all possible CPP image configurations.

Table B-15. CPP Conference Maximums Based on CPP Image Configurations

Image Configuration	Maximum Number of Conferences	Maximum Number of Parties Per Conference
4-image	8	25
7-image	4	48
10-image	2	72
13-image	2	96
16-image	1	120

Finally, CRCS and UCC can be used to control the composition and arrangement of the quad images.

Hardware and Software Requirements

Continuous Presence *Plus* requires a TN2237C, vintage 16 or higher Video Processor circuit pack. This circuit pack provides a full decode/encode technique.

The TN2237C circuit pack decodes video at any supported frame rate up to 30 frames per second (fps). The circuit pack supports the 1B, 2B, 4B, 6B, 8B, and 12B transfer rates.

The MCU allows users to view four, seven, 10, 13, or 16 participants simultaneously on the same screen. This requires multiple CPP conferences, each requiring a separate right-to-use, a separate VPP circuit pack, and a separate UCC port. For example, a 16-image conference requires at least five CPP conferences, five rights-to-use, and 5 VPP circuit packs. A 7-image conference requires two CPP conferences, two rights-to-use, and two VPP circuit packs.

Data Interfaces for Access Endpoints

Description

Several data interfaces allow endpoints either connected directly to the MCU or behind an encryption device or proprietary inverse multiplexor to participate in multipoint conferences at a rate in the range 112k to 1920k. These interfaces include EIA-530, RS449, and V.35.

Considerations

The Dedicated Access (DA) feature of the Lucent Technologies MCU allows T1, EIA-530, RS449, and V.35 connectivity. A DA endpoint type is used whenever the destination is intended to be an access endpoint. EIA-530, RS449, and V.35 connectivity is supported by the TN2236 Data Interface (DI) circuit pack. T1 connectivity is supported by the TN767 DS1 or TN2207 UDS1 circuit pack.

The Lucent Technologies MCU supports a maximum of 32 DI ports at transfer rates up to and including 1920k. The following DI port/rate maximums are supported: 16 ports at 2B, 1680k, and 1920k; 20 ports at 1008k through 1536k; 32 ports at 128k through 768k.

DI endpoints can interwork with BONDed, multirate, and DS1 DA endpoints. However, there is no integrated BONDing for DI traffic. Encryption devices and inverse multiplexors (IMUXs) must use DA to connect to the MCU via the data interface. Also, Direct Connect endpoints can connect to the MCU via the data interface.

Audio Add-On and Universal Conference Control (UCC) endpoints cannot access an MCU conference via the data interface. Finally, a Call Detail Record (CDR) is not generated for DI access-endpoints.

The MCU waits one minute before originating a call for any type of DA endpoint. This is done to allow codecs and IMUXs to disconnect completely from a previous call before connecting to a new conference. Once the call is originated, the MCU looks for H.221 framing continuously until the conference ends. Once framing is found, the usual framing recovery strategy becomes active. If framing is found and then lost, the system searches for framing for 40 seconds; if framing is not found again, the system drops the call. The call is then reconnected after one minute, and it resumes searching for framing indefinitely. The one-minute waiting period applies to all methods of originating a call (including redialing an endpoint, Dynamic Conference Resizing, adding a new conference, etc.).

Interactions

DI endpoints interact with the following features or capabilities with specific changes or considerations as noted.

- BONDing

  Integrated BONDing is not supported for DI traffic.

- Cascading

  A data interface can be used as a static cascade link between two Lucent Technologies MCU conferences.

Administration

Several forms are used to administer the data interface. Refer to the "Administering EIA-530, RS449, and V.35 Data Interfaces" section in chapter 2.

Also, Cascading is supported for all DA endpoint types. In the Conference Record form, the endpoint type should be administered as "CAS," and the access endpoint extension to be used as the physical cascade link should be included.

Hardware and Software Requirements

The TN2236B, vintage 5 or higher Data Interface circuit pack is required. Two such circuit packs are required for MCU endpoints operating at the 2B (112k or 128k) transfer rate.

DS1 Trunk Service

Description

DS1 trunk service provides a digital interface for the following:

- Robbed-Bit DS1 Tie Trunks
- ISDN-PRI Trunks
- CAS DS1 Tie Trunks

Tie Trunks

Robbed-Bit DS1 tie trunks permit data calling between the MCU and another switch, can connect the system with other digital switches, and can be used to dynamically access data SDN services.

The TN767D DS1 circuit pack is used to support Robbed-Bit DS1 tie trunks in the Robbed-Bit Signaling mode.

DS1 transmission facilities in the US and Canadian DS1 format is a 1.544-M digital signal consisting of a 1.536-M signal multiplexed with an 8-k framing signal.

E1 transmission facilities format is a 2.048-M digital signal and requires the TN2207 universal DS1 circuit pack. E1 supports CAS signaling used in the United Kingdom.

Considerations

Each DS1 circuit pack can support up to 24 trunks: 24 trunks for transmission in the Robbed-Bit Signaling mode or 23 trunks for clear data transmission, and one trunk to transmit signaling for the other 23 trunks in the signaling mode or ISDN-PRI signaling mode.

The system can support a maximum of 26 DS1 circuit packs.

The TN2207 circuit pack is used for E1 trunks to provide 32 ports and includes an E1 interface cable.

Interactions

None.

Administration

DS1 trunks are assigned on a per-line basis by the system administrator. The following items require administration:

- DS1 Circuit Pack — assign the circuit pack to the system before administration of the associated trunks.
- Synchronization Plan — administer to provide synchronization between the switch's DS1 circuitry and the digital facilities to which the switch is connected.
- Trunk Groups — associate the trunks to groups, if desired.

Hardware and Software Requirements

One TN767D DS1 circuit pack is required for every 24 trunks using Robbed-Bit Signaling or for every 23 trunks using ISDN signaling.

A TN768 circuit pack is required to provide synchronization for the DS1 trunks.

For E1 (32 port) trunks, use the TN2207 universal DS1 circuit pack.

No additional software is required.

Dedicated Access

Description

The Dedicated Access (DA) feature of the Lucent Technologies MCU allows T1, EIA-530, RS449, and V.35 connectivity of endpoints to the MCU.

Considerations

For T1 connections, each Dedicated Access endpoint is assigned a fixed number of DS0s (1 through 30) that are always identified with that endpoint. To connect in a conference, the endpoint uses one or more (depending on the conference bandwidth) of those assigned DS0s. Bandwidths of 1x56k, 1x64k, 2x56k, 2x64k, 112k, 128k, 168k, 192k, 224k, 256k, 280k, 320k, 336k, 384k, 448k, 512k, 672k, 768k, 1008k, 1152k, 1288k, 1344k, 1472k, 1536k, 1680k, and 1920k are supported for Dedicated Access.

Several data interfaces allow endpoints either connected directly to the MCU or behind an encryption device or proprietary inverse multiplexor to participate in multipoint conferences at a rate in the range 112k to 1920k. These interfaces include EIA-530, RS449, and V.35. For details regarding connectivity via data interfaces, refer to the “Data Interfaces for Access Endpoints” section in this chapter.

The MCU waits one minute before originating a call for any type of DA endpoint. This is done to allow codecs and IMUXs to disconnect completely from a previous call before connecting to a new conference. Once the call is originated, the MCU looks for H.221 framing continuously until the conference ends. Once framing is found, the usual framing recovery strategy becomes active. If framing is found and then lost, the system searches for framing for 40 seconds; if framing is not found again, the system drops the call. The call is then reconnected after one minute, and it resumes searching for framing indefinitely. The one-minute waiting period applies to all methods of originating a call (including redialing an endpoint, Dynamic Conference Resizing, adding a new conference, etc.).

Interactions

For T1 connections, Audio Add-On, cascade links, and BONDed facilities must be accessed via switched facilities but can participate in a conference if at the same bandwidth as the Dedicated Access endpoints.

For information on interactions involving connectivity via data interfaces, refer to the “Data Interfaces for Access Endpoints” section in this chapter.

Administration

Dedicated Access administration involves several forms and varies according to the type of connectivity required. Refer to the “Administering Dedicated Access” section in Chapter 2 for details.

Hardware and Software Requirements

EIA-530, RS449, or V.35 connectivity requires the TN2236B, vintage 5 or higher Data Interface (DI) circuit pack. T1 connectivity requires the TN767 DS1 or TN2207 UDS1 circuit pack.

Dial-Out

Description

The Dial-Out feature allows the MCU to place outgoing calls automatically to conference participants using an Administered Connection. This is in addition to the basic MCU capability that allows conferees to call the MCU to join a conference.

A conference convener can use the dial-out feature in order to convene a conference automatically. At the conference start time, the MCU automatically dials all conference participants simultaneously.

Considerations

To establish the dial-out links, the MCU automatically translates a dial-out administered connection for each conferee number. This administered connection remains in place until the conference is completed. When the conference is completed, the MCU automatically untranslates the administered connection and makes it available for reuse.

If one of the conference participant numbers is a cascade link (that is, another MCU), the dialing MCU attempts to connect to the second MCU first and then calls all the other endpoints simultaneously. If the dial-out call fails, the MCU uses the failure cause code to determine its next action depending on whether the link is a cascade or a non-cascade link and the type of failure. The number of retries pertaining to post connection (answer) failures is administered on the System-Parameters Features form.

Cascade Link Retry Strategy

The retry strategy that Dial-Out uses for a cascade link is as follows.

- If the failure is the first failure reported, retry immediately.
- If the failure occurred after a connection was established, retry the administered number of times, then raise a warning alarm on the administered connection and stop retries.
- For all other causes:

  If the alarm threshold has been reached, raise a warning alarm on the administered connection.

  Retry the connection in approximately one minute.

Since a cascade link can represent many conferees, the retry attempts for a cascade link are only halted after the administered number of post-connection retries are attempted. If the link is for a dedicated conference, the assumption is made that the link must always remain up. Therefore, retries are attempted no matter what the failure cause unless there is an administration mismatch between the two cascaded MCUs or the administered number of post-connection retries is reached.

Non-Cascade Link Retry Strategy

The retry strategy that dial out uses for a non-cascade link is as follows:

- If the failure is a far-end disconnect, stop trying to connect.
- If this is the first failure, retry immediately.
- If the failure occurred after a connection was established, retry the administered number of times, then raise a warning alarm on the administered connection and stop retries.
- For all other causes:

  Log an error in the hardware error log.

  If the alarm threshold has been reached, raise a warning alarm on the administered connection.

  Retry the connection in approximately one minute.

Interactions

The following features interact with the Dial-Out feature.

- Audio Add-On endpoints can be dialed out.

  Operator-Assisted Dial-Out allows you to enable or disable automatic dialout for both video and Audio Add-On endpoints on a per-endpoint basis. Enabled endpoints are dialed out automatically. Disabled endpoints are dialed out by an operator via the MCU-MT or an OpCenter or ResCenter user interface.

- AAR/ARS

  Dial out can use the AAR/ARS feature to route outgoing calls over the proper trunk facilities by including the AAR/ARS feature access codes in the destination number(s) of a conferee.

- BONDing

  Dial out can originate BONDed calls.

- Call-by-Call Service Selection

  Dial out can use the call-by-call service selection feature to route outgoing calls over specific ISDN-PRI networks.

- Cascading

  Cascading uses the dial-out feature to establish the inter-MCU link.

- CDR

  If the dial-out feature uses a trunk group with the CDR feature on, the origination extension (MCU extension), destination number, and conference billing ID are reported.

- Class of Restriction

  The system administrator can prevent stations connected to the MCU from making outgoing calls by administering the stations with an Outward Restriction COR.

- DS1 Trunk Service

  Robbed-bit trunks can be used as a network facility in routing dial-out calls.

- ISDN-PRI

  ISDN-PRI trunks can be used as a network facility in routing dial-out calls. In addition, the MCU extension is sent as the calling identifier if enabled on the trunk group form.

- Passwording

  Audio Add-On endpoints do not require a password to answer Dial-Out calls.

- System Measurements

  All trunks used for dial-out calls are measured as with any other trunk use.

- System Status Report

  The status AC command is supported for dial-out administered connections.

Administration

Administration for this feature is required for outgoing trunk groups and AAR/ARS routing patterns. Dial-out numbers must be administered for each conference where they are to be used. If the conference is a single-channel high-speed conference, a single dial-out number is administered. If the conference is a 112k/128k conference, two dial-out numbers are administered. In either case, if ARS or AAR are to be utilized in the calls, the access codes for those features should be the first digit(s) of the dial-out number(s).

The maximum number of post-answer retries must be administered in the ‘‘Dial-out Post Answer Failure Retry Attempts’’ field of the System-Parameters Features form.

Audio Add-On Endpoint Dial-Out

You can administer Audio Add-On endpoint Dial-Out via the MCU-MT or CRCS. From the MCU-MT, this is done via endpoint administration; from CRCS, this is done via the site options page. Administration is done by using an administered connection.

Operator-Assisted Dial-Out

Populate the *(Re)dial* field in the Conference Record form, as appropriate, to enable or disable Operator-Assisted Dial-Out. For example, leave the field blank if you want to assign automatic Dial-Out capabilities to the associated endpoint. Also, if you set the *(Re)dial* field to **y** during an active conference, and if the previous field value was **n**, the call is dialed out to the endpoint for the first time; otherwise, the Dial-Out call to the endpoint is redialed.

Hardware and Software Requirements

Dial-Out requires MCU software release 1.1 or later. The feature also requires the TN744C Call Classifier circuit pack.

Dial Plan

Description

This feature enables the administration of the video port extensions used within the MCU to identify and route calls to video ports. It also enables the administration of extensions for entities such as the maintenance alarm terminal and an CDR printer port extension. In addition, feature access codes are defined by this feature.

Each video port is identified by a unique extension. As an example, this extension is used on the Conference Record Form and within the maintenance subsystem to identify the appropriate video port associated with a given call.

The dial plan, or first-digit table, established during administration for each system, provides information in conjunction with the AAR features on the MCU on what to do with digits received from the network or internal sources such as the maintenance alarm terminal. The following parameters define extensions and feature access codes (used internally by the MCU or the maintenance alarm terminal).

- Extension numbers

  Used to define video port extensions, maintenance alarm terminal extensions, and other extensions when applicable.

- Feature access codes

  A minimum of one digit is required but a maximum of three digits can be used. The * and # buttons can be used as the first digit of a feature access code.

- Trunk access codes

  A minimum of one digit. The digit can be any number in the range 1 through 9. For example, you could set 9 for local trunks and 8 for tie trunks.

Considerations

None.

Interactions

This feature controls extension number formats for video port and other extensions. It must be completed prior to using extensions and/or feature access codes on the system.

Administration

The system administrator administers the dial plan on a per-system basis. The following administration is required:

- Area code where the MCU is located (if in the United States). Blank if the MCU is located in Australia, Singapore, or the United Kingdom.
- Enter ‘‘y’’ in the “Uniform Dial Plan” field

Hardware and Software Requirements

No additional hardware or software is required.

Dynamic Resizing

Description

This feature allows a reservations agent to add and remove endpoints from a conference both before and during the conference. This capability includes the cascade links along with the Audio Add-On, dial-out, and Meet-Me endpoints. Using this feature, dial-out numbers can be removed and changed, and conference stop times can be changed. In addition, this feature allows dial-out numbers to be changed to dial-in numbers (and vice versa) and primary/secondary MCU types to be changed in a cascade link. It also allows the basic/enhanced service flag, redial flag, and interworking status during a conference to be changed.

Considerations

Dynamic Resizing is capable of using existing MCU resources only. The feature can also be used to deallocate Continuous Presence *Plus* resources during an active conference. If the *Continuous Presence* field in the Conference Record is set to “no,” these resources are released. The *Quad 1* through *Quad 4* fields and the *Empty Quadrant* fields on the form can be changed during an active conference.

Also, the feature can be used to dynamically disable a conference that is enabled with Executive Conferencing. Also, dynamic changes can be made to the associated *Site 1-3* and *Force Video Now To* fields in the Conference Record form.

The feature cannot be used to allocate Continuous Presence *Plus* or Executive Conferencing resources in an active conference.

The video mode of an active conference can be changed from and to any of the “H.261 types,” including “H.261-down-only,” “H.261-w/upgrade,” “H.261-CIF-Floor,” “H.261-CIF-Now,” and “H.261-QCIF-Now.”

NOTE:
Be sure to refer to "ClearPresence @716 Audio Coder" on page B-63 for details regarding how the feature operates based on whether or not the ClearPresence @716 RTU is optioned.

Interactions

The Dynamic Resizing feature interacts as described in the individual feature descriptions. In particular, be sure to refer to "ClearPresence @716 Audio Coder" on page B-63 for details regarding how the feature operates based on whether or not the ClearPresence @716 RTU is optioned.

Administration

No additional administration is necessary to activate the Dynamic Resizing feature.

Hardware and Software Requirements

No additional hardware or software is required. The feature is available on all MCU models.

Executive Conferencing

Description

Executive Conferencing is a customer option that enables a conference to compose a picture of two or three “other” sites (endpoints) for each conference participant. With this feature, no endpoint ever sees itself.

Considerations

Executive Conferencing is provided with a right-to-use (RTU) fee for the customer.

With this feature, panoramic images of two or three other sites are seen by all of the participants in a conference. These sites are selected from three defined eligible video sites for the conference, and they are defined as video broadcaster sites. Each broadcaster site sees and hears the other two broadcaster sites in a dual panoramic image; all of the other video sites see and hear all three broadcaster sites in an upper-panoramic, tri-screen image. However, all of the sites in a conference that are *not defined* as a broadcaster site are not eligible video sources for the conference.

An illustration of the feature is provided in the following figure.

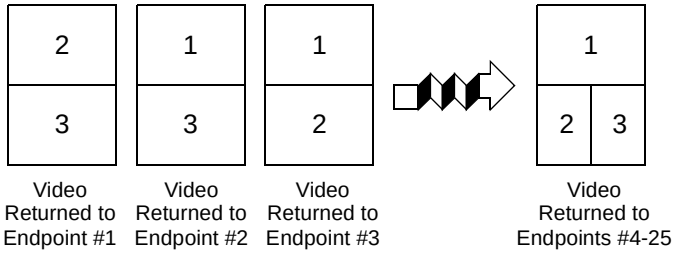

Figure B-7. Executive Conferencing

In this figure, sites #1, #2, and #3 are administered as the three eligible video sites (or broadcasters) for the conference. The remaining video sites (#4 through #25) in the conference can listen to return video but are not themselves eligible video sources for the conference.

For sites #1 through #3, the Executive Conferencing display is a dual-panoramic, no-self-view image of the other two sites. As the conference size grows beyond three endpoints, the additional video sites see an upper-panoramic, tri-image view of the three broadcaster sites.

This Executive Conferencing algorithm is based on the following rules.

- Endpoint designated as site #1 (for example, a CEO) is the main broadcaster
- All endpoints must see the site #1 (the main broadcaster) at all times
- All endpoints should see the present speaker from the eligible video sites #1, #2, and #3
- No site can ever see itself

The feature supports the 1x56k, 1x64k, 2x56k, 2x64k, 112k, 128k, 224k, 256k, 336k, 384k, 448k, 512k, 672k, and 768k video rates. The maximum received video frame rate in a conference with Executive Conferencing is 15 fps.

The user can transition to the Executive Conferencing dual-panoramic screen modes from full-screen mode or vice versa by using any of the following: UCC *Star*, UCC *Web*, MCU control commands, or OpCenter control commands.

Interactions

Executive Conferencing does not support the following: Broadcast w/Auto-Scan Mode, FECC, H.263/H.261 Transcoding, On-Screen Display, Presentation Mode, Speed Match, UCC P&C, the CTX, CTX+, MRV, and SG4 video modes, and the receipt of 30 fps with the multi-image mode enabled.

The following features and capabilites are supported with the considerations noted:

- Cascading

  Used with this feature, Executive Conferencing provides various video and audio “experiences” that differ from those provided by Cascading used with Continuous Presence *Plus.*

- Dynamic Resizing

  Executive Conferencing conferences can be dynamically disabled. Also, video sources for Executive Conferencing can be reassigned as eligible broadcasters during an active conference by administering the associated *Site 1-3* and *Force Video Now To* fields in the Conference Record form. However, Executive Conferencing resources cannot be dynamically enabled.

- Voice Activated Switching

  VAS within the panoramic panels is not supported.

Administration

For details on administering Executive Conferencing, refer to "Administering Executive Conferencing" on page 2-230.

Hardware and Software Requirements

One TN2237C, vintage 16 or higher Video Processor circuit pack per conference with Executive Conferencing is required.

One software Right-to-Use (RTU) for each active conference with the feature is required. This RTU can be purchased individually as an Executive Conferencing RTU or acquired via the Conferencing Suite RTU.

Facility and Non-Facility Associated Signaling

Description

This feature provides signaling for ISDN-PRI.

Facility Associated Signaling

Facility Associated Signaling (FAS) allows an ISDN-PRI DS1/E1 Interface D-channel to carry signaling information for only those B-channels located on the same DS1/E1 facility (circuit pack) as the D-channel.

Non-Facility Associated Signaling

Non-Facility Associated Signaling (NFAS) allows an ISDN-PRI DS1/E1 Interface D-channel (signaling channel) to convey signaling information for B-channels (video channels) on ISDN-PRI DS1/E1 facilities other than the one containing the D-channel. As a result, a D-channel can carry signaling information for numerous B-channels located on different DS1/E1 facilities. The MCU supports up to 20 T1s per NFAS group.

D-channel Backup

To improve reliability in the event of a signaling link failure, a backup D-channel may be administered. If a signaling link failure does occur, a switch to a backup D-channel will then take place.

D-channel Backup requires that one D-channel be administered as the Primary D-channel and that a second D-channel be administered as the Secondary D-channel. These assignments ensure that at certain times during D-channel Backup procedures that both D-channels are in the same state. This avoids the occurrence of both switches at each end of the DS1/E1 interface selecting the same D-channel to be put into service. In these cases, the Primary D-channel is given precedence over the Secondary D-channel.

The following figure shows a possible configuration involving three ISDN-PRIs between the MCU and a PBX or the public network. With DS1 (24 channel) interfaces, two of the ISDN-PRIs contain a D-channel and 23 B-channels, while the other ISDN-PRI contains 24 B-channels. One of the D-channels is the Primary D-channel, and the other is the Secondary D-channel. Together, this pair of D-channels will signal for all 70 (23+24+23) of the B-channels that are part of the three PRIs.

Since the D-channels are signaling for more than one DS1/E1 facility, the D-channel Backup feature requires the use of the NFAS feature. At any given time, one of the two D-channels will be carrying layer 3 signaling messages, while the other D-channel will be active at layer 2, but in a standby mode only. Any layer

3 messages received over the standby D-channel will be ignored. The two D-channels can provide signaling for only a predefined set of B-channels and cannot dynamically back up other D-channels on other interfaces.

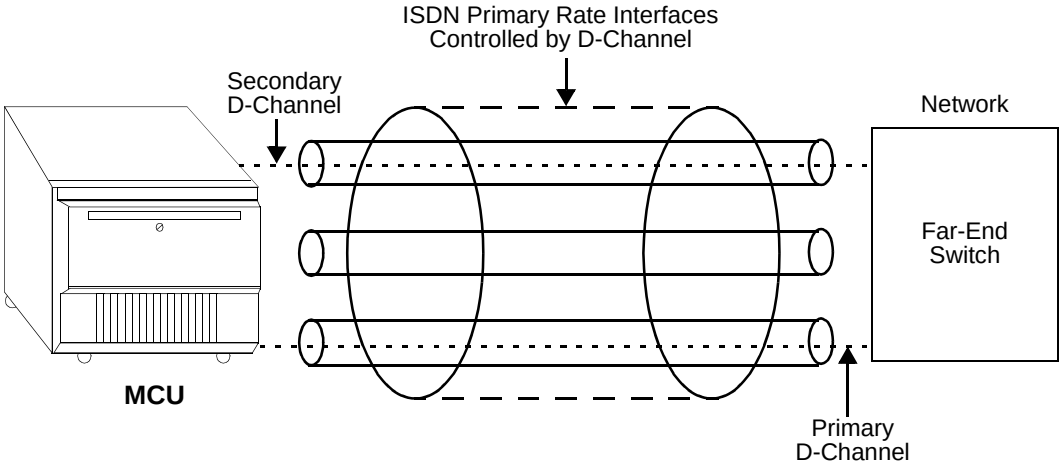

Figure B-8. Example D-channel Backup with Three ISDN-PRIs

D-channel Backup Activation

D-channel Backup can be invoked in response to the following events:

- D-channel Failure

  If the signaling link fails on the active D-channel (D1) or the hardware carrying D1 fails, then the system will send a message over the standby D-channel (D2), which requests that D2 become the active D-channel. D2 then becomes the active D-channel and will carry all subsequent signaling messages. When the signaling link or hardware on D1 recovers from the failure, D1 becomes the standby D-channel.

- System Technician Commands

  If a system technician command requests that a D-channel switch-over take place, the first action taken by the system will be to tear down the signaling link on D1. After this has been completed, a message is sent on a D2 to request that D2 become the active D-channel. D2 then becomes the active D-channel and the switch-over will be completed.

Considerations

NFAS allows four H0 calls on a DS1/E1 that does not contain a D-channel.

Only those ISDN-PRI facilities that use the NFAS feature will be capable of providing the D-channel Backup feature. The reason for this limitation is that the two D-channels must be located on different PRI DS1/E1 facilities. As a result, the D-channels must support NFAS so that they can signal for B-channels on different ISDN-PRI DS1/E1 facilities.

When a transition from one D-channel to another occurs (D-channel Backup is activated), all stable calls (calls that have been answered already) will be preserved. Some messages may be lost, resulting in a loss of call-related information, but the calls themselves will be maintained. The effect of the transition on unstable calls (those that have not been answered yet) is unpredictable since the results depend on which messages were lost and the contents of those messages.

NOTE:
Currently, NFAS is not used much on public networks outside of the US and Canada. Also, NFAS can work on private networks using most country codes.

Interactions

None.

Administration

The following provisioning and administration must be considered when implementing FAS and NFAS. Coordinate the following with the far-end switch for the DS1/E1 facilities to be used:

- Decide which DS1/E1 facilities will use FAS.
- Decide which of the remaining DS1/E1 facilities will carry D-channel signaling information on the 16th (E1) or 24th (DS1) channel. For those channels that have a D-channel Backup, D-channel pairs must be allocated.

- Define Signaling Groups (1 through 8). A Signaling Group is a group of B-channels for which a given D-channel (or D-channel pair) will carry the signaling information. Each Signaling Group must be designated as either a FAS or NFAS Signaling Group.
  - — A FAS Signaling Group must contain all the ISDN B-channels on the DS1/E1 interface associated with the group's D-channel, and cannot contain B-channels from any other DS1/E1 circuit pack. For 24-channel DS1 boards, some of the DS1 ports may use in-band (robbed-bit) signaling and be members in a tie trunk group rather than an ISDN trunk group. These tie trunks cannot be members of a Signaling Group.
  - — There is no restriction on which DS1/E1 ports can belong to an NFAS Signaling Group. Normally, an NFAS Signaling Group consists of one or two D-channels and several complete DS1/E1 interfaces.
- An Interface ID must be assigned to each DS1/E1 facility in an NFAS Signaling Group. For example, if the B-channels in a Signaling Group span three DS1/E1 facilities, a unique Interface ID must be assigned to each of the three DS1/E1 facilities. This designation is required to uniquely identify the same B-channel (port) number on each of the DS1/E1 facilities in the signaling group. Therefore, this designation must be agreed upon by both sides of the interface and administered prior to initialization.
- D-channel Backup involves two or more ISDN-PRI DS1/E1 facilities that interconnect the MCU to a PBX or to the network. Two D-channels must be present on the facilities. One of the D-channels is designated as Primary and the other as Secondary. This designation must be agreed upon by both sides of the interface and administered prior to initialization.

Hardware and Software Requirements

See the Integrated Services Digital Network — Primary Rate Interface feature description, elsewhere in this chapter, for hardware and software requirements.

Far-End Camera Control (FECC)

Description

FECC allows a site that requests far-end camera control to control the camera of the far-end site that it is currently viewing.

Considerations

The MCU provides FECC over a low-speed data (LSD) channel at 6.4k. A site gains access to a far-end camera via LSD token handling. An FECC-enabled conference declares and exchanges H.224-LSD commands and capabilities for an LSD channel with the participating sites. If an endpoint opens LSD but does not have H.224 capability, the token request is denied.

Whenever a 6.4k and H.224-capable terminal joins an FECC conference, the MCU opens a 6.4k LSD channel for the terminal. The terminal then requests an LSD token. If an LSD token is available, the MCU grants it. Assuming this happens, the terminal then broadcasts a list of H.224 protocols and applications over the LSD channel to all the other terminals in the conference. The terminal releases the token when it is finished with the FECC operation.

Whenever a terminal pushes the button for FECC, the request for an LSD token is passed, and the MCU grants the token. The terminal broadcasts the H.224 “packet” carrying the H.281 FECC command to control a far-end camera. The H.224 packet contains the terminal number for the camera that needs to be controlled. The terminal that matches the terminal number in the H.224 packet responds to the FECC command. The terminal then releases the token.

The MCU never forces the terminal to release a token. However, the MCU grabs the token whenever any of the following applies to the endpoint: it is holding a token and drops; it becomes a secondary audio-only endpoint; or, it moves to another conference.

Endpoints that do not support 6.4k LSD in an FECC-enabled conference become audio-only endpoints. If an endpoint can open 6.4k LSD, the Selected Communication Mode (SCM) provides the endpoint with audio and video capabilities. If, after the endpoint opens the LSD channel, the endpoint has not declared H.224 capability or does not open the H.224 data link, FECC is not made available to the associated terminal. However, the endpoint continues to receive audio and video.

FECC requires “enhanced” endpoints that support terminal numbers. Specifically, the feature requires at least one non-basic endpoint for the MCU to exchange H.224 commands and capabilities; also, it requires at least two non-basic endpoints for any FECC operation.

FECC is enabled via administration on a per-conference basis. It is available only in non-T.120 Data Conferencing and non-CPP conferences. A currently viewed site is accessible for far-end camera control via any of the MCU conferencing modes (including VAS, Presentation, etc.).

The feature is standard for all Lucent Technologies MCU models that support Chair Control.

Interactions

FECC does not support T.120 Data Conferencing, Cascading, Continuous Presence *Plus,* or Executive Conferencing.

The following features or capabilities interact with FECC with the specific changes or considerations as noted.

- Dynamic Conference Resizing

  FECC can be disabled in an active conference; however, it cannot be enabled dynamically. If FECC is disabled dynamically during a conference, the LSD channel is closed and the SCM is reset excluding LSD as one of the common modes.

  FECC-enabled non-basic endpoints can be dynamically added to or removed from FECC-enabled conferences. Added endpoints of this type have full FECC functionality. Removing an FECC-enabled endpoint does not remove FECC functionality for the rest of the endpoints in the conference; even if the last remaining FECC-enabled endpoint is removed, the LSD channel remains open.

- Moving Endpoints

  Whenever a terminal that has the LSD token in an FECC conference is moved to another conference, the MCU takes control of the token for the original conference and makes it available to other terminals in the original conference. The LSD channel for this endpoint is closed.

  Whenever an FECC-enabled terminal is moved to an FECC-enabled conference, the terminal opens the LSD channel in the destination conference and can implement FECC there.

Administration

To enable the feature, access the Conference Record form and enter **y** into the *Far End Camera Control* field. Also, ensure that both the *Continuous Presence* field and the *T.120 Data Mode* field are set to **n**, and that the Cascade Mode field is blank. (Basic/Enhanced?: Required value: **enhanced** or **h263_enh**)

NOTE:
Value **basic** can be administered for endpoints in FECC conferences. However, these endpoints do not have access to FECC.

Hardware and Software Requirements

The TN787K, vintage 6 or higher MMI circuit pack is required.

R6.0 or greater MCU software is required. Customers with R4.x MCU software must upgrade first to R5.0 MCU software, then to R6.0 MCU hardware and software, etc.

FECC requires “enhanced” endpoints that support terminal numbers. Specifically, the feature requires at least one non-basic endpoint for the MCU to exchange H.224 commands and capabilities; also, it requires at least two non-basic endpoints for any FECC operation.

H.261 Annex D Still-Image

Description

With this feature, you can transmit still images in the video channel of the H.221 format based upon procedures specified in H.261 Annex D. Still images can be transmitted at up to four times that of CIF images.

Considerations

An endpoint usually uses this feature in conjunction with the “See-Me” feature (Contributor mode) by first requesting “See-Me” control, then waiting a half a second, sending the still image, and finally relinquishing “See-Me” control. This specific process transmits a complete still image to other endpoints in a VAS conference, and it avoids disruptions.

Interactions

This feature interacts with H.320 terminals that declare H.261 still image capability in the cap set. The feature does not support Speed Match.

A Continuous Presence *Plus* conference currently operating using full-screen switched video allows high resolution still-frame graphics (H.261 Annex D) to be transmitted in full-screen. CLI Token-Controlled Still Frame Graphics (TC-SFG) can also be transmitted.

Administration

None.

Hardware and Software Requirements

Consult your Lucent Technologies service representative.

H.263 Video Mode and H.263/H.261 Transcoding

Description

H.263 video mode is a standard feature available in non-speed-matched full-screen or CPP conferences. As an option, H.263/H.261 transcoding is available for full-screen, non-CPP, non-SM conferences.

H.263 video mode allows the following:

- H.263-compliant endpoints to participate in a non-speed-matched CPP or full-screen multipoint multimedia conference
- H.263 endpoints to interwork with H.261 endpoints in the same full-screen conference.

Considerations

H.263 video mode is a standard feature on all Lucent Technologies MCU models.

H.263 video mode is available on a per-conference basis. Therefore, the system can enable or disable this mode on a per-conference basis. The system supports H.263 video conferences and H.261 conferences that are operating simultaneously. The current video mode administration supports the “H.263/1_down only” and “H.263/1 w/upgrade” values to enable selecting H.263 video mode in a conference.

H.261 and H.263 capabilities are hierarchical; therefore, any endpoint that declares H.263 capability also declares H.261 capability.

NOTE:
An “enhanced” administered endpoint is expected to be an H.263 endpoint. However, if the endpoint does not declare H.263 capabilities, the endpoint can actually be an H.261 endpoint.

The system supports H.263/H.261 transcoding in a non-CPP full-screen conference. H.263/H.261 transcoding requires a right-to-use (RTU) fee. In a heterogeneous conference with a mix of H.263 and H.261 endpoints, the user has the transcode between H.261 and H.263 video to allow the endpoints to interwork. (See the Administration section.)

If the conference video mode is administered with one of the H.263 modes, the MCU exchanges the H.263 capabilities only with “enhanced” endpoints (that is, endpoints that can support the Multibyte Extension (MBE) capability). The endpoint must announce MBE capability before the MCU will exchange the H.263 capability set with the endpoint. The MCU will send H.263 capabilities in the cap set only for H.263 full-screen non-SM conferences. For all other conferences, such capabilities will not be sent towards the endpoints.

The TN2237C VPP circuit pack supports H.263 video mode, but it does not support H.263 capabilities and the baseline H.263_4CIF and H.263_16CIF video formats (for example, arithmetic coding and Advanced Motion Prediction). For any H.263 conference that requires the VPP circuit pack, the capabilities and the H.263_4CIF and H.263_16CIF video formats are not sent as part of the H.263 capset. Such H.263 conferences include basic CPP conferences and full screen conferences with H.263/H.261 transcoding.

For the H.263 down only value, the conference SCM video mode starts at H.263. If all the endpoints declare H.263 capability, the SCM video remains at H.263. If any one endpoint does not declare H.263, the SCM video mode is downgraded to H.261 and remains at H.261 for the rest of the conference.

The following list discusses the implications of the H.263/1 down-only value:

- If all the endpoints declare H.263 capability, the SCM video is H.263; otherwise, the SCM video for the conference is set to H.261. (The H.263-capable endpoints operate with H.261 video.)
- If the video mode is set to H.263/1 down-only, and if an endpoint that is not administered as “enhanced” joins the conference, the SCM video is downgraded to H.261.
- If the last endpoint that joins a conference causes the downgrade of SCM video to H.261, the SCM video remains at H.261 for the duration of the conference.
- For conferences with H.263 or H.261 video mode, the automatic selection of common video format and frame rate for the conference is set to the highest common of these selections among the endpoints
- If the last endpoint joining a conference causes the downgrade of the SCM video format and frame rate, the SCM video format and frame rate remains downgraded for the rest of the conference.

The H.263/1 w/upgrade value is supported for full-screen conferences only. All of the information in the previous bullet item applies to this value. In addition, if an endpoint that caused a downgrade for video from H.263 to H.261 drops out of conference, the SCM video is upgraded to H.263. Also, if the last endpoint that joined a conference and caused the downgrade of SCM video format and frame rate drops out of the conference, the SCM video format and frame rate are upgraded to the highest common of these selections.

H.261- and H.263-capable endpoints can interwork in the same non-CPP, non-SM full-screen conference whenever H.263/H.261 transcoding is enabled.

The H.263/1 down-only and w/upgrade video mode settings in an H.263/H.261 transcoded conference apply down-only or upgrade rules only for the associated video format and frame rate. Moreover, the rules are applied separately to H.263 and H.261 endpoints.

There will be two sets of highest common parameters (SCMs); one set is for the H.261 endpoints, and the other set is for H.263 endpoints.

H.263/H.261 transcoded conferences support the same transfer rates that are supported for full-screen Speed Matching but at a maximum of 768k.

Interactions

The following features or capabilities are not supported by H.263 video mode or H.263/H.261 transcoding: H.261 Annex-D Still Frame Graphics and Speed Match.

- Conference Control Commands

  A conference control command can be used within homogeneous H.263 conferences to transition between CPP quad screen video and full-screen video. A command can also be used to change H.263 video mode to any fixed H.261 video.

  NOTE:
  This also disables H.263/H.261 transcoding.

- Conference Modes

  All conference modes, except CPP broadcast mode, are supported.

- Continuous Presence *Plus*

  Only H.263/1 down-only video mode is supported in a CPP conference. H.263/1 w/upgrade video mode and H.263/H.261 transcoding are not supported in a CPP conference.

- Dynamic Conference Resizing

  Endpoints can be added, removed or changed dynamically in an H.263 conference or an H.263/H.261 transcoded conference. However, dynamically adding or removing an endpoint may cause a change to the conference video mode because the SCM video is computed every time an endpoint joins or drops from a conference.

The video mode can be changed dynamically from H.263 (down-only or w/upgrade) video mode to any H.261 mode.

NOTE:
This disables H.263/H.261 transcoding (if it was enabled) and deallocates the associated video resources (if they were allocated).

Dynamically changing H.261 video mode to H.263 video mode is allowed only in full-screen (non-CPP, non-SM, non-transcoded conferences).

H.263/H.261 transcoding can be disabled—but not enabled—dynamically. Whenever this capability is disabled, the video mode is changed to the highest common video mode. (If all the endpoints have H.263 video mode, the highest common video mode is H.263; otherwise, it is H.261.)

- Universal Conference Control

  UCC can be used within homogeneous H.263 conferences to transition between CPP quad screen video and full-screen video.

Administration

H.263/H.261 video transcoding is available on a conference basis. The feature is administered via the Customer-Options form and the Conference Record form. Refer to "Administering H.263/H.261 Transcoding" on page 2-240 for details.

Hardware and Software Requirements

H.263 conferences require the TN787K, vintage 6 Multimedia Interface circuit pack or higher. H.263/H.261 transcoding requires a TN2237C, vintage 16 or higher Video Processor circuit pack or higher.

If the TN2237C circuit pack either goes out of service during an H.263/H.261 transcoded conference or is not available at the start of such a conference, the conference video is changed to the highest common video mode.

H.CTX Video Mode

Description

The MCU uses the Standard Proprietary Video feature to support Compression Labs Incorporated (CLI) endpoints using H.CTX and H.CTX+ proprietary video protocols in a standards format. The MCU also supports these protocols via CLI's extension to the ITU-TSS H.230/H.243 conferencing standards, and it provides the proprietary video within the H.221 frame structure.

Considerations

H.CTX/+ is available on all models. The range is 16 to 96 ports in 16-port increments.

H.CTX and H.CTX+ are controlled via administration of a port capacity limit. Each protocol is available in 16-port increments and has a maximum capacity of 96 ports.

The MCU can simultaneously support H.261 video and standard proprietary H.CTX/+ video on separate simultaneous conferences. (Two different modes within one conference result in the noncompliant endpoint being audio-only.)

Interactions

The following features interact with the CLI extension features:

- Token-Controlled Still-Frame Graphics

  Token-Controlled Still-Frame Graphics (TC-SFG) transfers are supported in H.261, H.CTX, and H.CTX+ conferences as defined in the CLI Extensions to TSS Conferencing Standard Specification. Still-Frame Graphics allow any site to send graphics from any video source by requesting, granting, and releasing "graphic tokens."

  TC-SFG is available only when the MCU is operating in VAS or Chair mode. In an H.CTX or H.CTX+ conference, once the MCU makes an endpoint that has requested the "graphics token" the video broadcaster, the MCU grants the token to that endpoint. While the token is assigned, and after the switch to the token requester, the MCU prevents Video Command Updates (VCUs) from being transmitted to the codecs. This effectively causes the Still-Frame Graphics transfer to occur without "switching" the video streams. The MCU sets a timer while the token is assigned. If the timer expires before the endpoint releases the token, the MCU reclaims the token and returns to the previous state.

Whenever an endpoint releases the “graphics token,” the MCU returns to the previous broadcast and return video sources, and it restores full-motion video. The MCU reclaims the token whenever the following happens while the endpoint is holding the token:

— Chair or Universal Conference Controller designates a video broadcaster

— Endpoint drops

If the conference is in Presentation or Broadcast with Auto-Scan mode, the MCU acknowledges graphics transfer requests only from the controlling mcu-extension. TC-SFG requests from all other endpoints are rejected. If the Chair or a UCC endpoint selects the broadcaster, the TC-SFG request is rejected unless it originates from the selected video source.

The MCU supports TC-SFG in cascaded conferences.

- Site Names

  The MCU supports the use of nonstandard capabilities, commands, and procedures to allow CLI endpoints access to terminal name information collected via H.243 standardized procedures. These procedures allow an endpoint to request the terminal name associated with another endpoint in a conference. The response is returned in another nonstandard message.

  To access this feature, the MCU must announce support of the SITE-NAMES capability in a CLI nonstandard capability set. If the endpoint is recognized as a CLI site, the MCU announces support of SITE-NAMES. Such support also enables TC-SFG.

  Whenever a codec capable of handling SITE-NAME joins the MCU, the MCU forwards the terminal names of any currently connected endpoints to the new site. If the codec signals support for unsolicited SITE NAME receipt, the MCU sends terminal names to that site once it receives them from other sites.

- H.261 Video

  Simultaneous support for H.CTX, H.CTX+, and H.261 video conferences on the same MCU is supported.

- Dynamic Conference Resizing

  This feature is supported in standard proprietary video conferences.

- Cascading

  An H.CTX or H.CTX+ conference can be cascaded with another similarly configured conference on a second MCU.

- Passwords

  Conferences with H.CTX/+ video may be password protected.

- T.120 Data Conferencing

  T.120 Data Conferencing may be enabled on H.CTX/+ conferences.

- Universal Conference Control

  An H.CTX/+ conference supports this feature.

- Basic/Enhanced Service Flag

  In an H.CTX/+ conference, the MCU must transmit MBEs to signal support for proprietary video and to turn on the video. This is required to enable TC-SFG and SITE-NAMES support.

Administration

To establish an H.CTX (or H.CTX+) conference, enter **H.CTX** (or **H.CTX+**) into the *Video Mode* field of the Conference Record form. H.CTX is supported for bandwidths between 112k and 384k, inclusive. H.CTX+ is supported for bandwidths between 336k and 1920k, inclusive.

H.CTX/+ port capacity is administered separately from the total MCU port capacity. The MCU can be administered with an H.CTX/+ port capacity greater than the system capacity; however, the system port capacity takes precedence. If the total MCU port capacity is 12, and if the H.CTX/+ port capacity is 16, only 12 ports can be active simultaneously. Otherwise, the system would support a combination of H.320 and H.CTX/+ ports up to the system capacity (although the number of H.CTX/+ ports would be limited by the administered H.CTX/+ capacity.

Hardware and Software Requirements

None.

Integrated Services Digital Network — Primary Rate Interface

Description

This feature allows connection of the system to an Integrated Services Digital Network (ISDN) by using a standard ISDN frame format called the Primary Rate Interface (PRI). The ISDN provides access to a variety of public and private network services and facilities. The MCU ISDN-PRI implementation is consistent with the International Telecommunications Union-Telecommunications (ITU-T) Recommendation Q.931 and Q.921.

The T1 is a digital transmission standard that in North America carries traffic at the digital signal level-1 (DS1) rate of 1.544 M. T1 facilities are also used in Japan and some Middle-Eastern countries. It consists of a 1.536-M signal multiplexed with an 8k framing channel. The 1.536-M signal is divided into 24 channels (DS0s) of 128k each, numbered 1 through 24. The ‘‘D’’ (data) channel multiplexes signaling messages for the ‘‘B’’ (bearer) channels carrying data. When a D-channel is present, it occupies channel 24.

The E1 is a digital transmission standard that carries traffic at a rate of 2.048 M. It is used in Europe and elsewhere. The E1 facility is divided into 32 channels (DS0s) of 64 kpbs information numbered 0-31. Channel 0 is reserved for framing and synchronization information. When a D-channel is present, it occupies channel 16.

ISDN-PRI signaling in the system is supported by the TN767D (for 24 channels). The D-channel (signaling channel) is switched through the TN765 circuit pack.

With the ISDN-PRI, the system can interface with a wide range of other products including network switches and PBXs. These products include the following:

- Public Network switches (for example, 4ESS, 5ESS®, Northern Telecom™ DMS250, etc.)
- DEFINITY® Switch Generic 2 and System 85 R2V4
- DEFINITY Switch Generic 1
- DEFINITY Enterprise Communications Server Generic 3
- Other products that adhere to the ISDN-PRI signaling protocol

As an example of how the ISDN-PRI is used in private and public network configurations, see the following figures. As seen in these figures, the ISDN-PRI can be used to interface the MCU with a PBX.

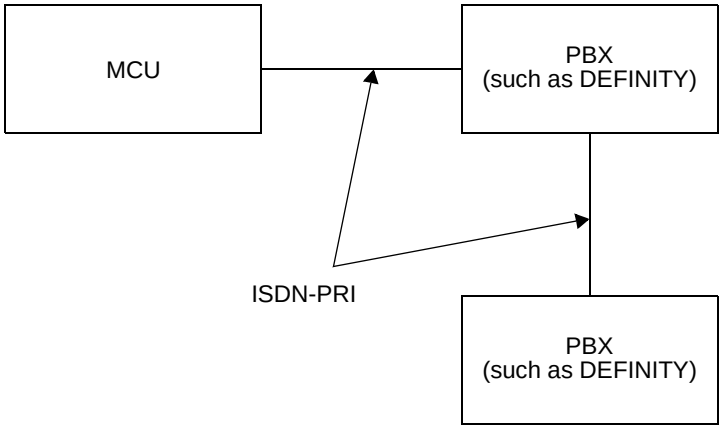

Figure B-9. ISDN-PRI Private Network Configuration

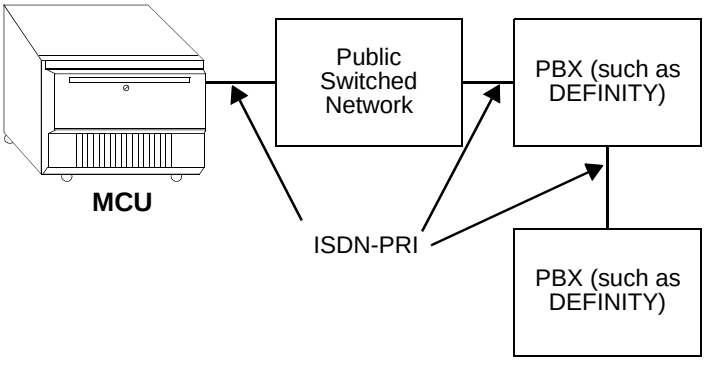

Figure B-10. ISDN-PRI Public Network Configuration

ISDN-PRI Services

The ISDN-PRI provides the following services:

- Private Network Services
- Multirate bandwidths of 112k, 128k, 168k, 192k, 224k, 256k, 280k, 320k, 336k, 384k, 448k, 512k, 672k, 768k, 1008k, 1152k, 1288k, 1344k, 1472k, 1536k, 1680k, and 1920k (both private and public networks) for video conferencing
- Call-By-Call Service Selection
- Access to Switched Network Services

Access to Switched Network Services

ISDN-PRI provides access to Switched Network Services such as Software Defined Data Network (SDDN), etc. An ISDN-PRI trunk group may be dedicated to a particular feature. Alternately, an ISDN-PRI Call-By-Call trunk group may provide access to several features. For a description of the ASN services accessible via ISDN-PRI (either via dedicated or Call-By-Call trunk groups), see the Call-By-Call Service Selection feature description elsewhere in this document.

ISDN-PRI Call Identification Display

Two types of identification numbers are provided with the ISDN-PRI. These identification numbers may be used in the various types of displays used with the ISDN-PRI. The two types of identification numbers are as follows.

- Billing Number (BN): The calling party's billing number that is provided to an inter-exchange network via Equal Access or Centralized Automatic Message Accounting (CAMA). This number is stored at either a local or network switch. If a customer is connected directly to the network, the BN is the customer's billing number stored in that network. If the CPN is not provided on an incoming ISDN-PRI call, the system uses the BN for the station identification number.

Private Network Services

In addition to providing access to switched public networks, the ISDN-PRI can provide private network services by connecting in an Electronic Tandem Network (ETN) configuration. This gives customers more efficient private networks that support new integrated voice and data services. Services are provided as follows:

- Multirate Bandwidth

  The primary function of the Multirate Bandwidth feature is to provide support for services that require large bandwidth, such as high-speed video conferencing at 112k, 128k, 168k, 192k, 224k, 256k, 280k, 320k, 336k, 384k, 448k, 512k, 672k, 768k, 1008k, 1152k, 1288k, 1344k, 1472k, 1536k, 1680k, or 1920k. These services have traditionally been handled by dedicated facilities. With the Multirate Bandwidth feature, dedicated facilities are no longer a requirement for these large bandwidth services.

Call-By-Call Service Selection

Call-By-Call Service Selection allows the same ISDN-PRI trunk group to carry calls to a variety of services or facilities (such as a SDDN and ACCUNET). This feature is described in detail under the Call-By-Call Service Selection feature description.

Access to Software Defined Data Network (SDDN)

With ISDN-PRI, the SDDN service may be accessed. SDDN provides virtual private line connectivity via the switched network (4ESS switch). The services provided by SDDN include voice, data, and video applications. SDDN services complement the Software Defined Network (SDN) voice services.

Access to Switched Digital International (SDI)

SDI provides 128k unrestricted connectivity to international locations via the switched network. It is also the backbone for the International ISDN network. SDI complements the ACCUNET digital service already available to US locations. This service can be accessed using the Call-By-Call Service Selection feature.

Considerations

ISDN-PRI is required for video conferences at data rates greater than 112k. CBC and NFAS data channel backup are available if required.

Interactions

None.

Administration

ISDN-PRI is administered on a per-system basis by the system administrator. The following items require administration.

- Communication - Interface Link
- Communication - Interface Processor
- DS1 Circuit Pack (for DS1 and E1)
- DS1 Synchronization Plan (for DS1 and E1)

Processor Interface Circuit Pack

- ISDN-PRI Trunk Group
- GRS Routing Patterns
- Signaling Group (See the Facility and Non-Facility Associated Signaling feature description elsewhere in this document).

Hardware and Software Requirements

- A TN767D DS1 circuit pack is required for assignment of a T1 signaling link and up to 24 ISDN-PRI Trunk Group members. The DS1 provides up to 24 ports.
- A TN2207 DS1 circuit pack is required for assignment of an E1 signaling link and up to 31 ISDN-PRI Trunk Group members. The DS1 provides up to 31 ports.
- A TN2182 Tone Clock circuit pack is required to provide synchronization for the DS1 circuit pack.
- A TN765 Processor Interface circuit pack is required for ISDN-PRI. The TN765 supports four ports (four D-channels).

One processor interface link is required per ISDN-PRI.

ISDN-BRI Trunk Interface

Description

ISDN-BRI Trunk Interface supports ISDN-BRI connections to the public network.

Considerations

ISDN-BRI Trunk Interface is intended primarily for customers with relatively small line sizes in countries and geographic locations where BRI lines are less expensive than PRI lines. The feature is also intended for customers in countries that do not have ISDN-PRI lines.

The availability of time slots on the MCU Time-Division Multiplexing (TDM) bus for conferences using the feature depends on the conferencing features being used (for example, conference size, bandwidth, CPP, SM, etc.) and whether breakage occurs. Breakage occurs whenever the channels of an endpoint must be broken up across multiple MMI/DS1 circuit pack pairs. Since ISDN-BRI Trunk Interface does not use MMI circuit packs, 100% breakage is assumed.

ISDN-BRI Trunk Interface supports the following transfer rates: 1B, 2B, and BONDing rates up to 768k. A maximum of five TN2185 ISDN-BRI circuit packs are supported; a maximum of three TN556 circuit packs are supported; and a maximum of three TN2198 circuit packs are supported.

The TN2185 circuit pack supports point-to-point calls at the 1B and 2B rates. The TN556 circuit pack can serve as a network T-interface for tie trunk applications. However, the circuit pack cannot replace the ISDN-BRI Trunk Interface functionality offered by the TN2185 circuit pack because it does not support trunk connections to the public network. The TN2198 circuit pack is a two-wire U-interface that allows longer tie-trunk connections using ISDN-BRI.

Multirates are not supported.

Interactions

The following features or capabilities interact with ISDN-BRI Trunk Interface with the specific changes or considerations as noted.

- Audio Add-On

  BRI ports on the TN2185 ISDN-BRI circuit pack can be used for including Audio Add-On endpoints into conferences. Therefore, whenever TN2185 ISDN-BRI circuit packs are being used, a DS1 TN767 circuit pack is not required to enable Audio Add-On.

- BONDing

  The TN2185 ISDN-BRI circuit pack supports BONDing at rates between 2B and 768k, inclusive.

- Dial-Out

  Whenever the TN2185 ISDN-BRI circuit pack is used to provide a BRI network interface, Dial-Out is supported.

- Direct Connect

  Endpoints can directly connect to the MCU via the TN556 ISDN-BRI circuit pack in configurations where the TN2185 ISDN-BRI circuit pack(s) are providing an ISDN-BRI network interface. However, the maximum number of TN2185 ISDN-BRI circuit packs is reduced in such a case due to the time slot usage by the directly connected endpoints.

- Universal Conference Control

  The TN2185 ISDN-BRI circuit pack supports UCC endpoints. Therefore, whenever TN2185 ISDN-BRI circuit packs are being used, a DS1 TN767 circuit pack is not required to enable UCC.

Administration

The following table shows the maximum possible number of conferences using ISDN-BRI Trunk Interface for varying bandwidths and conference sizes, and assuming a maximum of five TN2185 ISDN-BRI circuit packs.

Table B-16. Maximum Number of Conferences Using ISDN-BRI Trunk Interface

Conference Bandwidth	Conference Size (Number of Participants)		
	4	**12**	**20**
2B	10	3	2
6B	3	1	0
12B	1	0	0

To administer the feature, populate the forms discussed in the following sections according to the directives provided.

Customer Options Form

Populate the appropriate *Maximum Port Capacity* fields. Also, set the *BONDed Use Same DS1* field to **n**.

Optional Features Form

Ensure that the *ISDN-BRI Trunks* field in the Optional Features form is set to **y**. If the field is disabled, the BRI Trunk form is not accessible.

Feature Access Code Form

Populate the *ISDN Access Code* field with a four-digit code number. The number can contain any digits 0 through 9 along with the # and * symbols in any position (for example, ***256**).

System Capacities Form

Populate the I*SDN-BRI Ports* field with a value indicating the appropriate number of ISDN-BRI ports.

BRI Trunk Form

Populate the fields in the form according to the directives in "Administering ISDN-BRI Trunk Interface" on page 2-245.

Trunk Group Form

Populate the fields in the form according to the directives in "Administering ISDN-BRI Trunk Interface" on page 2-245.

Hardware and Software Requirements

The TN2185 circuit pack is required.

Lip Synchronization

Description

Lip Synchronization allows users to select the lip synchronization profile of their Continuous Presence *Plus* or Speed Match conferences. Such conferences can involve the TN2237C, vintage 16 or higher Video Processor Plus circuit pack.

Considerations

Lip Synchronization is standard for all MCU models. The feature may be used on a per-conference basis for Continuous Presence *Plus*, Speed Match, and H.263/H.261 Transcoding conferences.

Two Lip Sync modes are available:

- Preserved Lip Sync Mode

  With this mode enabled, the MCU provides additional audio delay for conferences in quad-screen mode and when the MCU is processing the video. This is done to preserve lip synchronization. The audio of all multimedia endpoints in the conference (except for the Presenter) is delayed in quad mode. The audio delay is implemented either automatically or via customer administration.

  NOTE:
  This includes “secondary” or “quad-view only” endpoints (that is, endpoints that can view the quad-image but cannot be a source into the CP conference).

  If the conference goes to full-screen mode, the MCU does not provide any deliberate audio delay because the MCU does no video processing in this mode.

  Preserved Lip Sync Mode can vary according to the type of video processor circuit pack used; it can also vary according to the transfer rate and frame rate of the Continuous Presence *Plus* conference.

- Audio Priority

  For this mode, the MCU provides no audio delay in either full-screen mode or quad-screen mode.

  This mode may be desirable for highly interactive conferences where minimizing the audio delay will prevent talkover annoyances. This mode may also be desirable in conferences where video cameras are not focused on the face and lips of any participants and, therefore, Lip Sync is not critical.

Interactions

Lip Sync Mode supports the following features or capabilities with specific changes or considerations as noted.

- Audio Add-On

  Lip Sync Mode has no impact on the voice path associated with Audio Add-On endpoints.

- Cascading

  For a cascaded MCU configuration with Continuous Presence *Plus* (CPP), the conference at the primary MCU must be optioned with Continuous Presence *Plus* in either Presentation mode or Voice-Activated Switching (VAS) mode. If Preserved Lip Sync is enabled on the primary MCU, and if the primary MCU is in VAS quad-mode, the audio delay is applied to all CPP-compliant video endpoints on the primary MCU, including the inter-MCU link. If Preserved Lip Sync is enabled on the primary MCU, and if the primary MCU is in Presentation mode, the audio delay is applied to the inter-MCU link since this link can be the video source for one of the quads in the quad-image returning to the Presenter. However, audio-delay is disabled for the Presenter.

  The secondary MCU must be administered as a non-CPP Presentation mode conference to force the inter-MCU link to be the full-screen broadcaster. Since this MCU is in non-CPP mode, Preserved Lip Sync cannot be administered on this MCU.

- Presentation Mode

  If Preserved Lip Sync is enabled on a CPP conference with Presentation mode, and if the Presenter is viewing a quad-screen, the audio from the non-Presenter sites is delayed. However, if the Presenter requests to view full-screen, the audio delay is removed from all the endpoints. The audio from the Presenter in a CPP conference is never delayed in any case.

Administration

Lip Sync mode is enabled via several forms. The feature can also be enabled during an active conference. For details, refer to "Administering Lip Synchronization" on page 2-235.

Hardware and Software Requirements

The feature requires TN787K, vintage 6 or higher MMI circuit packs. The TN787K firmware is backward compatible with all previous MCU software releases.

Low-/High-Speed Interworking

Low-/High-Speed Interworking allows lower-speed (56k and 64k) endpoints to join a higher-speed conference as Audio Add-On endpoints.

Description

A lower-speed endpoint can connect with a single channel to a single channel higher-speed conference as an audio-only endpoint. There is no limit to the number of low-speed endpoints that can join the conference in this way.

Considerations

Low-/High-Speed Interworking is allowed only for 56k and 64k endpoints. The low-speed endpoint connects with only one channel and only to high-speed single channel conferences. During a conference, this feature can be administered from off to on but not on to off.

Interactions

The Low-/High-Speed Interworking feature interacts with the following features.

- CDR

  CDRs for the call indicate the number of channels used by each call.

- Cascading

  Low-speed calls coming on a cascade link are blocked.

- BONDing

  The feature is not allowed in a BONDed conference.

- Dial-In/Dial-Out

  Only dial-in endpoints can use this feature.

- Passwords

  If passwords are administered for the conference/endpoint, the MCU requests a password from the low-speed endpoint even if that endpoint is joining as an audio-only endpoint.

- T.120 Data Conferencing

  Since low-speed endpoints join as audio-only, they cannot access the T.120 Data Conferencing features.

- Chair Control

  The MCU denies chair control to any audio-only endpoint.

- CRCS

  If interworking is enabled for a conference, then CRCS relaxes conditions for selecting endpoints but still does not allow BONDed ports since interworking is blocked in conferences with BONDed endpoints.

- Dynamic Resizing

  If dynamic resizing is enabled for a conference, then interworking can be administered from off to on during the conference. Interworking can never be administered from on to off during a conference.

Administration

Low-/High-Speed Interworking is enabled by administering it on the conference form for each conference.

Hardware and Software Requirements

Low-/High-Speed Interworking is available only on release 3.0 or later of the MCU.

Multirate Bandwidth

Description

The ISDN-PRI divides a T1 (E1 for international switches) trunk into 23 or 24 (30 for E1) information channels and one signaling channel for standard narrowband communication. Certain applications, such as video conferencing, require greater bandwidth, and it becomes necessary to aggregate several of those narrowband channels into one “wideband” channel to accommodate the extra bandwidth requirement. The Multirate Bandwidth feature provides an MCU with the ability to receive 112k, 128k, 168k, 192k, 224k, 256k, 280k, 320k, 336k, 384k, 448k, 512k, 672k, 768k, 1008k, 1152k, 1288k, 1344k, 1472k, 1536k, 1680k, or 1920k calls.

Technical Description

The Multirate Bandwidth feature provides end-to-end connectivity between endpoints according to the ITU-T and ANSI standards for 112k, 128k, 168k, 192k, 224k, 256k, 280k, 320k, 336k, 384k, 448k, 512k, 672k, 768k, 1008k, 1152k, 1288k, 1344k, 1472k, 1536k, 1680k, or 1920k calls.

NOTE:
Typically, a T1 facility provides for 23 B-channels and 1 signaling while an E1 facility provides 30 B-channels and 1 D-channel. If however, Non-Facility Associated Signaling (NFAS) is used, a group of T1/E1 facilities can be configured to share a single signaling channel allowing all but one of the T1/E1 facilities to be configured with an extra B-channel. The MCU supports up to 25 T1/E1 facilities.

An MCU can be configured for Multirate Bandwidth to support multipoint conferences at 112k, 128k, 168k, 192k, 224k, 256k, 280k, 320k, 336k, 384k, 448k, 512k, 672k, 768k, 1008k, 1152k, 1288k, 1344k, 1472k, 1536k, 1680k, or 1920k. Each B-channel can provide 64k transmission rates. There are two service types for encoding the aggregation of channels, both of which are supported by the MCU.

Nx64 allows the aggregation of any channel groupings, while 384k (H0) and 1536K (H11) have fixed channel assignments.

Channel Allocation

Multirate bandwidth channel allocation is performed using one of three allocation algorithms: fixed, flexible, or floating.

- Fixed allocation provides contiguous channel aggregation and the starting channel is constrained to a predetermined starting point. (This is used only for 384k (H0), 1536K (H11), and H12 calls.)
- Flexible allocation allows an Multirate Bandwidth call to occupy non-contiguous positions within a single T1 or E1 facility.
- Floating allocation enforces contiguous channel aggregation but the position of the first channel is not constrained like it is in fixed allocation.

Considerations

Blocking is the most pertinent trunking consideration when using Multirate Bandwidth.

Blocking

Blocking occurs when the number of B-channels required to make a call are not available. Narrowband (128k) calls require two channels and are usually not blocked. However, blocking occurs for Multirate Bandwidth calls when bandwidth is not available in the appropriate format (that is, fixed, floating, or flexible, see earlier description).

Interactions

The following features interact with the Multirate Bandwidth feature:

- ACA

  Treats Multirate Bandwidth calls as single trunk calls so that only a single ACA referral call is made if an ACA referral call is required. The referral call is on the lowest B-channel associated with the Multirate Bandwidth call.

- CDR

  Multirate Bandwidth calls trigger CDR records containing the Multirate Bandwidth Bearer Capability Class (BCC) and the call bandwidth.

- ISDN-PRI

  Standard ISDN-PRI interfaces can be administered for Multirate Bandwidth switching.

Administration

Multirate bandwidths are administered on a per-system basis via the Optional Features form. Also, the Trunk Group (ISDN-PRI) form must be administered for Multirate Bandwidth service.

Trunk Groups

Trunk group administration is required to specify the Multirate Bandwidth services that are to be supported by each trunk group. The parameters 384k (H0), 1536K (H11), and NXDS0 must be administered to identify the algorithm that the trunk group uses to provide Multirate Bandwidth service.

Hardware and Software Requirements

The TN767D or TN2207 circuit pack must be used for any facility that carries Wideband traffic. The only other additional hardware or software required for Multirate Bandwidth is that the PRI endpoints must adhere to the ISDN-PRI wideband interface requirements and to PRI line-side requirements.

The TN2207 is required for E1 facilities.

MultiRate Video (MRV)

Description

The Lucent Technologies MCU supports INTEL® ProShare™ endpoints using MRV proprietary video protocols in a standard format. The MCU supports MRV protocols via the INTEL extension to the ITU-T H.230/H.243 conferencing standards, and it provides the proprietary video protocols within the H.221 frame structure for multipoint calls.

Considerations

MRV video is available as a standard feature on all models.

The MCU can support simultaneously H.261 video as well as the following standard proprietary videos on separate simultaneous conferences: MRV, H.CTX, H.CTX+, and SG4. Any endpoint that does not support the Selected Communications Mode (SCM) is an audio-only endpoint.

MRV video makes use of conventional H.320 ports, and it is supported by the maximum number of ports allowed within the Lucent Technologies MCU. All port types except for Audio Add-On and UCC ports may be used as MRV ports within an MRV conference. INTEL ProShare endpoints support MRV video rates up to 128k.

Interactions

MRV video does not support Continuous Presence *Plus*. The following features are supported with specific changes or considerations as noted:

- Basic/Enhanced Flag

  In an MRV conference, the MCU must transmit Multiple Byte Extensions (MBEs) to all endpoints (including “basic” endpoints) to “turn on” video capabilites. MRV endpoints can therefore be designated as either “basic” or “enhanced” endpoints and receive video within an MRV conference.

  NOTE:
  With “basic” enabled, the user cannot access certain features or functions, such as Terminal Names and Chair Control.

- Cascading

  An MRV conference can be cascaded with another similarly configured conference provided each MCU involved is a Lucent Technologies MCU.

- Dynamic Conference Resizing

  The *Video Mode* field of the Conference Record, which may contain a value for MRV video, cannot be changed during an active or in-use conference.

- Rate Adaptation

  This feature is supported at a 56/64k rate for two channels.

- T.120 Data Conferencing

  Either a fixed-rate or variable-rate MLP data stream can be supported in an MRV conference.

- Universal Conference Control (UCC)

  UCC commands for Continuous Presence *Plus* are not supported.

Administration

The Conference Record form is used to administer MRV video on a per-conference basis. For instructions, see the “Administering MRV Video” section in Chapter 2.

Hardware and Software Requirements

MRV video requires the TN787K, vintage 6 or higher circuit pack.

Networking and Call Handling

Description

The MCU provides a flexible and robust networking interface. This networking interface consists of the network trunks (DS1 and PRI) that provide the physical connection to the network (private or public) and the call handling software that processes the incoming and dial-out calls. Additionally, the network interface supports flexible call handling such that each video port supports variable bandwidths (transmission rates) on a per-conference basis. Therefore, no hardware nor software modifications are required to the system beyond the setting of the bandwidth parameter on the Conference Record form. As an example, video port 51000 can be administered in a 128k conference that lasts from 10:00 to 11:00. At 12:00, a new conference administered for 384k (H0) can reuse video port 51000 without any hardware or software modifications.

The MCU supports Robbed-bit DS-1 and ISDN-PRI trunks for connecting to the public switched network or a PBX (see the ISDN-PRI section). All video calls gain access to the MCU via these trunks. For each incoming call, the MCU expects to receive an MCU number (that is, a network number assigned to the MCU). The received MCU number can be used by the reservations agent (or system administrator) to help identify which endpoint originated the call and is always used by the MCU to identify the video port to which the call is to be connected. As with other equipment connecting to a network, it is expected that the network dial plan and call routing is already provisioned and in place prior to attempting video calls to the MCU. In the same way, outgoing (dial-out) requires the MCU dial plans, trunk groups, and routing be provisioned and in place prior to attempting a video call to the network.

Considerations

The objective of the Network/Call Handling Subsystem is straightforward in that calls are received from or sent to the network and connected to the proper video port within the MCU. Once the MCU is connected to the network, and the network administrator has provisioned the network dial plan, calls to the MCU can be made and the dial-out feature can be used to make calls. For each incoming video call, the MCU receives the incoming digits (that is, MCU number). (See the “Dial-Out” section in this chapter for details on outgoing call (dial-out) MCU numbers) This MCU number is used internally to route the call to the proper video port. Each video port is identified via its extension (that is, video port extension). Once internal processing of the MCU number is completed, the call is connected to a video port.

Interactions

To implement the function outlined above, the networking or call handling subsystem uses the following MCU features:

- DS-1 Trunk Service

  Enables administration of one to several DS-1 trunks groups.

- ISDN-PRI

  The ISDN-PRI feature enables the administration of the ISDN-PRI parameters for trunks administered for ISDN-PRI signaling.

- Dial Plan

  Enables the administration of the video port extensions.

- AAR

  AAR provides the call handling capability. MCU numbers for incoming calls are analyzed and, in turn, converted to the appropriate Video Port extension. The Video Port extension is then used to connect the call to the appropriate video port.

To handle incoming calls properly, the following steps must be completed:

1. Each trunk group must be administered to prepend an 8 on each MCU number received from the network

   NOTE:
   The network administrator is responsible for properly administering the network to deliver the entire MCU number (that is, network number).

   The network should be provisioned to deliver to the MCU the MCU numbers such that they correspond to the numbers distributed by the Reservations Agent and thus the number dialed by the end user. To join a 128k or 384k conference, a MCU number must have an administered bandwidth of 128k/384k. To join a 112k conference, a MCU number may have an administered bandwidth of 112k or 128k/384k.

2. The Dial Plan feature must be administered. Once completed, the Video Port extensions exist (4- or 5-digit extensions) and the AAR feature access code (8) is provisioned.

3. The AAR Digit Conversion Table must be administered such that each MCU number exists in the ‘‘Matching Pattern’’ field.

   NOTE:
   The full 7-, 10-, 4-digit (or whatever dial plan is implemented in the network and in turn passed to the MCU) number must be entered in this field. Directly across from each entered MCU number in the ‘‘Replacement String’’ field is a Video Port extension. The Video Port extension must correspond to the Dial Plan administered via the Dial Plan feature.

See “Dial-Out” section of this chapter for details of the requirements that need to be met to properly handle outgoing (dial-out) calls.

Administration

See individual feature.

Hardware and Software Requirements

See individual feature.

Notification Package

This feature includes conference tones and text messages.

Description

Conference tones and text messages are provided during MCU conferences to alert users of the following: endpoints joining the conference, endpoints dropping from the conference, and impending conference end (warning tones and messages).

NOTE:
Refer to the “On-Screen Display” section in this appendix.

Considerations

NOTE:
Refer to the “On-Screen Display” section in this appendix.

Interactions

NOTE:
Refer to the “On-Screen Display” section in this appendix.

Administration

NOTE:
Refer to the “On-Screen Display” section in this appendix.

Hardware and Software Requirements

NOTE:
Refer to the “On-Screen Display” section in this appendix.

On-Screen Display

Description

On-Screen Display (OSD) provides various Notification Package enhancements and Automatic Number Identification (ANI) codes within the MCU video image. The feature enables images to be intertwined into an endpoint's decoded video stream and then encoded back as return video. OSD is provided only in non-speed-matched homogeneous CPP-only conferences. The following OSD applications are supported:

- End Notification — Text string display providing the traditional 10 minute end-of-conference warning notification (for example: "10 Minutes Remaining in Conference") as well as a subsequent two minute remaining warning (for example, "Less than Two Minutes Remaining"). The first string is displayed for approximately 10 seconds, and the latter string is displayed continuously.

  NOTE:
  The traditional audible tone for the 10 minute warning/end notification can be provided along with the text message. However, no two minute remaining tone is provided.

- Entry Notification — Either a text message display indicating the site name of the endpoint that has just joined the conference (for example: "Welcome NY") or the Automatic Number Information (ANI) code (for example, "7325551234 has joined") for such an endpoint. The message or ANI code is provided for both video and Audio Add-On endpoints. The first 15 characters of the site name as administered in CRCS or on the Conference Record form are displayed.

  NOTE:
  The traditional audible tone for entry notification can be provided along with the text message.

- Exit Notification — Either a text message display indicating the site name of the endpoint that has just left the conference (for example: "Goodbye NY") or the Automatic Number Information (ANI) code (for example, "7325551234 has exited") for such an endpoint. The message or ANI code is provided for both video and Audio Add-On endpoints. The first 15 characters of the site name as administered in CRCS or on the Conference Record form are displayed. Any additional characters in the site name are truncated.

  NOTE:
  The traditional audible tone for exit notification can be provided along with the text message.

Considerations

There is no individual right-to-use (RTU) fee associated with the OSD feature itself. Rather, the OSD capability is included within right-to-use fee for Continuous Presence *Plus* video processing. OSD functions at up to 30 fps at all CPP transfer rates up to 768k.

OSD is supported only in homogeneous CPP-only conferences. This implies that the feature can be administered only for a non-speed-matched CPP conference. The CPP conference must also be an H.261 homogeneous conference or an H.263 homogeneous conference. In addition, the OSD text messages can be displayed only whenever the CPP conference is in quad-screen or panoramic mode; the messages cannot be seen whenever the conference is in full-screen mode. However, if the conference switches from full-screen mode to either quad-screen or panoramic mode, the text messages are displayed.

The text strings for indicating conference end notification, site entry identification, and site exit identification are displayed across the top of the lower two quadrants (that is, quads 3 and 4). The text messages are displayed for an appropriate length of time to allow for the proper acknowledgment from conference participants. Thereafter, the messages are removed from the quad image to minimize any disruption to the natural meeting experience.

OSD text messages comprise white text encapsulated in a black box to ensure optimal visibility independent of the color composition of the live video stream from the sites in quads 3 and 4.

The intention of the OSD text messages is also fulfilled via the existing audible tone notification. The appropriate tone ensures that the conferees receive the appropriate notification in any case and, particularly, if the CPP conference switches to a full-screen conference. The tone also prompts the conferees to acknowledge the OSD message (if provided) before it is removed from the screen.

For OSD text messages to be displayed, the conference must be in CIF mode.

If numerous endpoints join a conference within a relatively short period of time, the entry notifications for the latest endpoints are queued and displayed after the OSD entry notifications for all endpoints that joined the conference previously have been displayed for the full length of time.

Interactions

OSD does not support the proprietary algorithms (including SG4, H.CTX, and MRV). The following features or capabilities interact with OSD with the changes or considerations noted.

- CPP Presentation Mode

  In CPP Presentation mode, only the presenter (who is viewing a quad-image) sees the text overlays. All other conferees with video capabilities see the presenter in full-screen and receive only the audible tone warnings.

- H.261 and H.263 Video Modes

  The OSD text overlay options are supported in H.263 homogeneous CPP-only conferences as well as in H.261 homogeneous CPP-only conferences.

- OneNumber Access

  For conferences with OneNumber Access, whenever the OSD text overlay associated with the entry/exit notification is tied directly to the data in the site names field for each endpoint, there is no clear way to associate a particular site with a specific conference endpoint number since many sites may share the same MCU extension. A variable number of endpoint numbers are available to each endpoint entering the conference. The endpoint number assigned in each case depends on the order in which the associated endpoint enters the conference. Therefore, the “text” and “text/tone” options for entry/exit alert are not allowed whenever OneNumber Access is being used within the conference.

- QCIF Mode

  With this mode, the text resolution is insufficient. Therefore, OSD text messages are suppressed.

- Terminal Names/Site Names

  The site name information provided via the OSD text overlay is the information manually administered within the MCU/CRCS via the ResCenter database or via the MSM Conference Record form.

Administration

This feature is administered via the System-Parameters Customer-Options form and the Conference Record form. Refer to the “Administering On-Screen Display” section in Chapter 2 for details.

NOTE:
It is highly recommended that text or ANI notifications along with tone notifications be provided to conference participants rather than just text or ANI notifications. This recommendation is made with regard to the various possible conference profiles within conferences that may not always allow text or ANI notification. For example, some conferences may include Audio Add-On or secondary audio-only endpoints; obviously, such endpoints cannot receive OSD notification if the text- or ANI-only option is enabled. Another benefit of using the text/tone or ANI/tone option is within CPP conferences that are in full-screen mode because such conferences are unable to display OSD text or ANI. Finally, even within conferences where text or ANI messages can be displayed, sounding an audio tone in conjunction with displaying a text message or ANI draws the attention of the conference participants to the monitor for their respective endpoints and therefore ensures that the participants read the text notification or ANI before it disappears.

Hardware and Software Requirements

The system must have CPP RTUs and TN2237C, vintage 16 or higher circuit packs to support OSD capabilities.

OneNumber Access

Description

OneNumber Access is a customer option that allows multiple endpoints in a conference to access the MCU from the same external extension. A OneNumber Access extension is a duplicated MCU-extension within a single conference. Endpoints sharing a OneNumber Access extension are not preassigned to a specific conference endpoint number; instead, these endpoints share a pool of endpoint numbers and receive a conference endpoint ID based on when they join the conference.

Considerations

Each MCU conference can have multiple OneNumber Access extensions. Conferences can be configured to allow a mixture of endpoints that use the OneNumber Access extension as well as the conventional unique number per endpoint (Meet-Me number) access. Also, OneNumber Access can be used when the Dial-Out feature is enabled.

Audio Add-On ports can use OneNumber Access; however, as in previous MCU releases, such ports use a different set of MCU-extensions from H.320 endpoints.

A OneNumber Access extension *cannot* be used by the following endpoint types:

- BONDed (BOND)
- Cascaded (CAS)
- BONDed Cascaded (BCAS)
- Controlling extension in Presentation mode (including one for a fixed extension or Empty Quadrant Extension)
- Dedicated Access (DA)
- Universal Conference Control (UCC)

Moreover, OneNumber Access supports only those transfer rates that allow H.320 endpoints. An example of an excluded rate is 336k, which can be used only by BONDed and DA endpoints. The feature supports Audio Add-On endpoints at any transfer rate.

All endpoints using the same OneNumber Access extension must have the same setting for the *Password*, *App Comp* (Application Compliant), and *Basic/Enh* (Basic/Enhanced) fields in the Conference Record form.

OneNumber Access is standard on all MCU models. There is no limit to the number of conferences for a given MCU that can use this feature. Also, there is no limit to the number of OneNumber Access extensions that can be used within a single MCU conference.

OneNumber Access is intended for 2x56k and 2x64k conferences in an Integrated Services Digitial Network (ISDN) environment where calling party information is available from all sites joining the conference via a common MCU-extension. Environments where calling party information is not available (for example, robbed-bit facilities) are not supported.

OneNumber Access for H.320 endpoints can be used in single-channel conferences where BONDing and DA are not used to access the conference. This implies that single-channel OneNumber Access is limited to Wideband non-BONDED (for example, H0 or multirate) calls.

OneNumber Access for Audio Add-On or single-channel endpoints does not require the availability of Calling Party Number (CPN) information.

Interactions

OneNumber Access supports the following features or capabilities with specific changes or considerations as noted.

- Dial-Out

  Dial-out endpoints are blocked from using an MCU-extension that occurs more than once in a conference; therefore, in such a case, OneNumber Access cannot be used by this endpoint type.

- Dynamic Conference Resizing

  OneNumber Access can be enabled during an active conference. This is done by changing the value of the *OneNumber Access* field in the Conference Record form from **n** to **y**. Changing the field value from **y** to **n** is not allowed.

  Additional occurrences of an MCU-extension are allowed if OneNumber Access is enabled. However, any endpoint using a repeated extension cannot be changed to a dial-out endpoint. Also, such an endpoint cannot be removed from a conference that is active or in use.

- Password Protection

  Conference reservation allows the administration of conference-level passwords for conferences using OneNumber Access extensions. All endpoints using the same extension must have a common setting for the *Password* field (either **y** or **n**).

- Universal Conference Control

  UCC is allowed in conferences using OneNumber Access. However, a UCC endpoint cannot use an MCU-extension that has been previously used. Therefore, conference reservation blocks the entry of a duplicated MCU-extension for the UCC endpoint type.

Administration

OneNumber Access is enabled via the Conference Record form. Refer to the “Administering OneNumber Access” section later in Chapter 2.

Hardware and Software Requirements

None.

Passwords

The passwords feature provides additional conference security by requiring a password entry for access to a conference on a per-conference basis.

Description

With per-conference passwords set at the conference reservation, the MCU prompts each conferee as they attempt to join the conference.

Audio-only conferees are given a single chance to provide the correct password and, if a password is incorrect, the endpoint is dropped from the conference. All other conferees are given three chances to enter the correct password before being dropped. In either of these cases, the Status Conference form shows “invalid password’’ as the drop reason for that endpoint.

Considerations

Audio Add-On Dial-Out calls are not prompted for a password.

Interactions

The following features are supported with the considerations noted:

- Dial-Out

  When the MCU dials out for an endpoint that is set for auto-answer, the MCU prompts for the password after the endpoint answers. After a timeout with no endpoint entry (administered in the range 30-300 seconds) the MCU prompts a second time. If the endpoint still does not respond within the timeout, the MCU prompts a third time. If there is still no response from the endpoint, that endpoint is dropped from the conference. In this case, the reason for drop is set to ‘‘UIN/password timeout.’’

Administration

The timeout value for auto-answer endpoints must be set to a value in the range 30-300 seconds. The timeout value is the same for all conferences. The password flags for per-conference for each conference must be selected at reservation time. The password flag must be selected at each time a conference is reserved.

Hardware and Software Requirements

None.

Rate Adaptation

Rate Adaptation provides a way for interaction of 2-channel (56k or 64k) endpoints in a conference.

Description

With Rate Adaptation, 56k endpoints can be included into a 64k conference. If a 56k endpoint joins a 64k conference, the 64k endpoints are dynamically altered to an effective rate of 56k. The 56k rate remains in effect until the conference ends.

Considerations

None.

Interactions

The Rate Adaptation feature interacts with the following:

- Dial-Out

  Dial-Out is capable of initiating calls using Rate Adaptation.

- Dynamic Resizing

  Rate Adaptation can be enabled (turned on) during a conference if Dynamic Resizing is enabled. It can never be changed from on to off during the conference.

Administration

The conference record must be administered to allow or disallow 56k or 64k rate adaptation in the conference.

Hardware and Software Requirements

None.

Recent Change History

Description

The Recent Change History feature allows the system administrator or reservations agent to view or print out a history report of the most recent administration and maintenance changes. This report may be used for diagnostic or information purposes.

The system maintains a log in a software buffer of the most recent administration and maintenance commands, up to a maximum of 250 data commands. The log is called the transaction log. The commands must be ‘‘data affecting’’ and successfully entered to be saved in the transaction log. The ‘‘data affecting’’ commands are called data commands.

The transaction log can be displayed or printed as a report by entering the **list history** or **list history print** command at the Management Terminal, or a remote terminal by the following users:

- Local Customer Administrator
- Local System Technician
- Remote Customer Administrator
- Remote System Technician

Considerations

A maximum of 250 data commands are stored in the transaction log.

The Permission Administration form shows the command permission categories that a user can access; for example, ‘‘Administer Features.’’ There are no permission restrictions for access to the Recent Change History reports. The local and remote customer and system technician administrators have unrestricted access.

Interactions

The following features interact with the Recent Change History feature.

- Other Users

  When a user requests a Recent Change History report, it takes a little time to read all the pages of the report. If during this time other users are entering data commands and altering the transaction log, the oldest entries in the transaction log may have been overwritten by the data commands entered by these other users.

- Set Time Command

  The use of the maintenance command **Set Time** to change the system clock could make the Recent Change History report look as if it were not in true last-in, first-out order.

Administration

None.

Hardware and Software Requirements

No additional hardware or software is required.

Report Scheduler and System Printer

Description

This feature allows the system administrator to schedule selected administration commands to be printed by an asynchronous printer. Reports are scheduled at 15-minute intervals for any combination of days of the week. Most **list**, **display**, or **test** commands may be scheduled.

Reports may be scheduled, changed, listed, and removed via the system's Management Terminal.

Considerations

With the Report Scheduler and System Printer, the system administrator can schedule most **list**, **test**, and **display** administration commands to be printed at various times on an asynchronous printer. By scheduling these reports to print automatically at the desired times, the system administrator saves valuable time that can be used to perform other administrative duties.

The system administrator can schedule a maximum of 50 individual reports. The system has a single asynchronous printer connection dedicated for use by the report scheduler. The system can be configured with other printers in the system such as those connected to the MCU-MT and the CDR printer. These are not used by the Report Scheduler feature.

Reports scheduled for the same time and day are printed according to their order in the Report Scheduler queue. The first report in the queue will be printed first.

To present the least possible impact on system performance, it is recommended that reports be scheduled at off-peak hours and staggered so they are not all scheduled to be printed at the same time.

Reports that are added to the scheduler queue and that are scheduled to be printed during the current time interval will not be printed until the next scheduled time.

If a system error is encountered while trying to print a scheduled report, the error will be printed on the report, just as it would be displayed for the same command on the Management Terminal screen.

Interactions

There is only one processor board EIA port available for asynchronous output. The port cannot be administered for both CDR and the Report Scheduler System Printer on the System-Parameter Feature form.

Administration

The system administrator may schedule, list, change, and remove the desired reports as previously described in this description. Before these procedures can be done, however, the system administrator must supply printer information on page 4 of the System-Parameters Features form by entering the following information:

- **Printer Extension:** ‘‘EIA’’ for the EIA port or a valid data module extension if the EIA port is not to be used.

  The system administrator must specify the printer link by selecting either the EIA port, if available, or a data-module extension. If the data-module extension is chosen, the system administrator must have administered the extension previously using the **add data-module** command.

- **EIA Device Bit Rate:** The speed of the printer (1200, 2400, 4800 or 9600 baud). The default is 1200.

- **Lines per Page:** The number of printed lines per page (24 to 132). The default is 60.

Hardware and Software Requirements

The asynchronous printer can be connected to the switch using either of the following methods:

- The printer can be connected directly to the EIA port on the switch’s processor board. In this case, the appropriate cable is required.

- The printer can be connected to the MCU with a 7400B data module and a port on a TN754B Digital Line circuit pack.

There is a single EIA port in the MCU. There may be contention between the CDR and the Report Scheduler feature for use of this port. If the Report Scheduler feature is using the EIA port and you would like to enable the CDR feature, it is recommended that you disconnect the system printer from the EIA port and use a data module for its connection, freeing the port for use by CDR.

A 475 or 572, or compatible printer that uses a serial interface may be used as the System Printer. A PC may be connected to the system printer port for collection of data; however, a serial interface on the PC must be provided for the connection.

RS366 Dialing

Description

RS366 Dialing is a companion feature to the V.35/RS449/EIA-530 data interface capability. RS366 Dialing enables the MCU to process dialing strings received from endpoints and devices connected to the MCU via a V.35/RS449/EIA-530 data interface. Specifically, H.320, H.CTX, and SG4 endpoints connected in this manner can do the following:

- Dial in to multipoint audio, video and/or data conferences
- Make spontaneous point-to-point calls at any rate except BONDed rates

The feature also provides a basis for including H.323 Local Area Network (LAN) endpoints into multimedia conferences with H.320 endpoints. To this purpose, a gateway is required.

Considerations

RS366 Dialing enables the following.

- H.320/H.CTX/SG4 codecs directly connected to the MCU via a V.35/RS449/EIA-530 interface to do the following:
  - — Dial in to multimedia, multipoint conferences. Accordingly, endpoints can join conferences after the conference start time, drop from conferences early, or participate in another conference or call.
  - — Place and receive spontaneous non-BONDed point-to-point calls with other endpoints that are directly connected to the MCU or reside anywhere else in the network via ISDN or robbed-bit trunks.
- H.323 endpoints connected behind H.323 gateways to participate in multimedia, multipoint conferences where the connection between the gateway and the MCU is V.35/RS449/EIA-530 (for which the rates are greater than 2B). The endpoints are provided dial-in capabilities.

The feature does not support BONDed spontaneous point-to-point calls from directly connected endpoints. However, such endpoints can participate in a conference or call with another endpoint that has joined at a BONDed rate.

RS366 Dialing from the MCU to devices such as inverse multiplexors (IMUXs) is not allowed. Therefore, endpoints behind IMUXs cannot participate in multimedia conferences as dial-out participants unless the connections from the IMUX to the endpoint are “fixed.” With the V.35/RS449/EIA-530 data interface capability, endpoints included within a switched network can participate in conferences only through manual dialing either from the endpoint into an IMUX or from an IMUX.

All transfer rates supported for conferencing by the Data Interface (DI) circuit pack within the MCU are supported for RS366 Dialing. These rates include the

following: 56k, 64k, 112k, 128k, 168k, 192k, 224k, 256k, 280k, 320k, 336k, 384k, 448k, 512k, 672k, 768k, 1008k, 1152k, 1344k, 1472k, 1536k, 1680k, and 1920k. H.320/H.CTX/SG4 and H.323 endpoints using RS366 Dialing can join conferences at any of these rates. Like the DI circuit pack, the feature does not support the following transfer rates: 1064k, 1176k, 1288k, 1400k, 1512k, 1624k, and 1736k.

H.320/H.CTX/SG4 and H.323 endpoints can make spontaneous point-to-point calls at any non-BONDed transfer rate that is a multiple between 1 and 30 of 56/64k, except for the rates not supported by the DI circuit pack. The lowest rate is 56/64k, and the highest rate is 1680/1920k.

The Lucent Technologies MCU supports at least 32 V.35/RS449/EIA530 ports at BONDed or multirate transfer rates up to and including 768k. The maximum number of 2B ports is at least 16 ports (alternatively, a maximum or 32 1B ports is allowed), with a maximum of at least 20 ports for the 1536k rate and 16 ports for the 1920k rate. RS366 Dialing is optional for this maximum number of ports.

If an endpoint supporting RS366 Dialing drops early from conference, the associated port is made available in another conference (that is, for another call). Also, if such an endpoint enters a conference by dialing in, and if the conference has been scheduled via the MCU-MT, the endpoint may drop from the conference prior to the scheduled end time of the conference, and it may enter another conference or make another call prior to the end time of the conference it has left. Whenever an RS366 dial string is received by the MCU on a DI circuit pack, the LED indicator for the circuit pack is yellow.

Interactions

The following features are supported with the changes or considerations noted.

- BONDing

  The feature does not support BONDed spontaneous point-to-point calls from directly connected endpoints. However, such endpoints can participate in a conference or call with another endpoint that has joined at a BONDed rate.

  NOTE:
  One solution for the lack of support for BONDed spontaneous point-to-point calls from directly connnected endpoints is to create a two-party dial-in conference at the desired rate.

Administraton

The feature is administered via the System-Parameters Customer-Options form, the Conference Record form, and the Access Endpoint form. Refer to the “Administering RS366 Dialing” section in Chapter 2 for details.

Hardware and Software Requirements

RS366 Dialing from the MCU to devices such as inverse multiplexors (IMUXs) is not allowed. Therefore, endpoints behind IMUXs cannot participate in multimedia conferences as dial-out participants unless the connections from the IMUX to the endpoint are “fixed.” With the V.35/RS449/EIA-530 data interface capability, endpoints included within a switched network can participate in conferences only through manual dialing either from the endpoint into an IMUX or from an IMUX.

Security Violation Notification

Description

This feature notifies the maintenance alarm terminal of an attempt to breach system administration access via an invalid login or Remote Access via an invalid barrier code. Information specific to access violations is available through the Monitor Security Violations status report.

Considerations

A maximum of one feature button (per system) may be assigned for each component of the security violation notification (SVN) feature [one for login security violations (LSVN) and one for Remote Access security violations (RSVN)].

An activated SVN feature button cannot be deleted by administration. The system will deny an attempt to delete an active SVN feature button.

Activation of an administered SVN feature button from the maintenance alarm terminal will be prohibited if the appropriate SVN feature component (either login or Remote Access) is not enabled in the Feature-Related System Parameters screen.

A maximum of one Referral Destination may be administered per MCU for each component of the SVN feature.

The Call Coverage, Call Forwarding, and Call Pickup (for example, within pickup group) features are not supported for SVN.

Unlike the ACA referral call, the SVN referral call is a priority call.

SVN does not currently support Speech Synthesis Circuit pack functionality.

Interactions

- ACA

  The originating extensions for login and remote access security violations cannot be the same as the originating extensions for ACA long and short holding time originating extensions. The destination extensions for both features can be the same.

Administration

The SVN feature is activated when either or both of the SVN Login Violation Notification Enabled or SVN Remote Access Violation Notification Enabled fields on the Feature-Related System Parameters form are enabled.

When the SVN Login Violation Notification Enabled field is enabled, the following additional fields appear on the Feature-Related System Parameters screen:

- Originating Extension

  This field requires the entry of an unassigned extension that is local to the switch and conforms to the dial plan for the purpose of originating and identifying SVN referral calls for Management Applications login violations.

  The originating extension initiates the referral call in the event of a login security violation. It also triggers the display of the appropriate alerting message on the referral destination's display module.

  NOTE:
  The SVN feature button does not have to reside on the referral destination station. It can be administered on the maintenance alarm terminal, but must be active if referral calls are to be placed.

- Referral Destination

  This field requires the entry of a previously administered maintenance alarm terminal. The destination may enable or disable the referral call through a feature button. The SVN feature buttons do not have to reside on the station(s) that is/are the referral destination(s). A referral call will not be made to the administered destination if its corresponding feature button is deactivated.

  The Referral Destination is the extension assigned to the maintenance alarm terminal that will receive the referral call in the event of a login security violation.

- Login Threshold

  This field requires the entry of the minimum number of invalid login attempts that will be permitted before a referral call is made. The value of this field, in conjunction with the Time Interval field, will determine whether a security violation has occurred.

- Time Interval

  This field requires the entry of the time period in which the login security violations must occur. The range for the time interval is one minute to eight hours (00:01 to 07:59), and is entered in the form xx:xx. For example, if you want the time interval to be one minute, you would enter 00:01. If you want the time interval to be seven and one-half hours, you would enter 07:30.

- Activating Station

The Activating Station must have an LSVN call button to activate and deactivate this feature.

When the SVN Remote Access Violation Notification Enabled field is enabled, the following additional fields appear on the Feature-Related System Parameters screen:

- Originating Extension

  This field requires the entry of an unassigned extension that is local to the switch and conforms to the dial plan for the purpose of originating and identifying SVN referral calls for remote access barrier code violations.

  The originating extension initiates the referral call in the event of a login security violation. It also triggers the display of the appropriate alerting message on the referral destination's display module.

  NOTE:
  The SVN feature button does not have to reside on the referral destination station. It can be administered on the maintenance alarm terminal, but must be active if referral calls are to be placed.

One SVN feature button (per system) may be assigned for each component of the SVN feature (one for login security violations and one for Remote Access security violations).

Hardware and Software Requirements

No additional hardware or software is required.

Selected Communications Mode (SCM) Upgrades

Description

The SCM configures conferences as endpoints join using the highest common approach for optimum audio mode, video resolution, and minimum picture interval (MPI). This feature also automatically upgrades the audio mode, video resolution, and MPI as endpoints leave the conference.

Consider the following example. If all the endpoints in a given conference support G.728, the audio mode is automatically set to G.728. If an endpoint joins the conference but does not support G.728 (and the conference does not require data sharing), all endpoints are downgraded to the audio mode that the new endpoint supports. If the new endpoint subsequently disconnects from the conference, all the endpoints are upgraded again automatically back to G.728. The same is true for upgrade and downgrade of QCIF or CIF video resolution and MPI.

Considerations

None.

Interactions

SCM upgrade interacts with the following features.

- Dynamic Resizing

  Dynamic resizing allows the automatic upgrading and downgrading during active conferences.

- Cascading

  SCM upgrade or downgrade is not allowed in a cascaded conference.

- Rate Adaptation

  SCM upgrade and downgrade are allowed in a conference that uses rate adaptation.

Administration

SCM upgrade can be administered on a per-conference basis.

Hardware and Software Requirements

None.

Service Flags

Description

Endpoints from several vendors tend to crash whenever these endpoints receive certains standard BAS commands/caps. To prevent this problem the Basic/Enhance Service Flag in the Conference Record can be set to *basic* (default). This disables commands/caps that are known to cause problems. If no such problems are apparent, the flag should be set to *enhanced*, where appropriate. Also, the basic/enhanced service and application compliance flags allow the types of messages allowed in a given conference to be identified.

Some endpoints do not operate properly with T.120 Data Conferencing messaging and must be identified in the conference to prevent problems. If all the endpoints in a conference can use the T.120 Data Conferencing messaging, the application compliance flag is set to allow those messages.

Considerations

None.

Interactions

The service flags feature interacts with the following features:

- T.120 Data Conferencing

  The application compliance flag is pertinent only to systems that have the T.120 Data Conferencing feature option.

Administration

Each conference record must be administered at conference reservation for these flags.

Hardware and Software Requirements

None.

SG4 Video Mode

Description

The Standard Proprietary feature allows the MCU to support PictureTel endpoints utilizing SG4 proprietary video protocols in a standards format. The MCU supports these protocols via the PictureTel extension to the ITU-TSS H.230/H.243 conferencing standards. Also, the MCU provides the proprietary video within the H.221 frame structure.

Considerations

SG4 is supported for bandwidths between 112k and 1920k inclusive.

SG4 is controlled via administration of a port capacity limit. It is available in 16-port increments, and it has a maximum capacity of 96 ports.

The MCU can simultaneously support H.261 video and standard proprietary SG4 video on separate simultaneous conferences. (Two different modes within one conference result in the noncompliant endpoint being audio-only.)

Availability

SG4 is available on all models. The range is 16 to 96 ports in 16-port increments.

Interactions

The following features interact with SG4:

- Audio Add-On

  A maximum of 25 Audio Add-On parties can be added to an SG4 conference.

- Still-Frame Graphics

  SG4 Standard Proprietary Still-Frame Graphics is supported.

- Selected Communications Mode (SCM)

  The Highest Common Video SCM is not applied to SG4 conferences. However, the Highest Common Audio SCM is applied to SG4 conferences.

- H.261 Video

  If an endpoint that supports only SG4 attempts to join a conference that is administered as H.261-down-only, H.261-w/upgrade, H.261-CIF-Floor, H.261-CIF-Now, H.261-QCIF-Now, H.CTX, or H.CTX+, the endpoint joins as a secondary (audio-only) terminal.

NOTE:
The Conference Reservation and Control System does not allow this to be administered.

The same applies to an H.261 endpoint that attempts to join an SG4 conference.

- Dynamic Conference Resizing

  All fields that can be changed on an active conference can be changed for an SG4 conference. However, the video mode may not be changed to or from SG4 while a conference is active.

- Rate Adaptation

  Rate Adaptation is supported for SG4 conferences.

- Cascading

  An SG4 conference can be cascaded with another similarly configured conference on an MCU. An SG4 conference can be cascaded with another SG4 conference on a non-Lucent Technologies MCU if the Lucent Technologies MCU is the primary MCU. Interoperability as a secondary MCU is not guaranteed.

- Passwords

  SG4 conferences may be password protected.

- T.120 Data Conferencing

  T.120 Data Conferencing may be enabled on SG4 conferences.

- Universal Conference Control

  An SG4 conference supports one UCC endpoint.

Administration

To establish an SG4 conference, enter “SG4” into the “Video Mode” field of the Conference Record form.

SG4 port capacity is administered separately from the total MCU port capacity. The MCU can be administered with an SG4 port capacity greater than the system capacity; however, the system port capacity takes precedence. If the total MCU port capacity is 12, and if the SG4 port capacity is 16, only 12 ports can be active simultaneously. Otherwise, the system would support a combination of H.320 and SG4 ports up to the system capacity (although the number of SG4 ports would be limited by the administered SG4 capacity).

Hardware and Software Requirements

None.

Speed Match

Description

Speed Match enables endpoints with varying transfer rates to participate in the same conference. The feature enables low-speed desktop video endpoints to interwork with high-speed group system video endpoints in a multipoint conference with full video and audio capabilities. Therefore, for example, 128k endpoints can join and participate in the same conference as 384k endpoints, and each endpoint can see and hear the other endpoints.

Considerations

Speed Match is an optional feature for all Lucent Technologies MCU models. The feature is available in both full screen and Continuous Presence *Plus* (CPP) conferences. Speed Match can be enabled or disabled in a conference only when the conference is inactive. The feature requires the TN2237C, vintage 16 or higher Video Processor circuit pack.

The MCU supports a maximum of eight Video Processor circuit packs. These circuit packs can accommodate a mixture of full screen-only Speed Match conferences, CPP Speed Match conferences, CPP-only conferences, H.263/H.261 Transcoding conferences, and Executive Conferencing conferences. The number of simultaneous conferences using some form of video processing is controlled by the number of each of the three right-to-use permissions as displayed on the Customer-Options form. In particular, the sum of the CPP-only, full screen Speed Match (SM), combination SM/CPP, H.263/H.261 Transcoding, and Executive Conferencing right-to-use permissions may not exceed eight since one dedicated Video Processor circuit pack is required to support each of these conference types. CPP permissions are valid only for non-Speed Match, single transfer rate CPP (or CPP-only) conferences. SM permissions are valid only for full screen SM or H.263/H.261 Transcoding conferences. However, Conferencing Suite permissions can be used to provide a CPP-only conference, a full screen-only SM conference, an SM conference that also provides CPP video to and from endpoints at various transfer rates, an H.263/H.261 Transcoding conference, or an Executive Conferencing conference. Therefore, as long as all of the available Conferencing Suite right-to-use permissions have not been used, any such available permissions may be applied to administer the conference types listed in the previous sentence.

Speed Match conferences support the following transfer rates: 56/64k, 2x56k, 2x64k, 112k, 128k, 224k, 256k, 336k, 384k, 448/512k, and 672/768k. At the 448/512k and 672/768k transfer rates, the maximum number of simultaneous SM conferences supported by the system is reduced. (See the “Administration” section.)

The feature supports H.320, V.35, and Dedicated Access endpoints as well as multirate, BONDing, and cascaded network connections. Also, UCC and Audio Add-On endpoints interwork in Speed Match conferences. Both Meet-Me and Dial-Out endpoints are supported. UCC can be used in CPP Speed Match conferences to transition from quad screen to full screen and vice versa. The feature does not support H.CTX, H.CTX+, MRV, or SG4 endpoints.

For cascaded MCUs, Speed Match hardware and software are required only at the primary MCU. The primary MCU conference can be configured as a Speed Match conference. The secondary MCU conference can be configured as a full-screen conference with the bandwidth set to one of the operating transfer rates in the primary MCU conference. This enables a larger number of participants in a Speed Match conference. Also, the secondary MCU conference cannot be a Speed Match or Continuous Presence Plus conference.

Speed Match is available in H.320 conferences at a maximum of 15 frames per second (fps). Endpoints within a Speed Match conference can operate at a maximum of four different transfer rates from the transfer rates supported for Speed Match. Each transfer rate can have a different video format and video frame rate.

Regardless of their transfer rates, all the speed-matched endpoints within a CPP SM conference see the same quad screen image. For example, in a CPP conference with one 2x64k endpoint and three 384k endpoints, the 2x64k endpoint sees itself in one quadrant and the speed-matched 384k endpoints in the other quadrants. The 384k endpoints see the other 384k endpoints as well as the 2x64k endpoint in the same CPP quad image. In a full screen SM conference, the broadcaster sees the last VAS party, and the other endpoints see the broadcaster.

An endpoint that does not arrive at the pre-administered endpoint transfer rate is allowed to join the conference as a secondary, audio-only endpoint. However, multirate calls are denied if the endpoint is not administered as a multirate endpoint, or if the endpoint is administered as a multirate endpoint, but the call transfer rate is higher than the administered transfer rate. All CPP features and functions operate within Speed Match conferences.

Speed Match is enabled and disabled on a conference basis. The MCU supports speed-match as well as non-speed-match conferences simultaneously.

Full-screen and CPP Speed Match conferences support Voice-Activated Switching (VAS) and Chair Control. VAS is used to determine the image seen by all the endpoints. All the image selection rules and algorithms that apply to VAS conferences apply to Speed Match VAS conferences; this includes mixing images in CPP conferences with five or more parties. Also, Broadcast and Presentation conference modes are supported in non-CPP full-screen speed-matched conferences. This allows Broadcast and Presentation conferences to support endpoints at different transfer rates.

If the required TN2237C circuit packs(s) are not available at the start of a Speed Match conference, the speed match for that conference is disabled. Also, the conference converts to full screen at the primary transfer rate; this allows the endpoints at the primary transfer rate to have video and audio capabilities. Depending on administration, speed-matched endpoints coming into such a conference may function as audio-only endpoints.

Interactions

Speed Match conferences support the following features, functions, and capabilities in the same manner as non-Speed Match conferences: Cascading, Meet-Me conferencing, and passwording.

Speed Match conferences do not support the following features or capabilities: Executive Conferencing, H.263 video mode, OneNumber Access, T.120 Data Conferencing, Lo-/Hi-Interworking, Rate Adaptation and Still-Frame Graphics.

The following features or functions interact with Speed Match with specific changes or considerations as noted.

- Audio Modes and Interworking

  ClearPresence @716 Not Optioned. Without this option, audio mode interworking is allowed in a Speed Match conference whenever the “automatic” audio mode is selected. Endpoints with G.728 audio mode interwork with G.711 audio mode endpoints. Similarly, G.722 audio mode endpoints interwork with G.711 audio mode endpoints. However, G.728 audio mode endpoints do not interwork with G.722 audio mode endpoints.

  The system automatically selects the highest common audio for endpoints operating at the same transfer rate. For example, in the same Speed Match conference, G.728 can be the highest common audio for all the 112k endpoints in the conference, and G.711 can be the highest common audio for all the 384k endpoints in the conference.

  G.711/G.728 interworking allows a 112/128k endpoint to operate at G.728 audio mode (lower bandwidth audio); endpoints in the same conference with a rate higher than 112/128k can operate at G.711 audio mode. G.711/G.728 interworking is selected if any of the transfer rates pre-administered in a conference is 112/128k.

  If all the transfer rates in a conference are higher than 112/128k, G.722 is selected as the highest common audio mode. Endpoints that do not support G.722 audio mode operate at G.711 audio mode in the same conference. For example, in the same Speed Match conference, G.711 can be the highest common audio for all the 336k endpoints in the conference, and G.722 can be the highest common audio mode for all the 384k endpoints in the conference.

  All the automatic audio interworking decisions for a Speed Match conference are made once the first endpoint joins the conference and, accordingly, the conference status changes to “in-use.” Therefore, if a

112/128k transfer rate is not administered in the conference before the conference status changes to "in-use," and if a 112/128k endpoint is subsequently added dynamically, the audio interworking is not changed to accommodate G.728 at the time this endpoint is added. As a result, a dynamically added 112/128k endpoint in this case has a lower video bandwidth because the G.711/G.722 audio requires a higher bandwidth.

No audio interworking is allowed whenever an audio mode for a conference is administered to a *fixed* mode of G.711, G.722. or G.728. Any endpoint of any rate and not meeting the fixed audio mode is forced to operate as a secondary, audio-only endpoint.

ClearPresence @716 Optioned. With this option, each endpoint experiences audio based on its capabilities and the audio mode of the conference as specified in Table B-11 on page B-64.

With this option, G.728 and G.722 can interwork in Speed Match conferences. Endpoints joining at a transfer rate less than or equal to 2B can experience G.728 audio; endpoints joining at a transfer rate greater than 2B transfer can experience G.722 audio.

- Continuous Presence *Plus*

  Continuous Presence *Plus* can be disabled within a CPP Speed Match conference. If this happens, the conference changes to a full screen Speed Match conference. Also, if CPP Speed Match ports are administered in the System Parameters Customer Options form, and if there are no CPP Speed Match conferences active at a given time, the appropriate number of MCU resources are freed up to allow one or more additional CPP-only or full screen Speed Match conferences.

- Dial-Out

  The Dial-Out rate within a Speed Match conference can be either the default primary rate or an additional transfer rate.

- Dynamic Conference Resizing

  A endpoint that is added to an ongoing Speed Match conference can be speed-matched. The newly added endpoint can have one of the following transfer rates: the default primary transfer rate; an existing additional rate at which another endpoint in the conference is administered; or a new additional transfer rate, as long as there are no more than three additional transfer rates in the conference. For each scenario, the added endpoint interworks with the other endpoints by matching its transfer rate to the other operating transfer rates within the conference.

  An endpoint can be removed dynamically from a Speed Match conference without disrupting the ongoing speed matching within the conference. In each case, the endpoint is dropped if it has been used, and the remaining conference continues to operate in speed matching mode. Also, the endpoint level resources are deallocated.

The transfer rate of an endpoint cannot be changed *dynamically.* However, you can change an endpoint's rate by first removing it from the conference and then adding it back to the conference at the desired new rate.

Finally, CPP within a Speed Match conference can be disabled; this changes the conference to a full screen Speed Match conference.

An endpoint at a specific transfer rate can drop from a Speed Match conference without terminating the speed matching for that endpoint's transfer rate or any other transfer rates in conference. If an endpoint at a specific transfer rate drops, the remaining endpoints in the conference at that transfer rate continue to speed match. The remaining endpoints at other transfer rates also continue to operate and speed match with other endpoints.

- Lip Synchronization

  This feature is supported in both full screen Speed Match conferences and CPP Speed Match conferences.

  Full-screen SM and CPP/SM conferences support the automatic ("auto") and "custom" audio delay options of Lip Synchronization. For the "custom" option, the customer can define his or her own set of custom "default" audio delay values.

  Automatic Lip Sync is set by default. It can be disabled by setting the appropriate *Lip Sync* field in the Conference Record form to **none**.

  Automatic Lip Sync continues to use the current algorithm used to provide Lip Sync in a CPP conference. However, since there can be several transfer rates operating in a Speed Match conference, the audio delay for each transfer rate is computed separately.

- OneNumber Access

  Speed Match supports OneNumber Access only for Audio Add-On endpoints. Multimedia endpoints are not supported.

- T.120 Data Conferencing

  T.120 Data Conferencing at the following fixed MLP rates is allowed within SM conferences: 6.4k, 8k, 14.4k, 16k, 22.4k, 24k, 30.4k, 32k, and 38.4k. Unless the ClearPresence @716 RTU is optioned, the feature at the following fixed MLP rates is not allowed within SM conferences: 40k and 46.4k. (This RTU allows all fixed MLP rates within SM T.120 Data Conferencing conferences.) Also, variable rates (indicated by value "var_mlp") are not allowed.

- Video Modes

  Speed Match is supported for the standard H.261 fixed or automatic video modes. Proprietary video modes (for example, H.CTX+, MRV, and SG4) and H.263 video mode are not supported.

  The H.261 video mode can be administered for an automatic selection of video format and the video frame rate (H.261-down-only or H.261-w/upgrade), or it can be fixed to a specific format and frame rate

(H.261-CIF-floor, H.261-CIF-now, or H.261-QCIF-now video format along with fixed or automatic frame rate). The maximum frame rate is restricted to 15 fps. The administered video mode applies to all the transfer rates in a speed-matched conference, and the mode is managed separately per transfer rate. The default and recommended video mode is “auto.”

If the video mode is one of the fixed modes (H.261-CIF-floor, H.261-CIF-now, or H.261-QCIF-now), all the endpoints must operate at the designated fixed video mode. For example, if the video mode is H.261-CIF-floor at maximum of 15 fps, all the endpoints, regardless of the transfer rate, must operate at H.261-CIF-floor at a maximum of 15 fps. Any endpoint that is incapable of that video mode becomes a secondary, audio-only endpoint. However, if the mode is one of the automatic modes (H.261-down-only or H.261-w/upgrade), the video format (CIF/QCIF) and the video frame rate can fluctuate within each transfer rate in a conference. For example, in a speed-matched conference with two transfer rates (for example, 384k and 2x56k), all the 384k endpoints can operate with CIF format at maximum of 15 fps, and all the 2x56k endpoints can operate with QCIF at 7.5 fps.

Note that the highest common video modes for all the transfer rates can be the same if the highest common video mode for the endpoints in each transfer rate domain settles down to be the same video mode. On the other hand, there can be up to four different video modes operating in a speed-matched conference (that is, one video mode for each transfer rate in the conference).

Administration

Speed Match is an optional feature for all Lucent Technologies MCU models. The feature is available in both full screen and Continuous Presence *Plus* (CPP) conferences. Speed Match can be enabled or disabled in a conference only when the conference is inactive.

The MCU supports a maximum of eight Video Processor circuit packs. These circuit packs can accommodate a mixture of conference types, including SM.

The right-to-use option for Speed Match is preset by the Video Technical Center (VTC) upon delivery to the customer. To disable the feature, the port capacity for Speed Match and CPP/SM must be set to **0** on the System-Parameters Customer-Options form. To enable the feature, the port capacity must be set to a non-zero number corresponding to the “right-to-use” number for the feature desired by the customer.

A maximum of eight conferences administered via the *CPP/SM* field at transfer rate 448/512k is allowed. A maximum of six such conferences administered at transfer rate 672/768k is allowed.

NOTE:
These conference maximums assume a three-party BONDed configuration with two parties administered at the indicated “high” rate and one-party administered at the 2B rate.

By default, all the endpoints within a Speed Match conference operate at the primary transfer rate. The transfer rates of the endpoints (except Audio Add-On and UCC endpoints) that will operate at a rate other than the primary transfer rate must be administered.

For complete details on administration, refer to "Administering Speed Match Conferences" on page 2-237.

Hardware and Software Requirements

The feature requires one TN2237C, vintage 16 or higher Video Processor Plus circuit pack for each Speed Match conference administered. Therefore, since the MCU can support a maximum of eight Speed Match conferences, eight TN2237C circuit packs are required to support full Speed Match capabilities per MCU. These circuit packs are treated as a pool of resources. Depending on the administered capacities, some of the circuit packs can be used for full screen or CPP Speed Match conferences, and the rest of the circuit packs can be used for CPP conferences.

System Measurements

Description

System measurements provide reports on items such as trunk group usage, hunt group usage and efficiency, maintenance alarm terminal activity and efficiency, and security violations.

Individual reports are available as detailed in Chapter 3 of this document.

All reports are on-demand reports. None are given automatically. Reports are available on the remote administration terminal. The reports can be printed if a printer is associated with the terminal. The reports can also be scheduled to print at the system printer via the Report Scheduler and System Printer feature.

Considerations

Reports provided by System Measurements contain data that is useful to determine group efficiency.

Traffic measurements are automatically accumulated by the system and are available on demand. However, reports are not archived. If needed, reports must be requested periodically. Obtaining a printed copy can aid in maintaining a history of the system traffic.

Detailed information of each call handled by a specific trunk group, if required, must be provided by the CDR feature. Processed CDR data can also provide detailed information on trunk group usage. However, if individual call details are not required for bill-back or cost-allocation, System Measurements should be considered as the means to determine and maintain trunk group efficiency.

Interactions

None.

Administration

Measurements for the Coverage Paths, Coverage Principals, Route Patterns, Trunk Group Hourly, and Wideband Trunk Group Hourly reports are only collected for objects that have been administered using the **change meas-selection** commands.

Hardware and Software Requirements

System administration terminals are required to monitor system measurements. A system printer is required to generate paper copies of the reports. No additional software is required.

System Status Report

Description

The system status report allows the user to view data associated with the maintenance alarm terminal, major and minor alarms, and traffic measurements. The information is displayed on the Management Terminal, and it presents a basic picture of the system condition. The report can only be displayed by the system administrator and maintenance personnel.

The Status Report is displayed by entering one of the commands listed below. Once the command is entered, the system continually displays the report until it is canceled.

NOTE:
Canceling a **monitor system view1** or **monitor system view2** report results in automatically logging the user off the system.

- **monitor system view1**

  This command displays the following information:

  — Activation status of the maintenance alarm terminal (updated every minute)

  — Maintenance status, which includes major and minor alarms for trunk ports, terminal ports, and all maintained objects in the system except terminals and trunks (updated every minute)

  — Traffic measurements for trunk groups, hunt groups, and the maintenance alarm terminal (updated every hour)

- **monitor system view2**

  This report is a subset of the view1 report and displays the same information listed for the view1 report except the last hour's measurement for the hunt groups.

Considerations

In addition to providing status reports, this feature also provides an indication that the administration terminal is functional. Any attempt to stop the ''monitor system view1/view2'' reports logs the administration terminal off the system. Therefore, no unauthorized administration can be performed.

Interactions

None.

Administration

None required.

Hardware and Software Requirements

No additional hardware or software is required.

T.120 Data Conferencing

The MCU supports a multipoint data conferencing feature based on the H.221 MLP and the T.120 protocol stack, as specified in ITU-T Recommendations T.122, T.123, T.124 (Generic Conference Control) and T.125 Multipoint Communication Service (MCS). T.120 Data Conferencing is a customer option available in standalone MCU environments.

Description

T.120 Data Conferencing supports multipoint service for the interactive, network-independent transport of data. The feature is not directly used by conferees but is accessed via the use of other conference applications that use T.120 Data Conferencing primitives to access this feature via the Lucent Technologies MCU. Users within a T.120 Data Conferencing conference can set up dynamic sub-conferences where any one of the users can broadcast data.

Considerations

The Lucent Technologies MCU provides the following types of T.120 Data Conferencing services:

- As many as 24 T.120 Data Conferencing connections per ESM
- Both static and dynamic channel IDs
- Normal send and uniform send of multicast data
- Multiple users per node

These services support T.120 Data Conferencing applications such as whiteboard, application sharing, and file transfer.

In addition, the T.124 stack is called Generic Conference Control (GCC), and it is based on a protocol implementation from Databeam. GCC provides several conference control functions, including the following:

- Establishing and terminating conferences
- Managing the roster of terminals participating in a conference
- Managing the roster of applications at each terminal
- Computing application abilities
- Providing registry services for use by applications
- Coordinating conference conductorship

GCC provides equal support for standard and non-standard applications. It enables these applications to declare and discover that they implement specific application protocols. An instance of an application protocol excuting an MCU is called an application protocol entity. Different application protocol entities communicate among themselves in one or more distinct sessions.

The feature is supported within H.320 MCU conferences at all network transfer rates between 1B and 1920k. All fixed MLP rates between 6.4k and 46.4k are supported within non-SM T.120 conferences, with one exception: MLP rates 40k and 46.4k are not supported within nx56k conferences.

Variable rate MLP is also supported. Variable-rate MLP uses the intial channel capacity that remains in the initial channel (56/64k per second) after deducting the framing signals (1.6k per second) and the audio bit rate (16k, 48k, or 56k per second). Variable-rate MLP can use fixed-rate MLP channels.

The Expansion Services Module (ESM) is required to support T.120 Data Conferencing.

Interactions

The feature supports Continuous Presence *Plus* and H.263 video. The feature can use any MLP rate within same-rate, full-screen conferences. Also, both enhanced and basic endpoints can be included into such conferences. The feature does not support Cascading and H.263/H.261 transcoding.

T.120 Data Conferencing feature interacts with the following features with the changes or considerations noted:

- Audio Mode

  The feature supports the G.711, G.722, and G.728 audio modes, as well as the “auto-w/upgrade” and “auto-down_only” options. However, G.711, G.722, “auto-w/upgrade,” and “auto-down_only” are allowed only for MLP rate 6.4k.

  SM T.120 Data Conferencing conferences are allowed with audio mode “ClearPresence @716” and MLP rate “fixed” (and not “var_MLP”). Also, these conferences are allowed with audio mode “auto” and MLP rate “6.4k.”

- Cascading

  The feature is not supported over a cascaded link to another MCU.

- Dynamic Resizing

  T.120 Data Conferencing endpoints can be added or dropped during a conference. However, T.120 Data Conferencing cannot be enabled or disabled for an active conference. The T.120 Data Conferencing option is set at the time the conference is reserved and cannot be changed dynamically.

- Application Compliance Flag

  When the application compliance flag is turned off for an endpoint that supports both var-MLP and G.728, the Lucent Technologies MCU blocks any data sends or receives to or from that endpoint.

- Selected Communication Mode (SCM)

  The SCM of a conference specifies whether or not the conference is using MLP. Whenever the MLP option is off, no MLP data channels are opened for the conference; this is true regardless of whether the endpoints support MLP. Whenever the MLP option is in effect, an MLP channel is opened only to the endpoints that support both var-MLP and G.728 (LD-CELP). Endpoints that do not declare a suitable MLP capability become audio-only secondary endpoints upon joining the conference. Endpoints that declare an acceptable MLP capability but do not declare G.728 (LD-CELP) also become audio-only secondary endpoints.

- Speed Match

  T.120 Data Conferencing at the following fixed MLP rates is allowed within Speed Match conferences: 6.4k, 8k, 14.4k, 16k, 22.4k, 24k, 30.4k, 32k, and 38.4k. The feature at the fixed MLP rate 40k or 46.4k is not allowed within SM conferences unless ClearPresence @716 audio is optioned. Also, variable MLP rates (var_mlp) are not allowed.

  SM T.120 Data Conferencing conferences are allowed with audio mode “ClearPresence @716” and MLP rate “fixed” (and not “var_MLP”). Also, these conferences are allowed with audio mode “auto” and MLP rate “6.4k.”

Administration

T.120 Data Conferencing is activated on a per-system basis. To enable T.120 Data Conferencing, specific fields in a number of forms must be populated correctly. For details, refer to the “Administering T.120 Data Conferencing” section in Chapter 2.

Hardware and Software Requirements

T.120 Data Conferencing requires an ESM module and two TN787K, vintage 6 or higher MMI circuit packs. One MMI (H.221-MMI) is used to terminate the H.221 bit stream and the second one (ESM-MMI) performs the MLP bit rate adaptation. Also required are the TN765 Processor Interface and the TN2207 UDS1 circuit packs.

The MSM and ESM hardware are required for a Lucent Technologies MCU to use the T.120 Data Conferencing features. The MSM and ESM should be within 50 feet of each other.

Terminal Names

Description

Terminal names allows the MCU to query and pass endpoint terminal names as defined in the ANSI243 (ANSI version of H.243) recommendation. With the terminal names option, endpoints can identify themselves and other endpoints involved in a conference.

Considerations

None.

Interactions

None.

Administration

Terminal names can be administered on a per-conference basis.

Hardware and Software Requirements

None.

Universal Conference Control

This feature is a customer option that enables an endpoint to manage an H.320 conference out-of-band. Universal Conference Control (UCC) is intended to provide a number of powerful services similar to and in lieu of H.243 Chair Control.

Description

The controlling terminal for UCC is a regular touchtone telephone, such as that used by an Audio Add-On party. This terminal *need not be* H.320-compliant. The terminal is joined to the conference solely for control purposes. As such, the terminal does not receive audio, video, or data from the conference it is managing, and it does not transmit this media to the H.320 conference multiplex. To join a conference, the UCC controlling terminal dials a specific MCU-extension.

Unlike Chair Control, requests from the UCC controlling terminal reach the MCU through a touchtone interface rather than via BAS commands. Accordingly, the user can use touchtone control to send commands to the MCU to do the following:

- Control the broadcast video source (the return video of the selected broadcaster is voice energy-switched)
- Change the conference mode to Voice Activated Switching (VAS) (this cancels any previous request to force the video broadcaster and revert back to the VAS method of video switching)
- Mute one endpoint or all endpoints except for a specific endpoint
- Unmute a single endpoint or all endpoints
- Cause selected endpoints to be dropped from the conference
- Terminate the conference (this request drops all the endpoints connected to the conference, including the controller itself)
- Adjust the audio volume for an Audio Add-On party
- Manipulate the most recent endpoint to join the conference (for example, broadcast the endpoint, mute the endpoint, etc.)

In addition, UCC can perform the following actions if Continuous Presence *Plus* is enabled:

- Transition full-screen image to quad-screen video, and vice-versa
- Place a specified endpoint into a specified quadrant

A number of options (explained later in this section) are available to perform these functions.

Considerations

A maximum of 25 UCC endpoints is allowed per MCU conference. UCC is not allowed in a Chair conference.

A tone is provided to the UCC controller after it is successfully connected to the conference. A tone is also given to the UCC controller to indicate positive or negative feedback after every command entered.

A positive feedback tone is given if the MCU is able to switch to the specified broadcaster. The MCU plays a negative feedback tone and disregards the controller's request to broadcast an endpoint if the endpoint does not satsify certain conditions. (See the "Select Broadcaster" section).

The UCC controller can also cause selected endpoints to be dropped from the conference, including itself. In addition, the UCC controller can terminate the conference. In so doing, it drops all endpoints connected to the conference, including the controller itself.

Options

This section discusses in full the UCC options. The following table summarizes the UCC options and access methods.

Table B-17. UCC Services Access

Desired Service	Required Keypad Entry	Description
Unlock Video	****8**	Releases any fixed broadcaster and returns the conference to Voice Activated Switching (VAS) mode.
Select a site to be a video broadcaster	*** *N***	Locks in the specified site ***N*** as the broadcaster (broadcasting site). The broadcaster sees the site that is currently speaking, or, if no other sites are speaking, the site that spoke most recently.
Browse sites (after selecting a broadcaster)	****2**	Shows the video from one specific site to the broadcasting site. All other sites, including the site whose video is being shown to the broadcaster, continue to view the broadcaster. Pressing "#" allows the broadcaster to view each of the other sites in the prescribed order.
Conduct Rollcall of all sites	****7**	Shows the video from a specific site to all the sites. Pressing "#" shows the video from each of the other sites, in the prescribed order, to all the sites in the conference.
Scan to the Next Site (when in Rollcall or Browse option)	#	Pressing "#" displays the next site in Rollcall and Browse modes. The sites are scanned in order of their site numbers on the Conference Record form.
Change image from full-screen to quad-screen	****4#**	Partitions the full-screen image into four areas (quadrants). This allows up to four different sites to be viewed at one time.
Assign site to quadrant 1 (top left), 2 (top right), 3 (bottom left), or 4 (bottom right)	****4*QN***	Enables the user to choose what is seen in each quadrant (***Q***). The video from a single site can be locked into a quadrant. Also, a quadrant can display video according to VAS. To set a quadrant according to VAS, use ***0*** as the site number (***N***).
Change image from quad-screen to full-screen	****1**	Returns the quad-screen image back to full-screen. The layout of the quad-screen is remembered.
Transition to upper-panoramic three-image layout	****46**	Displays a three-image CPP screen layout where the top half of the screen displays one image while the bottom half of the screen is segmented into two screen displays.

Table B-17. UCC Services Access — *Continued*

Desired Service	Required Keypad Entry	Description
Transition to lower-panoramic three-image layout	****47**	Displays a three-image CPP screen layout where the bottom half of the screen displays one image while the top half of the screen is segmented into two screen displays.
Transition to dual-panoramic screen layout	****48**	Displays a two-image CCP screen layout where the top half of the screen displays one image and the bottom half of the screen displays one image.
Transition to Executive Conferencing Mode	****49**	Displays a dual-panoramic screen layout for the three broadcaster endpoints and an upper-panoramic screen layout for the other conference participants.
Mute site	****61 *N***	Mutes the incoming audio of site ***N***. Other sites do not hear anyone at the muted site, but the muted site continues to hear audio from other unmuted sites. **NOTE**: This command overrides the “mute” button of the video equipment at Site *N*.
Unmute site	****62 *N***	Unmutes the incoming audio of site ***N.*** This site can be heard by all the other sites. **NOTE**: This command does not override the “mute” button of the video equipment at Site *N*.
Solo Audio	****63 *N***	Mutes the audio of all the sites except site ***N.*** Therefore, only the “solo audio” site can be heard by all the other sites.
Unmute all sites	****6#**	Unmutes all the sites. All the sites can speak and be heard.
Adjust Audio Volume for Audio Add-On Party	**64**	(This is the command code used within the commands for the specific services provided.)
— Decrease	****64 *N* 7**	Decreases the volume of endpoint ***N*** by 3 db.
— Reset	****64 *N* 8**	Resets the volume of endpoint ***N*** to the default (6 db loss).
— Increase	****64 *N* 9**	Increases the volume of endpoint ***N*** by 3 db.

Table B-17. UCC Services Access — *Continued*

Desired Service	Required Keypad Entry	Description
Manipulate Most Recently Joined Endpoint	**99**	(This is the command code used within the commands for the specific services provided.)
— Broadcast	***99**	Broadcasts the most recently joined endpoint.
— Mute	****6199**	Mutes the most recently joined endpoint.
— Unmute	****6299**	Unmutes the most recently joined endpoint.
— Solo Audio	****6399**	Performs Solo Audio on the most recently joined endpoint.
— Set in Quad	****4Q99**	Sets the most recently joined endpoint into a quadrant.
— Drop	****399**	Drops the most recently joined endpoint.
Drop site	****3*N***	Drops a single site ***N*** from the conference.
End conference	****91**	Drops all sites in the conference, including the UCC port, and terminates the entire conference.

Select Broadcaster

This option allows the controller to select the video broadcaster. The broadcaster is viewed in full screen. This service is allowed in VAS conferences but not in conferences with Presentation mode since this mode implies that the video broadcaster (that is, the Presenter) may not be changed. Also, this service may be used in conferences with or without Continuous Presence *Plus*.

When the controller activates this service, the conference operates in a manner similar to a Chair conference, in which the Chair terminal sends the Video Command Broadcast (VCB) BAS code. The selected video broadcaster is essentially “locked” as a result of this request.

When the controller enters the Select Broadcaster command followed by an endpoint number in the range of 1 through 25, the MCU broadcasts the full-screen video of the specified endpoint to other endpoints in the conference. All the other endpoints view the video of the selected endpoint in full screen. The return video of the selected broadcaster is chosen based on the voice-activated switching (VAS) algorithm. The broadcaster designated by the controller remains the video broadcaster until any one of the following occurs:

- Controller chooses another endpoint with the Select Broadcaster command.
- Controller requests the Rollcall feature.
- Selected broadcaster drops or becomes an invalid video source.
- Video locks are cleared with the Unlock Video command.

  NOTE:
  The current broadcaster may actually remain the broadcaster if the VAS algorithm selects that endpoint.

- Image transitions to quad screen with the Force Image to Quad-Screen command.
- UCC drops.
- Conference ends.

This service can be requested at any time during the conference. For example, the controller may select a new broadcaster while a fixed broadcaster is currently in force, while an MCV broadcaster is in force, or when the conference is operating in VAS mode. In all cases, the MCU attempts to switch to the specified endpoint. For an MCV broadcaster, the controller's request to change the broadcaster overrides the current MCV broadcaster's status. If Rollcall or Browse is in effect, a request to select a broadcaster cancels these services.

If the conference is currently viewing a quad-screen image, a request for this service first causes an automatic transition to full screen with the specified endpoint as the broadcaster. Video locks imposed on the quadrature image remain unaffected. For example, a previous request to change the composition of the quad screen via an administrative change to the conference record, CRCS Agent control, the MT conference control interface, or the UCC Set Quadrant command are unaffected. If the image transitions back to quad screen, the most recent composition requested is shown.

The MCU broadcasts an endpoint if it has video suppressed (sent VIS), but not if it is an invalid video source (sent video-off, etc.). If an endpoint suppresses its video while it is the fixed broadcaster, the MCU does not switch away. However, the MCU removes the lock on the selected broadcaster if the endpoint becomes an invalid video source. In such a case, another broadcaster is chosen from the speaking endpoints.

The controller hears a positive feedback tone if the MCU is able to switch to the specified broadcaster. The MCU plays a negative feedback tone and disregards the controller's request to broadcast an endpoint if any of the following are true:

- Endpoint number is not in the range 1 through 25.
- Specified endpoint is not video source capable.
- Specified endpoint is not reserved in the conference.

- Specified endpoint is not joined to the conference.
- Conference is administered as a Presentation conference.

Unlock Video

This service allows the controller to cancel any video locks imposed on the conference by CRCS Agent Control, the MT conference control interface, or UCC commands. This command essentially returns the conference back to the operation mode specified by conference administration. This service may be used in conferences administered with VAS or Presentation mode. In addition, it may be used in conferences optioned with Continuous Presence *Plus*.

For a VAS administered conference that is currently viewing a full-screen image, a request for this service cancels any previous control command to lock a video broadcaster. For example, this service cancels any locks created by the Select Broadcaster, Rollcall, or Browse commands, and it returns the conference back to the VAS administered method of selecting a broadcaster. In VAS mode, a video broadcaster is chosen from among those endpoints who are speaking.

For a Presentation conference in which the Presenter is currently viewing a full-screen image of a return video broadcaster that has been locked due to the Browse command, the Unlock Video command cancels the fixed status of the return broadcaster and returns to the VAS administered method of selecting the return video.

If the conference is viewing a quad-screen image in either a Presentation or VAS conference, this service cancels all locks created by any of the control interfaces (including the UCC Set Quadrant command) and returns the conference back to the administered view of the conference.

The controller hears a positive feedback tone if the MCU is able to honor a request for this service. This request is disregarded by the MCU and a negative feedback tone is provided if the current broadcaster has not been designated by a controller (for example, the current broadcaster is an MCV broadcaster). This service does not cancel any mute controls requested by the Mute and Solo Audio commands.

If the conference is already operating with its administered defaults, the MCU takes no action. However, a positive feedback tone is provided to indicate that the conference is in the requested mode.

This feature does not prevent the controller from controlling the conference. It simply clears all controlled video locks. Services that lock video broadcasters (for example, Set Quadrant, Select Broadcaster, Browse, and Rollcall) may be requested by the controller any time after the Unlock Video command is issued..

This service is similar to canceling a Video Command Broadcast request (cancel-VCB) in Chair Controlled conferences.

Drop Endpoint

The controller can drop an endpoint from the conference at any time. When this command is issued, the MCU disconnects the selected endpoint from the conference and drops all calls associated with that endpoint. If the conference is optioned with tones, an exit tone is heard by the remaining parties on the conference when the endpoint is disconnected. If the endpoint is a dial-out endpoint, the MCU does not attempt to retry that endpoint. After being dropped, a dial-in endpoint may redial its mcu-extension and be reconnected to the conference.

The controller hears a positive feedback tone if the MCU is able to drop the specified endpoint. The MCU plays a negative feedback tone and disregards the controller's request to drop an endpoint if:

- Endpoint number is not in the range 1 through 25.
- Specified endpoint is not reserved in the conference.
- Specified endpoint is not joined to the conference.

The controller is able to drop any endpoint associated with the conference, including the UCC party itself. If the controller drops itself, no exit tone is heard by the parties remaining in the conference.

This feature is similar to dropping an endpoint with the CCD BAS command (Chair Command Disconnect) in Chair Controlled conferences.

End Conference

This service drops all endpoints joined to the conference, including the controller itself. The effect of this service is the same as all the parties in the conference intentionally hanging up and ending the conference.

For this service, the MCU does not automatically retry dial-out endpoints in the conference. Also, it does not change the status of the conference to "file" (if the conference was administered as "dedicated") or "inactive" (if the conference was administered as "reserved"). For a "reserved" conference, the MCU does not retry dial-out parties until the conference reaches its Stop Time and then becomes active again by changing the conference type to "file" and also the Start and Stop Time.

Since the status of the conference does not change, endpoints may join either a reserved conference that has not yet reached its Stop Time or a dedicated conference after this service is requested. The controller hears a positive feedback tone.

This feature is similar to terminating a conference with the CCK BAS command (Chair Command Kill) in Chair Controlled conferences.

Browse

This service allows a fixed, full-screen broadcaster to scan the other endpoints in the conference. This service is similar to Broadcast with Auto-Scan with one exception: the UCC party controls when the next scan of return video is to take place. Only the return video seen by the broadcaster changes; all other participants continue to view the broadcaster in full-screen.

Browse may be used in VAS or Presentation mode administered conferences in which a full screen broadcaster has been fixed with the Select Broadcaster command. In full-screen Presentation conferences, Browse allows the Presenter to scan in full screen the other endpoints currently in the conference.

This service is invoked by entering the Browse feature access code. The MCU immediately locks the return video on the endpoint with the lowest endpoint number, and a positive tone is given. A request for this service is valid only if the current broadcaster is currently fixed with the Select Broadcaster feature or if the broadcaster is the Presenter in a Presentation conference. This command is denied if:

- Broadcaster has not been fixed with the Select Broadcaster command, and the broadcaster is not the Presenter
- Current broadcaster is an MCV broadcaster

Once in Browse, the "#" key selects the next return video. The endpoints are scanned in order of endpoint number given to each endpoint at conference reservation. The current broadcaster is skipped in Browse unless it is the only party in the conference. A positive tone is given to indicate a switch of new return video.

While in Browse, the return video is essentially “locked,” and speaking conference participants therefore do not cause any video switch. MCV requests are denied while Browse is in effect; however, the request for MCV is remembered. The MCU continues to respond to scan requests as long as the "#" key is pressed.

The MCU broadcasts an endpoint selected for Browse if it has video suppressed (sent VIS), but not if it is an invalid video source (sent video-off, etc.). If an endpoint suppresses its video while it is providing the return video during Browse, the MCU does not switch away. However, the MCU switches to the next endpoint in Browse if the endpoint becomes an invalid video source.

If the return video in Browse drops, the next endpoint in Browse is automatically selected by the MCU.

Browse is canceled if any of the following occurs:

- Another broadcaster is selected with the Select Broadcaster command.
- Controller requests the Rollcall feature.
- Video locks are cleared with the Unlock Video command.

- Broadcaster drops or becomes an invalid video source.
- Conference transitions to quad-screen.
- Conference ends.

Rollcall

This service allows the controller to enable each endpoint in turn to be the broadcast video source. In Rollcall, all participants view the broadcaster in full screen, including the broadcaster itself. (The broadcaster receives looped video.) The UCC party controls when the next endpoint in the conference becomes the broadcaster. Rollcall is allowed only in VAS administered conferences. It is not allowed in Presentation conferences.

It is not necessary to “fix” a broadcaster prior to activating the feature. However, since Rollcall is a full-screen feature, the MCU automatically transitions the conference to full screen if this service is requested while viewing quad screen.

Rollcall is invoked by entering the Rollcall feature access code. If the conference is currently viewing a quad screen, the conference is first transitioned to full screen. The MCU then immediately locks the broadcast video on the lowest numbered endpoint, and positive tone is given.

If there is a full-screen MCV broadcaster, the controller's request to begin Rollcall overrides the current MCV broadcaster's status, and the BAS code *VCR* is sent to the current broadcaster. Rollcall automatically cancels the Select Broadcaster and Browse features.

Once in Rollcall, the “#” key causes the MCU to select the next broadcast video. Positive tone is given to indicate a switch to the new broadcaster. The endpoints are scanned in endpoint number order. While in Rollcall, the broadcast video is “locked” and speaking conference participants do not cause any video switch. Therefore, requests for MCV are also denied while Rollcall is in effect. The MCU continues scanning as long as the "#" key is pressed.

The Rollcall command is denied if at least one of the following is true:

- Conference is administered as a Presentation conference.
- There are no valid video sources in the conference.

The MCU broadcasts an endpoint selected for Rollcall if the endpoint has video suppressed (sent VIS), but not if it is an invalid video source (sent video-off, etc.). If an endpoint suppresses its video while it is the broadcast video during Rollcall, the MCU does not switch away.

Rollcall is canceled if any of the following occurs:

- New broadcaster is selected with the Select Broadcaster command.
- Video locks are cleared with the Unlock Video command.

- Broadcaster drops or becomes an invalid video source.
- Conference transitions to quad screen.
- Conference ends.

Transition Video Image

In a conference optioned with Continuous Presence *Plus*, the UCC controller is able to transition the conference between the panoramic and quad-screen mixed video operations and full-screen switched video. This service provides commands to transition to quad-screen, panoramic, and full screen. Both commands may be used in Voice-Activated Switching and Presentation conferences. In a conference optioned with Executive Conferencing, the UCC controller is able to transition the conference between the dual- or upper-panoramic mixed video operations and full-screen switched video.

Force Image to Dual-Panoramic View

If the conference is currently viewing a full-screen image and this service is requested, the MCU immediately transitions the conference to dual-panoramic view.

Force Image to Lower-Panoramic View

If the conference is currently viewing a full-screen image and this service is requested, the MCU immediately transitions the conference to lower-panoramic view.

Force Image to Upper-Panoramic View

If the conference is currently viewing a full-screen image and this service is requested, the MCU immediately transitions the conference to upper-panoramic view.

Force Image to Executive Conferencing

If the conference is currently viewing a full-screen image and this service is requested, the MCU immediately transitions the conference to Executive Conferencing view.

Force Image To Quad Screen

If the conference is currently viewing a full-screen or any panoramic image and this service is requested, the MCU immediately transitions the conference to a quad-screen operation. When this happens, all controlled video locks imposed on the conference while in the full or panoramic screen are canceled. For example, if Rollcall, Browse, or Select Broadcaster is active and the controller requests a transition to quad screen, a request for any of these services is canceled and is not remembered if the conference subsequently transitions to full screen. In addition, the MCV locking effect on the full-screen broadcast video is broken (and the BAS code VCR is sent to the current broadcaster). Transitioning the image to quad screen does not cancel any Mute Controls activated by UCC.

A request to transition the conference to quad-screen causes no effect if the conference is already operating in quad screen. Positive feedback tone is given to indicate the conference is operating in the requested mode. This service is denied and negative tone is provided if either of the following is true:

- Conference is not optioned with Continuous Presence *Plus*
- Continuous Presence *Plus* is optioned for the conference, but resources have become unavailable.

Force Image To Full Screen

If the conference is currently viewing a quad-screen, panoramic, or Executive Conferencing image and this service is requested, the MCU immediately transitions the conference to full-screen operation. When this happens, the current composition of the quad image is remembered and is not automatically cleared as a result of switching to full screen. However, other control interfaces (including UCC) may change the “current” composition while viewing full screen. Transitioning the image to full screen does not cancel any Mute Controls activated by UCC.

Once the conference is in full-screen mode, it operates according to the video mode administered on the Conference Record form. Specifically, the *Video Mode* field indicates how the MCU should manage video capabilities for the conference while it is in full screen. For example, if *Video Mode* indicates that capability upgrades are enabled for the conference, the MCU initiates a cap exchange as a part of transitioning to full-screen mode.

A request to transition the conference to full screen causes no effect if the conference is already operating in full screen. Positive feedback tone is given to indicate the conference is operating in the requested mode.

This service is never denied since a conference must always be able to transition to full screen.

Set Quadrant

This service allows the UCC controller to change the composition of the quad image. It may be used in both VAS and Presentation conferences that are optioned with Continuous Presence *Plus*. The service allows the controller to broadcast the video of a specific endpoint in a quadrant. It also allows a quadrant to be designated as a VAS quadrant. In a VAS quadrant, the video displayed in that quadrant is chosen from speaking endpoints according to the VAS algorithm. Therefore, this service allows the quadrant to be “locked” as either a “fixed” or VAS quadrant.

To use this service, the UCC controller enters the Set Quadrant command followed by a single-digit quadrant number in the range of 1 through 4. Following the quadrant number is the endpoint number. To designate the quadrant as voice-energy switched (VAS), the endpoint number should be 0. To fix a specific endpoint in a quadrant, the endpoint's number (from 1 to 25) should be entered. A

command to fix an endpoint in a quadrant overrides the current designation of the quadrant. If another endpoint is currently broadcasting in the quadrant, it is removed from the quad image and the video of the new endpoint is displayed.

NOTE:
This endpoint may reappear in a different quadrant due to VAS selection or if a control interface fixes it in a different quadrant.

Placing an endpoint into a quadrant has no effect on other fixed quadrants in which it is already broadcasting. Therefore, if an endpoint is already broadcasting in fixed quadrant 1 and is then placed into fixed quadrant 2, it remains in both quadrants. However, if the endpoint is already broadcasting in a VAS quadrant, the VAS algorithm removes it and replaces it with another speaking endpoint (if available).

There is no restriction on how often a single endpoint can be designated as broadcasting in the quad-image. In other words, the controller is allowed to place a specific endpoint into more than one quadrant. Therefore, for example, if endpoint 3 is already “fixed” in quadrant 1, the controller can also use the Set Quadrant command to place endpoint 3 into quadrant 2, 3, or 4. Similarly, one or more quadrants can be designated as VAS quadrants. With this command, the controller is also able to compose the image such that the video of the "Empty Quadrant Endpoint" occupies one or more quadrants.

Optioning this service has an immediate visual effect only if the conference is viewing a quad-screen image. For example, if the conference is viewing a quad-screen image and a change to quadrant composition is requested, the quad-image is immediately updated to reflect the controller's request. However, if the conference is currently viewing a full-screen image, the requested composition is stored. The conference remains in full-screen mode and does not transition to quad-screen mode until there is an explicit request to do so. When the conference does transition to quad screen, the most recent composition requested by the controller is provided.

The controller hears a positive feedback tone if the composition can be changed as requested. Negative feedback tone is provided if any of the following are true:

- Endpoint number is not 0 (for VAS) or not in the range 1 through 25
- Quadrant number is not a single digit in the range 1 through 4
- Specified endpoint is not reserved in the conference.

- Specified endpoint is not joined to the conference.
- Specified endpoint is the UCC party or an Audio Add-On party
- Specified endpoint is the Presenter.

An endpoint that is an invalid video source or has its incoming video suppressed (because either the MCU received VIS from the endpoint or a control interface requested that video be suppressed) is allowed to be selected as a fixed broadcaster in a quadrant. If an endpoint occupies a fixed quadrant, and if the endpoint's video is subsequently suppressed or the endpoint becomes an invalid source, the MCU does not choose another endpoint to occupy that quadrant. Instead, the video of the Empty Quadrant Endpoint or a default fill is broadcast until the endpoint's video is unsuppressed and/or the endpoint becomes a valid source. This strategy also applies to endpoints that are designated in fixed quadrants and are not yet joined or are in the process of joining the conference (that is, have not yet reached primary status). A fill pattern is displayed in the quadrant until the endpoint is able to transmit video.

A endpoint loses its fixed status in a quadrant *only if* the endpoint drops or a control interface requests that a change be made to the quadrant.

With UCC, the controller is able to change only one quadrant at a time.Other control interfaces (for example, CRCS Agent and the MCU-MT) can change more than one quadrant at a time.

Mute Controls

The UCC controller is able to affect the incoming audio of endpoints in the conference with various UCC Mute commands. These commands can be used to isolate "noisy" endpoints and essentially create a broadcast conference where only one endpoint in the conference can be heard. UCC Mute options can be used in VAS and Presentation conferences, regardless of the audio mode (G.711, G.728, or G.722) or the bandwidth (2B, 336k, 384k, 768k, etc).

Mute controls are intended for every endpoint in the conference except the UCC endpoint. Accordingly, the Presenter, the full-screen broadcaster, an endpoint broadcasting in a quad image, or an audio-only endpoint can all be muted and unmuted. Any UCC service, such as Rollcall, Browse, or Select Broadcaster, may be used while a mute feature is in effect. The Status Conference form indicates the UCC-muted status of each endpoint.

Video is provided to the conference even if mute options are activated. For example, video broadcasters are chosen first from the speaking (unmuted) endpoints by the VAS mechanism. However, the VAS algorithm arbitrarily selects a video broadcaster from silent or muted endpoints if the MCU needs to select a video source but is unable to select one from the speaking endpoints (because, for example, none of these endpoints are a valid video source).

Mute Endpoint

This option allows the controller to mute the incoming audio of the specified endpoint. Whenever this option is activated on an endpoint, audio from that endpoint is not included into the conference. Accordingly, other endpoints in the conference do not hear that endpoint if the endpoint speaks, and the MCU does not consider that endpoint whenever it is selecting a speaking endpoint for VAS.

This option has no effect on the audio being transmitted to that endpoint; the endpoint can still hear the conferenced audio. This behavior is similar to what happens when the MCU receives the Audio Indicate Mute (AIM) BAS code from that endpoint. (This code indicates that the endpoint has muted.)

The MCU sends the AIM BAS code to all endpoints not receiving any audio due to UCC muting or local endpoint muting. AIM is sent to indicate that an endpoint is hearing no audio. Upon receiving this command, an endpoint can display “far-end mute,” provided the endpoint is equipped to do so.

The Mute Endpoint command is denied if any of the following are true:

- Endpoint number is not in the range 1 through 25
- Specified endpoint is not reserved in the conference
- Specified endpoint is not joined to the conference
- Specified endpoint is the UCC party.

If a UCC-muted endpoint subsequently unmutes its local audio by sending the BAS code Audio Indicate Active (AIA), the MCU remembers that AIA has been sent but does not allow the endpoint's audio to be included into the conference until the UCC Mute option is canceled on that endpoint. This option is canceled via a UCC Unmute command, which is discussed in the next section.

If the controller requests that an endpoint be muted, but the endpoint has already been muted by UCC, positive tone is provided.

Except for in a conference with Solo Audio mode (explained later), a UCC-muted endpoint that disconnects from a conference and later rejoins the conference rejoins the conference as an *unmuted* endpoint.

Whenever the UCC party disconnects from the conference, the MCU automatically unmutes all endpoints that were muted by UCC.

Unmute Endpoint

This option allows the controller to cancel the UCC Mute feature on an endpoint. Therefore, the option allows audio from that endpoint to be included into the conference. However, the option does not override the endpoint's AIM status. If the endpoint has sent AIM, the MCU does not include its audio into the conference until the endpoint sends AIA. In other words, an endpoint is considered unmuted only if it is sending AIA and is not muted by UCC.

As a result of UCC Unmute, the MCU sends the AIA BAS code to other endpoints if AIM was previously sent to all the endpoints. Accordingly, each endpoint can remove the “far end mute” message.

The Unmute Endpoint command is denied if any of the following are true:

- Endpoint number is not in the range 1 through 25
- Specified endpoint is not reserved in the conference
- Specified endpoint is not joined to the conference
- Specified endpoint is the UCC party.

If the controller requests that an endpoint be unmuted but has already been unmuted by UCC, positive tone is provided.

Solo Audio

This option allows the controller to mute all endpoints in the conference except the specified endpoint. Whenever this option is activated, only the audio of the specified “solo” endpoint is included into the conferenced audio. This gives the effect of a “Broadcast” conference.

With Solo Audio optioned, the solo endpoint does not hear the audio of any endpoint in the conference. The behavior of muting endpoints with the Mute Endpoint command (described previously) also applies to the Solo Audio option. For example, the AIM and AIA status of each endpoint is tracked by the MCU.

Once this option is activated, the conference enters Solo Audio mode. All other endpoints that have either already joined the conference or will join it once this option is activated join the conference in a muted state.

The UCC controller may use any of the Mute features after the conference is in Solo Audio. For example, the Unmute option can be used to select some endpoints to be heard. However, unmuting any endpoint after Solo Audio is active turns off the Solo Audio state. The other muted endpoints remain muted in this case, but any endpoints joining the conference thereafter do not join muted.

If the Solo Audio command is used while the conference is already in Solo Audio mode, the MCU reconfigures the audio to satisfy the request.

The Solo Audio command is denied if:

- Endpoint number is not in the range 1 through 25.
- Specified endpoint is not reserved in the conference.
- Specified endpoint is not joined to the conference.
- Specified endpoint is the UCC party.

Solo Audio is also canceled with the Unmute All Endpoints command (discussed in the next section), if the solo endpoint drops, or if the UCC party disconnects from the conference. Once this option is canceled, the audio for all endpoints is included into the conference.

Unmute All Endpoints

This option allows the controller to unmute quickly all endpoints (if any) that have been muted by UCC. The option also cancels the Solo Audio feature. The Unmute All request is never denied.

Adjust Audio Volume for Audio Add-On Party

This option allows you to increase, decrease, or reset the volume of specified endpoints. It can be accessed only via UCC; it is not available via CRCS, conference control command, or administration.

The endpoint number within the command for the option must contain two digits. For endpoint numbers less than 10, a ‘0’ must be placed before the digit (for example, 01, 02, 07, etc.). Also, endpoint number ‘00’ indicates that the specific Adjust Audio Volume request is to be applied to all Audio Add-On parties within the conference.

The commands for this option are denied if the specified endpoint has not joined the conference or is not an Audio Add-On party.

Whenever you enter a valid command for this option, the conference is put into an audio volume control mode. This mode allows you to adjust the volume by merely entering the volume control digit (7, 8, or 9); you need not reenter the associated endpoint number. The MCU ensures that the audio level of the endpoint never exceeds a 3 db gain or a 15 db loss.

For example, entering ****64 02 7** decreases the audio level of endpoint 2 by 3 db. Thereafter, entering **9** increases the endpoint’s audio level by 3 db. Entering **9** again increases the endpoint’s audio level by another 3 db. Thereafter, entering **7** decreases the endpoint’s audio level by 3 db. Finally, entering **8** resets the endpoint’s audio level back to the default (6 db loss).

To exit audio volume control mode, enter *#*. If you enter a valid command followed by *#*, the MCU first processes the comand and then exits audio volume control mode. For example, if you enter ****64 02 7#,** the MCU first decreases the audio level of endpoint 2 by 3 db and then exits audio volume control mode.

You can also exit audio volume control mode by entering the command for any other UCC option (including Select Broadcaster, Rollcall, etc.).

Manipulate Most Recently Joined Endpoint

This option allows you to perform various operations on the endpoint that most recently joined the MCU conference. These operations include broadcasting the endpoint, muting the endpoint, etc.

If the most recent endpoint to join the conference is an invalid video source, entering ***99** (to broadcast the most recently joined endpoint) is denied. Also, if there are no endpoints currently joined to the conference (except for UCC endpoints), all commands for this option are denied.

Interactions

The UCC feature interacts with the following:

Audio Add-On

An Audio Add-On endpoint and a UCC endpoint are allowed to be joined to the same conference. UCC and Audio Add-On parties share the same pool of mcu-extensions. However, UCC and Audio Add-On have separate port capacities. These capacities are administrable on the Customer-Options form. On the Conference Record form, a UCC party is distinguishable from an Audio Add-On party by its endpoint type, which is “UCC.” The UCC controller can drop an Audio Add-On party in a conference by issuing UCC Drop Endpoint or End Conference commands. Also, the UCC party can use any of the Mute commands on the Audio Add-On party.

If the controller attempts to select the Audio Add-On party for a video broadcaster by using the Select Broadcaster or Set Quadrant command, the MCU provides a negative feedback tone and disregards the request.

Application Compliant Flag

A UCC party cannot participate in an MLP data conference. Therefore, there is no interaction with this feature, and the *App Comp* field for the UCC party in the Conference Record is blank.

Basic/Enhanced Service Flag

A UCC party cannot participate in multimedia functions. Therefore, there is no interaction with this feature, and the the *Basic/Enhanced* field for the UCC party in the Conference Record form defaults to **basic**. However, this field may be set to **enhanced** with no effect.

Conference Bandwidth

A UCC endpoint may be added to a conference of any bandwidth that is administered in VAS or Presentation conference mode.

Conference Redial Flag

A dial-out endpoint dropped by the UCC party can be redialed by setting the endpoint's per-user conference *Redial?* flag in the Conference Record to **yes** .

For a UCC party, the *Redial?* field *cannot be set* to **yes** because UCC is restricted to dial-in endpoints only. Therefore, this field will be blank.

CRCS Monitoring

The CRCS real-time monitoring interface recognizes any updates made to the conference via UCC. The CRCS polls the MCU approximately every five seconds for status. This allows the CRCS to update screens to reflect current broadcasters, muted or unmuted status, etc.

Cascading

UCC can be used in cascaded conferences. Required entries to the Conference Record include **UCC** in the *Type* field and an appropriate nonblank value in the *Cascade Mode* field.

A UCC party located on either the primary MCU or secondary MCU may select the inter-MCU link as a video broadcaster. However, in cascades with Continuous Presence *Plus* optioned, the secondary MCU is administered in Presentation mode and the inter-MCU link is administered as the Presenter. Therefore, the same UCC restrictions that apply to a noncascaded Presentation conference also apply to a cascaded Presentation conference (that is, Select Broadcaster and Rollcall are not allowed).

Chair Control

UCC cannot be used in a Chair-controlled conference. Therefore, "UCC" endpoint types cannot be administered in "Chair" conferences. Moreover, an H.320 endpoint is not granted a request for the Chair token in a VAS or Presentation administered conference. If an endpoint requests the Chair token by sending the Chair Command Acquire (CCA) BAS code, the MCU denies the request by sending the Chair Command Refuse (CCR) BAS code back to that endpoint.

Continuous Presence *Plus*

UCC can control conferences optioned with Continuous Presence *Plus*. Interactions between UCC and Continuous Presence *Plus* are briefly discussed in the following list:

- UCC can change the composition of the quad image by placing a specific endpoint into a quadrant, or by designating a quadrant as a VAS quadrant with the Set Quadrant command.

- UCC can transition the image in a Continuous Presence *Plus* conference between full screen, panoramic, and quad screen with a set of commands.
- Full-screen options, such as Select Broadcaster, Browse, and Rollcall, can be used in Continuous Presence *Plus* conferences. However, these options cause an immediate transition to full screen if the conference is currently viewing quad screen.
- The Browse option can be used in Continuous Presence *Plus* conferences. However, the conference can be viewing full screen, or it can be in presentation quad mode. The MCU does not automatically transition the conference to full screen when this command is requested.
- The UCC Unlock Video command can be used in Continuous Presence *Plus* conferences. If this command is requested when the conference is in full-screen mode, the command cancels all video locks created by the UCC full-screen options (such as Select Broadcaster, Browse, or Rollcall). If the conference is in quad-screen mode, the command returns the quad composition back to the administered defaults.
- UCC Mute Controls can be used on endpoints during a Continuous Presence *Plus* conference. This option can be used on any endpoint, regardless of whether the video for the endpoint is currently being displayed in a quadrant. If an endpoint is muted in a VAS-designated quadrant, the VAS algorithm attempts to replace it with a speaking endpoint.
- The UCC Drop Endpoint and End Conference options do not have any specific interactions with Continuous Presence *Plus*. Endpoints are dropped.
- Whenever the UCC party disconnects from a Continuous Presence *Plus* conference, video locks imposed on the full-screen image are removed. However, video locks created for the quad-screen image by UCC or other control interfaces are not cleared.

Dedicated Access

UCC endpoints cannot access the MCU via dedicated DS1 facilities. Dedicated Access endpoints have a unique type on the Conference Record that distinguish them from UCC endpoint types.

Direct Connect

A UCC endpoint cannot be a locally attached BRI/DCP endpoint. If a locally attached voice endpoint dials a UCC mcu-extension, it is not allowed into the conference.

Echo Cancellation for Audio Add-On

The UCC party cannot cause other parties in the conference to hear echo because it is not providing (or receiving) audio to the audio conference.

Executive Conferencing

Many of the UCC options can be used for Executive Conferencing. However, the only UCC options causing a screen mode change that can be used for the feature are those that change Executive Conferencing mode to full-screen mode, and vice versa. Therefore, options involving dual-panoramic, upper-panoramic, lower-panoramic, and quad-screen are rejected. For UCC *Star*, reject notification is provided via feedback tones.

UCC commands can be used to assign conference sites within an Executive Conferencing conference to a particular sub-picture. Also, UCC control can be used to establish an Executive Conferencing endpoint as a broadcaster endpoint. This includes such endpoints that are moved from one conference to another.

H.CTX and H.CTX+

UCC control is allowed in conferences configured with these proprietary videos from CLI. However, since Continuous Presence *Plus* cannot be optioned in H.CTX conferences, use of the Continuous Presence *Plus* commands specific to UCC are denied in such conferences.

SG4

UCC control is allowed in conferences configured with PictureTel's SG4 video. However, since Continuous Presence *Plus* cannot be optioned in SG4 conferences, use of the Continuous Presence *Plus* commands specific to UCC are denied in such conferences.

International MCUs

UCC can be used on internationally-configured MCUs. The TN744C circuit pack detects touchtones generated by switching systems located internationally. The interdigit pause time can be administered on the System-Parameters Country-Options form.

Positive and negative feedback tones generated by the TN780 circuit pack are consistent for the country administered. If the default feedback tones chosen (for example, "Call Waiting" and "Hold Recall") are inappropriate for the country, custom tones may be administered by changing the System-Parameters Country-Options form.

Low-/High-Speed Interworking

Like an Audio Add-On party, a UCC party may join a conference of any bandwidth. Therefore, a UCC party arriving on 56k or 64k facilities may join higher bandwidth (for example, 112k/128k 2B, H0, 336k/384k BONDed, 768k, etc.) conferences. The UCC party does not cause any changes to the selected communications mode (SCM) of the conference.

Notification Signaling

Video source BAS indications (VIN and VIN2) are sent by the MCU to an endpoint. These indications serve to identify to the endpoint the video source that the endpoint is viewing. The indications are also sent whenever the UCC selects a video source.

However, video source indications are not sent to endpoints in conferences with Continuous Presence *Plus* optioned. Therefore, VIN is not sent as a result of UCC selecting endpoints for full-screen or quad-screen.

Password Protection

UCC provides an optional Digital Tone Multifrequency (DTMF) password security capability. Also, the UCC party uses the same password as Audio Add-On parties. Whenever the *Password Scope* field in the Conference Record form is administered as **audio** or **all**, the UCC party is required to enter a password before being allowed to join the conference.

Presentation Mode

A UCC party is allowed in a conference administered with Presentation mode. The following UCC options are allowed in Presentation mode:

- Browse
- Unlock Video
- Mute, Unmute, Solo Audio, Unmute All
- Force to Quad, Set Quad (only if administered for Continuous Presence *Plus*)
- Force to Full
- Drop Endpoint
- End Conference

The Select Broadcaster and Rollcall options are denied in Presentation mode.

See-Me (MCV)

See-Me is a full-screen feature used by H.320 endpoints to request that the MCU broadcast its video to all other participants in the conference. Endpoints make this request by sending to the MCU the MultiPoint Command Visualize (MCV) BAS code. A request from the UCC controller to select the full-screen video broadcaster overrides any endpoint's request for MCV. However, all MCV requests are still remembered by the MCU, even if they are overridden by the UCC party.

The following list summarizes the interactions between UCC and MCV:

- If the UCC party requests that the conference return to the VAS method of video switching (via Unlock Video), and if an MCV broadcaster is in effect, the MCU disregards the UCC request and provides a negative feedback tone. The current MCV broadcaster retains its MCV status.
- If the UCC party requests a change in broadcaster, and if an MCV broadcaster is in effect, VCR is sent to the current MCV broadcaster, and the endpoint selected by the UCC party becomes the new broadcaster (if it is a valid video source).

  If the endpoint selected by the UCC party is currently broadcasting due to MCV, the endpoint loses its MCV status, VCR is sent toward it, and it continues to broadcast.

  In both cases, the MCU remembers that the endpoint had sent MCV until cancel-MCV is received. If an Undo Video Locks command is received at this point, an MCV requestor (if any) is selected as broadcaster.
- If the UCC party disconnects from the conference while an MCV broadcaster is in effect, the broadcaster is unaffected, and it retains its MCV status.
- If the conference is changing back to VAS via the Unlock Video command, the MCU first chooses a video broadcaster from those endpoints that have requested MCV. If no endpoints are currently requesting MCV, the VAS algorithm chooses a broadcaster from the speaking endpoints. If no endpoints are speaking, the current broadcaster remains (if it is a valid video source).
- If a conference transitions from full screen to quad screen, and if an MCV broadcaster is in effect, VCR is sent to the MCV broadcaster, and the image transitions to quad screen. The MCU remembers that the endpoint has sent MCV until a cancel-MCV is received.

Terminal Names and Numbers

The UCC party is not considered part of the multimedia conference. Therefore, the MCU does not assign a terminal number to it (via TIA), and other terminals are not notified of its presence (via TIN or TIL). Also, the MCU does not query the UCC party for a terminal name (via TCS-2) or provide a name to other terminals (via TIP).

TN744 Resource Failure

If a port on the TN744 Call Classifier fails while actively connected to a UCC party, the MCU disconnects the UCC party. To reenter the conference, the UCC party must dial back into the MCU. Status conference reports that a resource failure caused the drop.

Video Switching Mode and Broadcaster Notification

A proprietary nonstandard command (ns-cmd) is used to announce a change in the current video switching mode and/or broadcaster to all endpoints. A unique value for the switching mode is used to indicate that the broadcaster has been designated by the UCC controller. An ns-cmd is sent to all endpoints whenever any of the following occurs:

- UCC party designates a new broadcaster with Select Broadcaster or Rollcall. (A new broadcaster is sent in the command and mode changes to UCC.)
- UCC party cancels all video locks with Unlock Video. (The conference mode is changed from UCC to VAS or Presentation.)
- UCC party disconnects from the conference, and a fixed full-screen UCC broadcaster is in effect. (The conference mode is changed from UCC to VAS.)
- Browse is canceled while in Presentation mode. (The conference mode is changed from UCC to Presentation.)
- UCC party drops a broadcaster. (A new broadcaster is sent in the command.)
- UCC broadcaster becomes a non-video source. (A new broadcaster is sent in the command, and the mode changes from UCC to VAS.)
- Broadcaster and/or mode notification are not supported in Continuous Presence *Plus* conferences. Therefore, there is no command sent when endpoints or VAS are selected for quadrants or when transitioning between full and quad-screen.
- If the MCU disregards a UCC request to change the broadcaster or clear video locks, the MCU does not honor the request, and no ns-cmd is sent. A negative feedback tone is provided to indicate that no action was taken.

Resource Availability—24 Hour Scheduler/CRCS

The resource availability checks performed by the conference scheduler (both on-line 24-hour and CRCS) ensure that the number of UCC controlling parties active in the MCU at the same time does not exceed the system capacity administered on the System-Parameters Customer-Options form. This port capacity check is performed when the conference form is submitted via either the **add conference** or **change conference** command.

The port capacity calculation reflects that a single UCC port type consumes no VC or MMI resources. The UCC port uses one TDM timeslot for transmitting touchtones to the MCU.

The advertised maximum port capacities assume that UCC and Audio Add-On parties are not administered in any conference. Therefore, including a UCC party

and/or an Audio Add-On party in any conference may decrease the maximum port capacity for any random mix of rates

Notification Tones

The UCC party is not a part of the multimedia conference. Therefore, an entry tone is not played to endpoints in the conference just before the UCC party joins the conference. Also, an exit tone is not played to conference participants after the UCC party disconnects.

The UCC Mute feature does not affect entry, exit, and warning tones. Even if an endpoint is muted, it can still hear these tones. Also, an exit tone is still played if the endpoint drops.

Dynamic Conference Resizing

Dynamic Conference Resizing allows a UCC party to be added to or removed from a conference once that conference becomes active. This process is subject to the UCC administered port capacity for the time period of the conference and to the one UCC party per conference limit. If a UCC party is removed by administration from the conference, it is dropped from the conference, and subsequent incoming calls to the deleted mcu-extension are rejected and given intercept tone if this mcu-extension is unused in another conference. If added to a conference in progress, the UCC party can immediately dial its mcu-extension number and conference password (if applicable) and be connected to the conference.

UCC parties can be added to a conference via this feature by the CRCS reservations agent or by submitting the appropriate Conference Record from the MCU-MT.

Impact on Endpoint Numbering

The addition and/or removal of endpoints other than the UCC party can cause endpoint numbering to change. For example, an endpoint might be removed from the conference but then added back to the conference via a location number that is different from the one established for the endpoint on the Conference Record. In such a case, the mcu-extension for the endpoint is also different from the one originally established at conference reservation. Moreover, endpoints removed or added are not reflected on the original confirmation report provided by CRCS. Whenever the addition or removal of participants occur, CRCS guarantees that endpoint numbering between the MCU and the CRCS remains consistent. Consistent numbering is also maintained when information about a conferee is changed (dial-out number, type, etc).

Since these numbers are essential to the endpoint specific features of UCC, it is recommended that resizing changes that result in changes to endpoint numbering are prevented in UCC conferences. However, if conference resizing is performed, a new CRCS confirmation report should be printed, faxed, or electronically mailed to reflect the most current endpoint numbering.

UCC Control Overrides

Changes made by UCC can be overridden via dynamic changes to the Conference Record. For example:

- Changing the fields that control the picture composition for Continuous Presence *Plus* overrides any composition made via the UCC Set Quadrant command.
- Changing the field that allows the conference to be optioned with Continuous Presence *Plus* can be turned off, causing a transition to full screen. This overrides a previous UCC request to transition to quad screen.

The UCC party does not receive any feedback when a previous request is cancelled or overridden.

Dial-Out

UCC-controlling parties can be Dial-Out parties.

Call Detail Recording

The MCU provides Call Detail Recording (CDR) output for incoming and outgoing trunk calls. Calls occurring on a UCC mcu-extension are included into the CDR record in a manner similar to calls occurring on an Audio Add-On mcu-extension.

Administration

The UCC feature must be administered on the System Parameters Customer Options form. For each conference that allows UCC, the Conference Mode must be set appropriately on the conference form. Also, the conference must be enabled with Voice-Activated Switching (VAS). Finally, the feature is implemented in 4-port increments.

Hardware and Software Requirements

This feature can be upgraded in increments of eight UCC endpoints to a maximum of 24 and is available only with the Release 3.0 or later version of the MCU. One TN744C Call Classifier circuit pack for every eight UCC ports is required for this feature.

Uniform Dial Plan

Description

The Uniform Dial Plan (UDP) provides a common 4- or 5-digit dial plan (specified by Dialplan administration) that can be shared among a group of MCUs. UDP maps an incoming 7-digit number into a local MCU number.

Interactions

The following features interact with the UDP feature.

- AAR

  After the system determines the RNX of the switch being called, AAR routes the call to the correct switch. The required subset of AAR is provided with the UDP software. If the AAR feature is provided in addition to the UDP, the seven-digit AAR number will provide the exact same routing as the UDP.

- Dial Plan

  All of the extension numbers on a switch are not necessarily part of the UDP. Any that do not belong to the UDP are handled by a regular, non-UDP dial plan associated with the local switch. When administering the dial plan and designating a group of extensions as UDP non-local, you can specify on the Dialplan form whether you want to search for local extensions first or last. This allows some flexibility in the changing of extensions from local to non-local. However, after the dial plan is changed to make an extension UDP, nothing can be administered with these extensions on the local switch.

Administration

The UDP is administered by the system administrator. The following items require administration:

- Whether UDP has 4- or 5-digit extension numbers PBX Codes (expands first one, two, three, or four digits of dialed extension to an RNX)
- AAR Analysis Table (used by AAR to route calls to the correct switch)
- Routing Patterns
- Node Number Routing (used to route ENP calls)
- Extension Codes and type(s) of conversion
- AAR Digit Conversion (to define conversions for AAR Code extensions or to define home location codes)

NOTE:
If the user changes the Uniform Dialing Plan field value from a ‘‘y’’ to an ‘‘n,’’ then a warning message is generated to inform them that this action causes all UDP extension codes to be lost. The same warning message is applied if the Plan Length field value is changed from a ‘‘4’’ to a ‘‘5’’ or from a ‘‘5’’ to a ‘‘4.’’

Hardware and Software Requirements

UDP software is required.

Video Conference Control

Description

Video conference control is performed using one or more of the following methods:

- Voice-activated switching control (sometimes referred to as automatic control)
- Chair control (part of user control)
- Presentation (part of advanced control)
- Broadcast with auto-scan (part of advanced control)

A detailed description for each is provided in the following sections.

NOTE:
Several features interact with video conference control methods. In particular, refer to "Continuous Presence Plus" on page B-71 and "Executive Conferencing" on page B-100.

Voice-Activated Switching

A speaker’s endpoint becomes the broadcasting endpoint. The broadcasting endpoint sees the last broadcaster. MCU software and hardware provides for suppression of a ‘‘ping-pong’’ effect to avoid excessive switching during a rapid conversation. As an example, imagine a four-party video conference between endpoints named Site A, Site B, Site C, and Site D. Site B is the first endpoint to speak. When Site B finishes speaking, Site A begins to speak. As Site A begins, the voice-activated switching function displays Site A to the other three endpoints, while Site B is still displayed on Site A’s screen as depicted in the following figure.

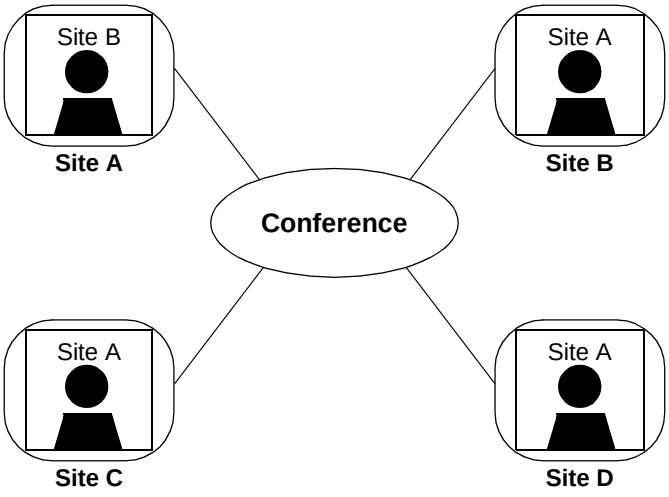

Figure B-11. Previous Speaker=Site B

The following figure illustrates the scenario when the speaker at Site A relinquishes to the speaker at Site B. In this setting, Site B sees Site A (the previous speaker), while all the other endpoints see Site B (the current speaker).

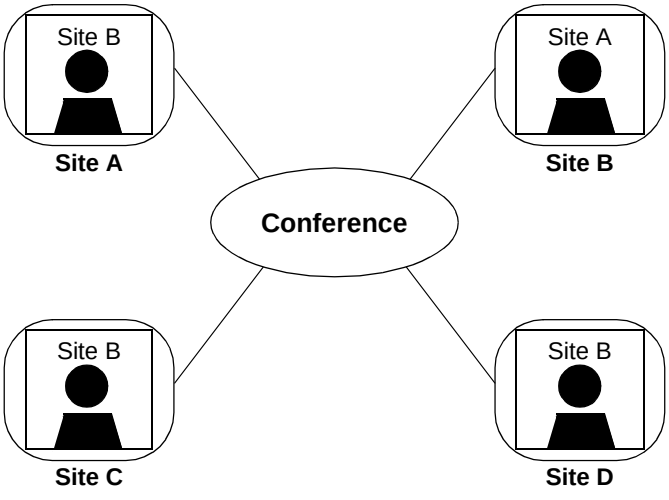

Figure B-12. Previous Speaker=Site A

In the final voice activated switching example, Site D begins to speak after Site B finishes, as shown in the following figure. Site D sees Site B on their screen while all other endpoints see Site D.

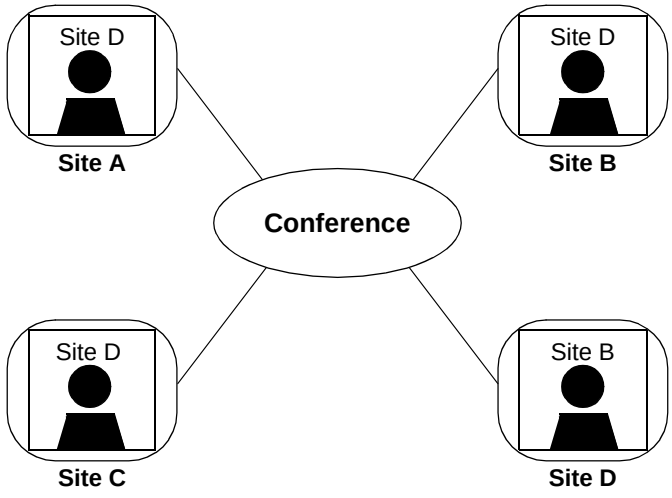

Figure B-13. Previous Speaker=Site B

Contributor

This video function can be used for tasks such as sending high-resolution video still images to other endpoints. It allows a user to broadcast graphics to all the other conferees. To do so, the broadcasting endpoint uses a code to gain the broadcast control. After broadcasting, the endpoint sends a second code to relinquish control. This is sometimes called the ‘‘See Me’’ function.

Chair Control

This video function allows a conference chair to select which endpoint video is being broadcast to all the other endpoints. An endpoint with chair control can request and relinquish control and drop conferences. The endpoint that claims ‘‘chair control’’ is the chair endpoint. In the following figure, Site A has the chair token and has selected Site C as the broadcast source while Site D is speaking. In this example, Site C (the broadcaster) is displayed at all endpoints except Site C. Site C sees Site D (the speaker).

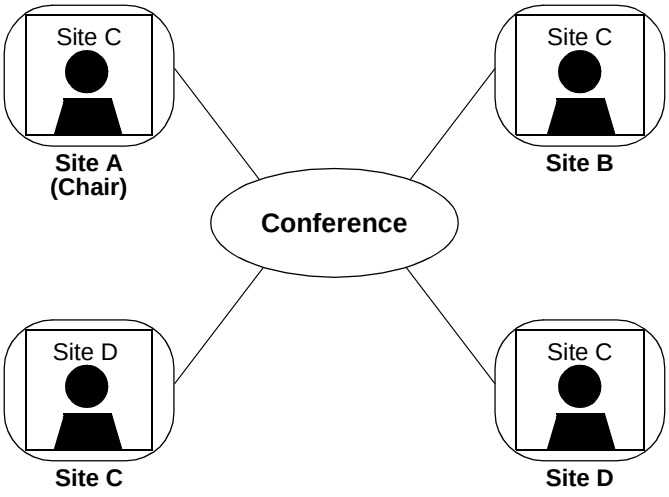

Figure B-14. Broadcaster=Site C, Speaker=Site D

After a time, Site D finishes talking and Site C resumes talking. No change occurs to the video streams as shown in the following figure.

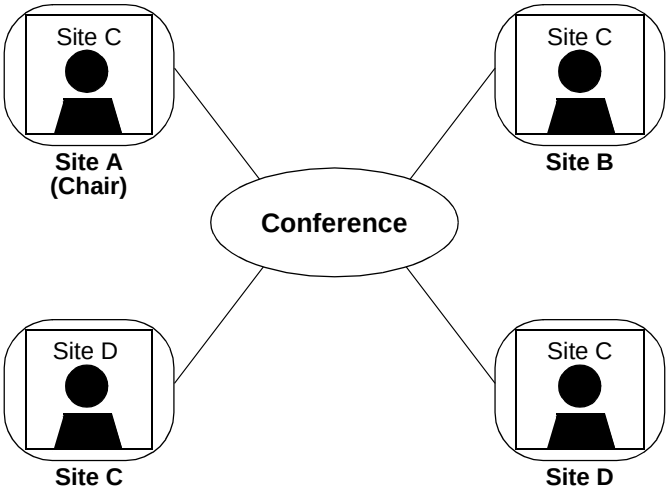

Figure B-15. Broadcaster=Site C, Speaker=Site C

Site A begins to talk when Site C asks for responses. Now, as shown in the following figure, Site C sees Site A (the speaker) on their screen while all other endpoints continue to see Site C (the broadcaster).

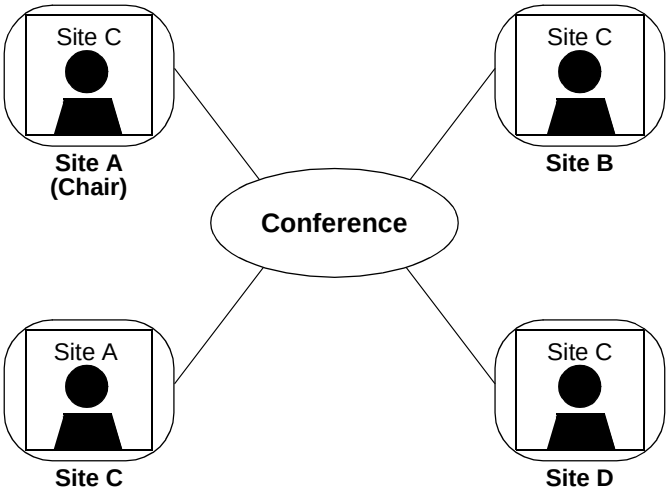

Figure B-16. Broadcaster=Site C, Speaker=Site A

Presentation

This function does not require special endpoint support. It is a standard feature on all MCU models. This function configures the system to broadcast a single endpoint (*presenter*) to all other endpoints. The receiving endpoints are set to *voice-activated switching* so the broadcaster sees the endpoint selected by that function while all other endpoints see the broadcaster. One endpoint is administered as the broadcaster on the MCU prior to the conference. In the following figure, Site A is administered as the broadcaster while Site D is speaking. In this case, Site A (the broadcaster) sees Site D (the speaker) while all other endpoints see Site A.

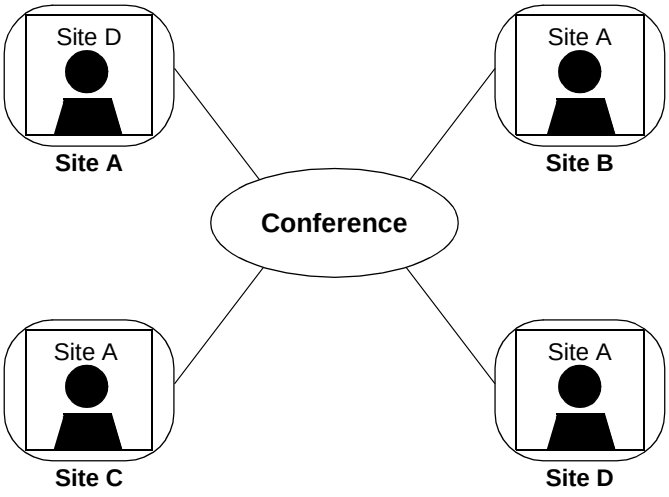

Figure B-17. Speaker=Site D

The following figure illustrates what happens when Site D stops speaking and Site C begins. Site A remains the broadcaster and sees Site C (the speaker) while all other endpoints still see Site A.

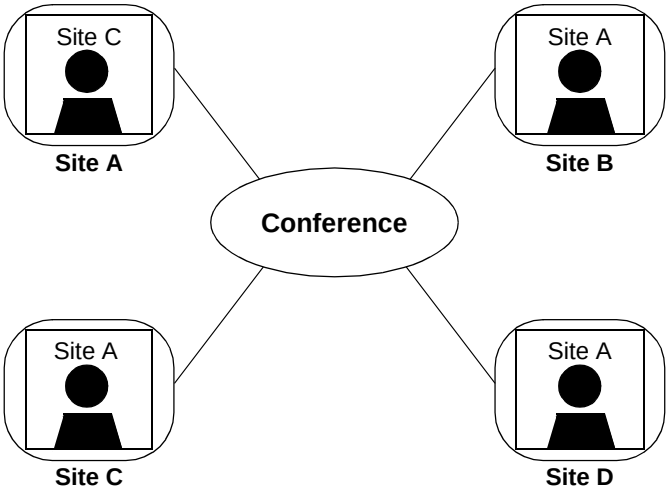

Figure B-18. Speaker=Site C

Broadcast with Autoscan

This is a function that does not require special endpoint support. Broadcast with auto-scan provides for a single endpoint to broadcast audio and video to all other endpoints. The broadcasting endpoint does not receive audio but sees video from each of the endpoints one at a time. The interval each endpoint is displayed is administrable on the MCU. In the following figure, Site A was identified as the broadcast endpoint when the conference was scheduled. The scanning period was also set during conference reservation. Site A views each endpoint in turn for the scanning period one at a time until the conference ends. All other endpoints see and hear Site A the entire time.

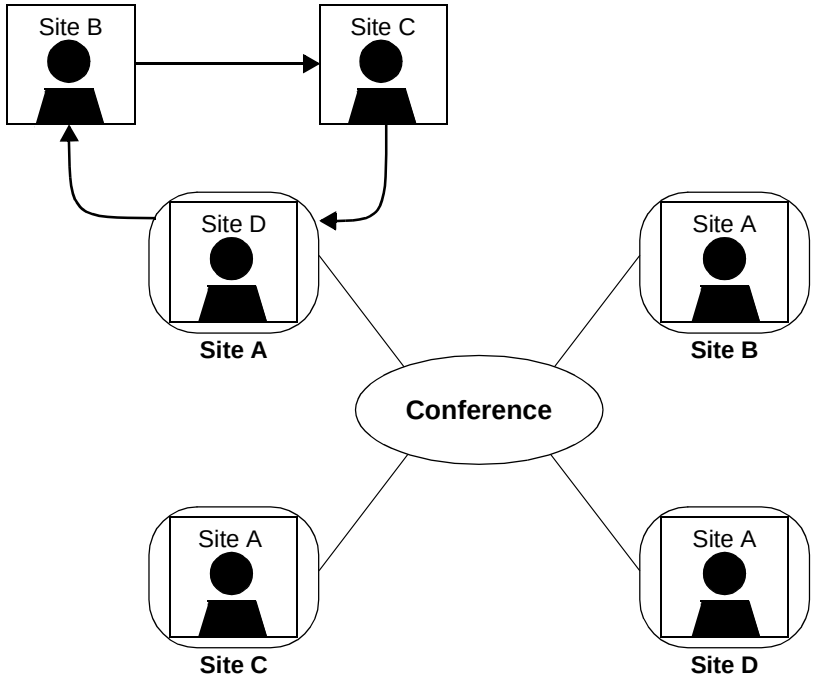

Figure B-19. Broadcast with Autoscan

Considerations

As described in the detailed descriptions of video conference methods.

Interactions

None.

Administration

None.

Hardware and Software Requirements

Consult your Lucent Technologies service representative.

Video Modes

Description

The Lucent Technologies MCU supports the following standard video modes:

- H.261 and its variations, including H.261-down_only, H.261-w/upgrade, H.261-CIF-Floor, H.261-CIF-Now, and H.261-QCIF-Now
- H.263 and its variations (which involve H.263/H.261 transcoding), including H.263/1-down_only and H.263/1-w/upgrade

The MCU also supports the following proprietary video modes:

- H.CTX/H.CTX+
- SG4
- Multirate Video (MRV) and variation MRV-w/upgrade

Considerations

The MCU also supports H.263/H.261 transcoding.

Some MCU conferences may not accommodate all video modes due to feature interactions.

Interactions

Refer to the appropriate sections concerning video modes in this appendix for a complete discussion of the feature interactions for the following video modes: H.263 video mode, H.263/H.261 transcoding, H.CTX/H.CTX+, MultiRate Video (MRV), and SG4.

In addition, be sure to refer to the “Interactions” sections in this appendix for the following features, for which there is a discussion of one or more video mode interactions: Continuous Presence *Plus*, On-Screen Display, Speed Match, T.120 Data Conferencing, and Universal Conference Control.

Administration

Video modes are administered via the *Video Mode* field in the Conference Record form, which is discussed in Chapter 1 of this document. Refer to the “Video Mode” bullet item within the “Conference Record Form” section for details on administering video modes.

To administer H.263/H.261 transcoding, refer to the “Administering H.263/H.261 Transcoding” section in Chapter 2 of this document.

Hardware and Software Requirements

Consult your Lucent Technologies representative.

Administration Commands

C

Overview

This appendix provides a summary of the available MCU administration commands and their associated objects. You can display a list of the available actions, objects and qualifiers at any time by pressing the Help key. The list that appears contains only those words that are appropriate based on your login permissions and the words already entered on the command line.

Usage Restrictions

You cannot execute an administration command if any of the following are true:

- You do not have permission to execute the command.
- System configuration does not require a command.
- Command is a member of a set of commands that is being used by another user.

Add

The **add** command allows you to add objects to the system. The object that you add must not already exist in the system. For example, if you enter the **add conference *124*** command, there must not already be a “conference 124” in the system. If the name you selected already exists, an error message appears, and you must rename the object.

The following objects are available for use with the **add** command. You can display this list by pressing the Help key after you enter the **add** action word.

- ***access-endpoint***
- ***administered-connection***
- ***bri-trunk-board***
- ***conference***
- ***data-module***
- ***ds1***
- ***hunt-group***
- ***intra-switch-cdr***
- ***login***
- ***modem-pool***
- ***mcu-extension***
- ***second-digit***
- ***signaling-group***
- ***station***
- ***trunk-group***

Change

The **change** command allows you to change an object that is currently administered in the system. You can change the following objects.

- ***aar***
- ***access-endpoint***
- ***administered-connection***
- ***alphanumeric-dial-table***
- ***ars***
- ***bri-trunk-board***
- ***bulletin-board***
- ***circuit-packs***
- ***communication-interface***
- ***conference***
- ***cor***
- ***cos***
- ***data-module***

- ***dialplan***
- ***ds1***
- ***feature-access-codes***
- ***hunt-group***
- ***isdn***
- ***lxc-codes***
- ***login***
- ***mcu-extension***
- ***meas-selection***
- ***modem-pool***
- ***mst***
- ***password***
- ***permissions***
- ***report-scheduler***
- ***rhnpa***
- ***route-pattern***
- ***second-digit***
- ***signaling-group***
- ***station***
- ***synchronization***
- ***system-parameters***
- ***toll***
- ***trunk-group***
- ***udp***

Clear

The **clear** command allows you to clear measurements for security violations and to turn off message waiting lamps. You can clear the following objects.

- ***audits***
- ***errors***
- ***hardware-group***
- ***interface***
- ***isdnpri-testcall***
- ***link***
- ***measurements***
- ***mst***
- ***pkt***
- ***port***

Display

The **display** command allows you to view parameters that are associated with a specific object/qualifier. For example, if you enter the **display station *4444*** command, the voice terminal screen form currently administered for extension 4444 is displayed. If you use the ***print*** qualifier (for example, **display station 4444 *print***), the form is printed on the printer associated with the administration terminal.

The following object words are available for use with the **display** command.

- ***aar***
- ***access-endpoint***
- ***administered-connection***
- ***alarms***
- ***alphanumeric-dial-table***
- ***ars***
- ***available-ports***
- ***bri-trunk-board***
- ***bulletin-board***
- ***button-location-aca***
- ***capacity***
- ***circuit-packs***

- ***communication-interface***
- ***conference***
- ***cor***
- ***cos***
- ***data-module***
- ***dialplan***
- ***digit-absorption***
- ***disabled-tests***
- ***ds1***
- ***errors***
- ***feature-access-codes***
- ***hunt-group***
- ***initcauses***
- ***internal-data***
- ***isdn***
- ***ixc-codes***
- ***login***
- ***mcu-extension***
- ***meas-selection***
- ***memory-configuration***
- ***mst***
- ***modem-pool***
- ***permissions***
- ***port***
- ***rhnpa***
- ***route-pattern***
- ***second-digit***
- ***signaling-group***
- ***software station***
- ***synchronization***
- ***system-parameters***
- ***time***
- ***toll***
- ***trunk-group***
- ***udp***

List

The **list** command allows you to display parameters that are associated with groups of objects. Use this command to obtain a list of either all or a range of voice terminals, trunks, and other features that have been administered. You can use the **list** command and its associated qualifiers to retrieve data for either all the ports in the system or a specified subset of ports. The following objects are available for use with the **list** command.

- ***aar***
- ***aca-parameters***
- ***access-endpoint***
- ***administered-connection***
- ***ars***
- ***asg-history***
- ***conference***
- ***configuration***
- ***cor***
- ***data-module***
- ***disabled-MOs***
- ***extension-type***
- ***history***
- ***hunt-group***
- ***intra-switch-cdr***
- ***isdnpri-testcall***
- ***logins***
- ***marked-ports***
- ***mcu-extension***
- ***measurements***
- ***members***
- ***mmi***
- ***modem-pool***
- ***mst***
- ***performance***
- ***report-scheduler***
- ***route-pattern***

- ***signaling-group***
- ***station***
- ***suspend-alm-org***
- ***toll***
- ***trunk-group***
- ***udp***
- ***video-processors***

Remove

The **remove** command allows you to display the form for the associated object and then delete the object from the system. For example, if you type the **remove station *8000*** command, the form for station 8000 appears. If you want to remove this station, press ENTER to delete it from the system. Press CANCEL to leave the station as it is. The following objects are available for use with the **remove** command.

- ***access-endpoint***
- ***administered-connection***
- ***bri-trunk-board***
- ***conference***
- ***data-module***
- ***ds1***
- ***hunt-group***
- ***login***
- ***mcu-extension***
- ***modem-pool***
- ***report-scheduler***
- ***second-digit***
- ***signaling-group***
- ***station***
- ***trunk-group***

Status

The **status** command displays the operational status of the facility indicated by the object word. The following object words are available for use with the **status** command. You can display this list by pressing the Help key after you enter the **status** action word.

- ***access-endpoint***
- ***administered-connection***
- ***all-conference***
- ***audits***
- ***bri-port***
- ***card-mem***
- ***cdr-link***
- ***cleared-alarm-notif***
- ***conference***
- ***data-module***
- ***esm***
- ***hardware-group***
- ***health***
- ***interface***
- ***isdnpri-testcall***
- ***link***
- ***logins***
- ***mst***
- ***packet-control***
- ***periodic-scheduled***
- ***processor-channels***
- ***signaling-group***
- ***sp-link***
- ***station***
- ***synchronization***
- ***system***
- ***trunk***

Glossary

A

AAR
See *Automatic Alternate Routing*.

access endpoint
Endpoint connected to the MCU via non-signaled T1/E1, V.35, RS449, or EIA-530 connections. The ports designated for the endpoint are dedicated to the endpoint.

Access Security Gateway
A challenge/response authentication engine that is intended to provide some remote access security for the MCU.

ACD
See *Automatic Call Distribution*.

active
A reserved conference status when the conference is scheduled to begin but no conferees have joined. A dedicated conference is active at all times.

Adjust Audio Volume for Audio Add-On Party
UCC option that allows the user to increase, decrease, or reset the volume of specified endpoints within an MCU conference. The commands for this option are denied if the specified endpoint has not joined the conference or is not an Audio Add-On party.

Advanced mode
A feature of the MCU that offers Presentation and Broadcast with Auto-Scan conference modes.

AE
See *access endpoint*.

American National Standards Institute
Professional and technical association in the US that supports a variety of standards.

ANI
See *Automatic Number Identification*.

ANSI
See *American National Standards Institute*.

Application Compliant flag
A per-user flag in the Conference Record that indicates which endpoints use non-compliant applications.

ARS
See *Automatic Route Selection*.

ASG
See *Access Security Gateway*.

Asynchronous Transfer Mode
A high-speed, connection-oriented switching and multiplexing technology that uses 53-byte cells (including a 5-byte header and a 48-byte payload) to transmit different types of traffic simultaneously, including voice, video, and data. The mode is asynchronous because information streams can be sent independently without a common clock.

ATM
See *Asynchronous Transfer Mode*.

Audio Add-On
Feature that allows the MCU to support up to six non-H.320 audio-only endpoints per non-cascaded conference and up to 12 such endpoints per cascaded conference.

Audio Add-On Echo Control
An echo canceler function implemented in the MCU to prevent echoes caused by Audio Add-On conferees who dial into a conference through an analog circuit. Echoes occur at the two- to four-wire conversion (hybrid) in the network or in PBXs and are caused by impedance mismatches at those points.

audio port
The port used for an Audio Add-On party to join a conference. See also *port*.

Automatic Alternate Routing
Feature that sends calls to a route other than the route of first choice when facilities are unavailable.

Automatic Call Distribution
Feature that answers calls and then, depending on administered instructions, delivers messages appropriate for the caller and routes the call to an agent when one becomes available.

Automatic mode
Feature of the MCU that offers voice-activated conference mode.

Automatic Number Identification
Representation of the calling number for display or for further use to access information about the caller.

Automatic Route Selection
Feature that allows the system to choose automatically the least cost-effective way to send a toll call.

B

B-channel
A bearer channel provided by the network from a video endpoint to the MCU.

bandwidth
Specifies a transfer rate or range of frequencies for an amount of data moving from one point to another. Used in context for video bandwidth rates, the more free bandwidth for video, the better the picture quality transmitted. The MCU port capacity provides for 112k/128k bandwidth (2B-channel 56k/64k), 1-channel multirate ISDN bandwidths of 1x56k, 1x64k, 112k, 128k, 168k, 192k, 224k, 256k, 280k, 320k, 336k, 384k, 448k, 512k, 672k, 768k, 1008k, 1288k, 1344k, 1472k, 1536k, 1680k, and 1920k, or BONDing bandwidths of 112/128k, 168/192k, 224/256k, 280/320k, 336/384k, 448/512k, and 672/768k.

Bandwidth on Demand
Feature that allows the Lucent Technologies MCU to establish conferences at bandwidths greater than 128k without the use of ISDN-PRI and Wideband H0 channels.

Basic Rate Interface
A standard ISDN frame format that specifies the protocol used between two or more communications systems. BRI runs at 192 Mbps and provides two 64k B-channels (voice and data) and one 16k D-channel (signaling). The D-channel connects, monitors, and disconnects all calls. It also can carry low-speed packet data at 9.6k.

Basic/Enhanced Service Flag
A flag in the Conference Record that disables commands/caps that are known to cause problems (such as crashing) to specific endpoints.

BONDing
See *Bandwidth on Demand*.

BONDing ports
The ports in an MCU conference that use one of the following BONDed bandwidths: 112/128k, 168/192k, 224/256k, 280/320k, 336/384k, 448/512k, or 672/768k.

BRI
See *Basic Rate Interface*.

BRI/DCP Direct Connect Interface
Feature that allows the user to connect BRI or DCP endpoints directly to the MCU without involving a public or private network, PBX, or MUX. The feature allows up to 12 BRI stations or up to four DCP group systems to connect directly to the MCU.

Broadcast w/Auto Scan
A conference mode where one video endpoint's video, audio and data are broadcast to all other conferees. The broadcaster views the other locations one after another for a fixed duration of time. The broadcaster is constantly viewed by all other locations.

broadcaster
A video endpoint is considered the broadcaster when it sends the same video, audio and data signal to two or more locations.

Browse
A Universal Conference Control option that allows the broadcaster to scan endpoints in the conference without specifying an endpoint number for each scanned endpoint. With this option, only the broadcaster sees a change in video caused by the scan.

C

Call Detail Recording
Feature that uses software and hardware to record call data.

cascading
Connecting two or more Lucent Technologies MCUs via an H.320 communications link. This feature allows endpoints connected on different MCUs to participate in the same conference.

CDR
See *Call Detail Recording*.

Chair control mode
A conference mode where one site controls which endpoints are added and/or dropped during a conference.

CIF
See *Common Intermediate Format*.

class
A Conference Record has one of three classes: reserved, dedicated, or file.

Class of Restriction
Feature that allows a maximum of 96 classes of call origination and call termination restrictions for voice terminals, voice-terminal groups, data modules, and trunk groups.

Class of Service
Feature that uses a number to specify if voice-terminal users can activate any of the following features: Automatic Callback, Call Forwarding All Calls, Data Privacy, or Priority Calling.

ClearPresence @716 Audio Coder
Algorithm developed by Lucent Technologies that provides 7 kHz of audio bandwidth at a bit rate of 16k. This feature provides a more natural sounding voice quality than other audio modes, such as G.728 and G.711, which provide 3.5 kHz.

Common Intermediate Format
An industry-standard means for encoding video signals for high-quality pictures. QCIF MPI specifies picture interval numbers. Also known as Full CIF (FCIF).

complete
A reserved conference has a status of complete when the stop time has passed and the last conferee has dropped from the conference.

conferee
A video endpoint participating in a multipoint video conference.

Conference Appointment Form
A form that is used to record all the information necessary to schedule and reserve a multipoint video conference.

Conference Redial flag
A flag in the Conference Record that allows an endpoint to be “redialed” during a conference (therefore, this flag is in effect for Dial-Out calls).

Continuous Presence *Plus*
A customer feature that enables a multipoint video conference to compose a picture consisting of the video from multiple sites participating in the conference. As such, the feature allows one or more conferees to see up to 16 sites in the conference at one time.

convener
The person who schedules a conference and is responsible for distributing the Meet-Me numbers.

COR
See *Class of Restriction*.

COS
See *Class of Service*.

CPP
See *Continuous Presence Plus.*

CRCS
See *Lucent Technologies Conference Reservation and Control System*.

D

DA
See *Dedicated Access*.

Data Communication Equipment
Equipment that provides the functions to establish, maintain, and terminate a connection, the signal conversion, and coding required for communication between the Data Terminal Equipment (DTE) and the data circuit.

Data Terminal Equipment
Equipment comprising the data source, data sink, or both the source and sink.

DCE
See *Data Communication Equipment*.

DCP
See *Digital Communications Protocol*.

dedicated
A conference class that reserves a specified number of MCU ports for a multipoint video conference at any time.

Dedicated Access
A feature that enables endpoints to participate in multipoint audio and video conferences via non-signaled T1 facilities. For this feature, the MCU connects to a Multiplexer (MUX), a Digital Access Crosspoint System (DACS), or to H.320 endpoints via DS1 facilities. A maximum of 20 DS1 connections to the MCU are supported.

Dial-Out
A feature that allows H.320 calls to originate (dial out) from the MCU in addition to receiving dial-in calls. Therefore, conference participants can join conferences by receiving calls from the MCU.

Digital Communications Protocol
Proprietary protocol used to transmit both digitized voice and digitized data over the same communications link. A DCP link is made up of two 64k information (I-) channels and one 8k signaling (S-) channel. The DCP protocol supports two information-bearing channels and, therefore, two telephones or data modules.

DTE
See *Data Terminal Equipment*.

Dynamic Conference Resizing
A feature that allows the user to add and remove conference participants before the conference starts as well as during the conference.

E

electronic tandem network
Tandem tie-trunk network that has automatic call-routing capabilities based on the number dialed and the most preferred route available. Each switch in the network is assigned a unique private network office code (RNX), and each voice terminal is assigned a unique extension.

encoding
Changing video and/or audio signals into digital form for more efficient and accurate transmission (movement) from one entity to another.

endpoint
A video unit such as a camera, PC, video recorder, speakers, and other equipment used for transmitting multimedia (voice, audio, full-motion video, and graphics) during an MCU conference. The video capabilities of an endpoint can be linked to a specific site.

entry tone
A tone that alerts conferees already on a conference that another conferee has joined in the conference.

ETN
See *electronic tandem network*.

Executive Conferencing
A customer option that enables a conference to compose a picture of two or three “other” sites (endpoints) for each conference participant. With this feature, no endpoint ever sees itself.

exit tone
A tone that alerts conferees on a conference that a conferee has left the conference.

F

FAC
Facility Access Code.

far end
When troubleshooting problems on the MCU, the video endpoint is often referred to as being on the far end of the connection.

Far-End Camera Control
A feature that allows a site that requests far-end camera control to control the camera of the far-end site that it is currently viewing.

FCIF
See *Common Intermediate Format*.

FECC
See *Far-End Camera Control*.

file
A conference class that saves a copy of a Conference Record for future use.

G

G.711 Audio
See *Pulse Code Modulation*.

G.722 Audio
An audio mode that enables the system to bridge G.722 (7kHz) audio. 7kHz is the audio bandwidth provided by G.722. This bandwidth allows for a more natural sounding voice conference. The MCU supports G.722-conferenced audio at 48k or 56k, as defined in ITUT-T Recommendation G.722.

G.728 Audio
See *LD-CELP*.

GCC
See *Generic Conference Control*.

Generic Conference Control
A high-level framework for conference management encompassing conference establishment and termination, conference roster, application roster, remote actuation, conductorship, bandwidth control, and registry services.

H

H0
A one-channel 384k bandwidth call.

H11
A one-channel 1472k or 1536k bandwidth call.

H12
A one-channel 1920k bandwidth call.

H.320
The multipoint conferencing standard adopted by the ITU-T. This standard includes some of the other H-series standards governing the definition of multimedia equipment and equipment transmission (movement of data) capabilities and methods.

H.323
ITU standard that governs videoconferencing over local area networks (LANs) and the Internet.

H.323 Gateway
Device that enables an Internet protocol-based H.323 terminal to communicate with an H.320 terminal and with a regular telephone on a switched network.

H-series
See *H.320*.

handshake
An expression relating to the establishment of communications between the MCU and a video endpoint.

heterogeneous conference
A conference in which the endpoints are expected to be either H.261 or H.263 video capable.

homogeneous conference
A conference in which all the endpoints are expected to have the same video mode (for example, H.263 video).

I

IMUX
See *Inverse Multiplexor*.

In use
A conference status when a reserved conference has begun and at least one conferee has joined the conference.

Inactive
A conference status when a conference is reserved but has not begun and when a conference has a class of file.

INADS
See *Initialization and Administration System*.

Initialization and Administration System
An asynchronous modem that originates alarms to a remote maintenance system and allows remote technicians to run maintenance and administration commands.

International Telecommunications Union-Telecommunications
An international group established in 1993 to develop standards to allow video endpoints to communicate with each other.

Inverse Multiplexor
Device used to create a single higher-speed data channel by combining and synchronizing two or more lower-speed data channels. This device is used primarily with video codecs and data bridges/routers for speeds up to 2 Mbps.

ISDN-BRI Trunk Interface
A feature that provides ISDN-BRI connections to the public network. This feature is intended primarily for customers with relatively small line sizes in countries and geographic locations where BRI lines are less expensive than PRI lines.

ITU-T
See *International Telecommunications Union-Telecommunications*.

L

LD-CELP (G.728)
A means of encoding audio so a smaller portion of bandwidth is allocated to audio. This results in greater bandwidth for video, thus improving picture quality.

Lip Synchronization
Feature that provides additional or no audio delay within an MCU CPP conference.

login
A keyword assigned to a user to access the MCU.

login password
A code word that can be customized by the user that corresponds with the user's login.

Low-/High-Speed Interworking
A feature that allows 56k or 64k endpoints to join higher-speed conferences as Audio Add-On endpoints via conventional dial-in H.320 ports. Multiple H.320 ports in a conference can implement this feature simultaneously

Lucent Technologies Conference Reservation and Control System
A PC-based reservation system that manages the MCU reservation system, automates the scheduling tasks, performs conflict resolution, and ensures that participating video endpoints have the proper capabilities to join the conference.

Lucent Technologies MultiPoint Conferencing Unit
Equipment that provides high-quality multimedia conferencing with video endpoints that communicate via the ITU-T H.320 standards.

Lucent Technologies Video Technical Center
A video technical support group that is available to help with questions about the Lucent Technologies MCU.

M

Management Terminal
A 715GBCS terminal used primarily for administration and maintenance functions.

Manipulate Most Recently Joined Endpoint
UCC option that allows the user to perform various operations on the endpoint that most recently joined the MCU conference. These operations include broadcasting, muting, unmuting, and dropping the most recently joined endpoint; performing Solo Audio on the most recently joined endpoint; and setting the most recently joined endpoint into a quadrant.

Manufacturer and Product Information Exchange
The MCU exchanges ns-caps with each endpoint that joins the conference. Besides information regarding capabilities, ns-cap messages contain manufacturer and product information for the MCU or endpoint. Such information received by the MCU from the endpoint is displayed on the Status Conference form.

MCU
See *Lucent Technologies MultiPoint Conferencing Unit*.

MCU-extension
An extension on the MCU that corresponds to the number a conferee dials to join a multipoint video conference.

MCU-Extensions and Numbers to Dial Form
A list of all MCU-extensions, supported bandwidth, and corresponding Meet-Me numbers that conferees dial to join a conference.

MCU Express
A process that focuses on installing and provisioning the MCU more expediently. Components include customer self-installation and self-maintenance, improved customer training, firmware upgrade, and CRCS ease-of-use enhancements.

MCU-MT
See *Management Terminal*.

MCU-ST
See *Scheduling Terminal*.

Meet-Me number
A number a conferee dials to join a multipoint video conference.

Mixed conference mode
Feature that allows the MCU to support both Meet-Me and Dial-Out endpoints in the same conference.

modem pooling
Capability that provides shared conversion resources (modems and data modules) for cost-effective access to analog facilities by data terminals. When needed, modem pooling inserts a conversion resource into the path of a data call. Modem pooling serves both outgoing and incoming calls.

MSM
Multimedia Server Module.

multimedia
The use of a variety of media, including audio, data, graphics, and full-motion video.

multipoint video conferencing
A technique of combining video, audio, and data from three or more video endpoints onto a single video conference call.

multirate bandwidths
Bandwidths of one channel that use an ISDN-PRI facility from an endpoint that provides the appropriate size bandwidth. These bandwidths include 112k, 128k, 168k, 192k, 224k, 256k, 280k, 320k, 336k, 384k, 448k, 512k, 672k, 768k, 1008k, 1288k, 1344k, 1472k, 1536k, 1680k, and 1920k. The 56k increments can be multirate only when the endpoint is directly connected to a Data Interface circuit pack.

N

network service
The telephone company that provides the trunks and Meet-Me numbers to access the MCU.

network type
Specifies whether a video endpoint is connected to a private network (behind a PBX) or a public network.

non-transcoded conference
A conference that does not support transcoding between H.263 video and H.261 video.

notification tones
The entry, exit, and warning tones available with some of the MCU models.

O

OneNumber Access
Feature that allows multiple endpoints in a conference to access the MCU from the same external extension. A OneNumber Access extension is a duplicated MCU-extension within a single conference. Endpoints sharing a OneNumber Access extension are not preassigned to a specific conference endpoint number; instead, these endpoints share a pool of endpoint numbers and receive a conference endpoint ID based on when they join the conference.

P

PacketStar Access Concentrator- (AC-) 60
A component that provides an ATM interface to the MCU The AC-60 aggregates H.320 video traffic at premises on the user side, adapts this traffic to ATM format, and then transmits it over an ATM network on the network side.

panoramic
Continuous Presence *Plus* configuration in which a video image appears within each of the two lower quadrants of and the upper half of the screen. The user can therefore see up to three participants. Also known as “three-image.”

paper-based scheduling system
A method of recording and tracking conference reservations to prevent overbooking of the MCU.

PC-based scheduling
A scheduling method that uses the Lucent Technologies Conference Reservation and Control System to accept reservations and qualify video endpoints to participate in a multipoint video conference.

PCM
See *Pulse Code Modulation*.

per-conference password
A password for a conference that is provided by the customer at conference reservation time. Conferees are prompted for the password when they join the conference.

per-user password
A feature that allows individual passwords to be assigned to each dial-out H.320 endpoint as well as to the Audio Add-On port (if present) in the conference. Therefore, the feature provides a higher level of security than does the per-conference password feature.

point-to-point
A conference involving only two sites whose video endpoints are not connected through the MCU.

port
A logical entity where the video call terminates on the MCU. It provides a specific bandwidth (data rate) capacity, such as 56k, 64k, etc.

Presentation
A conference mode where a video endpoint’s video is constantly broadcast to the other locations. However, the audio portion of the conference switches according to who is speaking. Any conferee can be a speaker. The video endpoint that is the presenter views endpoints determined by voice-activated switching.

PRI
See *primary rate interface*.

primary rate interface
A standard ISDN frame format that specifies the protocol used between two or more communications systems. PRI runs at 1.544 Mbps and, as used in North America, provides 23 64k B-channels (voice or data) and one 64k D-channel (signaling). The D-channel is the 24th channel of the interface and contains multiplexed signaling information for the other 23 channels.

protocol
A set of international industry-standard rules governing the exchange of data between two entities.

Pulse Code Modulation
A means of encoding audio by assigning less bandwidth to video. G.711 audio mode is used.

PVC
See *permanent virtual circuit*.

Q

QCIF
See *Quarter Common Intermediate Format*.

Quarter Common Intermediate Format
An industry-standard means of encoding video signals to produce medium-quality pictures by encoding only a quarter of the video resolution of CIF.

R

Rate Adaptation
Feature that allows the Lucent Technologies MCU to interwork endpoints operating with 2B-channels that are on both 56k and 64k networks.

RBS
Robbed-Bit Signaling.

Real-Time Conference Status
Feature that allows the real-time status of a conference and its associated conferees to be displayed on the MCU-MT. The **monitor conference** command is used for this purpose. Once the command is issued, the display is automatically updated approximately once every 30 seconds. Accordingly, this feature is intended for troubleshooting purposes.

recurring meeting
A meeting that occurs more than once in a definite pattern, such as every Thursday.

reservations agent
The person responsible for accepting multipoint conference reservations, making reservations on the MCU-ST or CRCS, and resolving minor problems.

reserved
A conference class that indicates a conference will begin and end within the next 24 hours.

RNX
See *Route-Number-Index*.

Rollcall
A Universal Conference Control option that allows the broadcaster to scan endpoints in the conference without specifying an endpoint number for each scanned endpoint. With this option, all endpoints in the conference see a change in video caused by the scan.

rotation scan time
The number of seconds that a broadcaster views each location before viewing the next location in the rotation.

Route-Number-Index
Private network office code.

RS-232C
A physcial interface specified by the Electronic Industries Association (EIA). RS-232C transmits and receives asynchronous data at speeds of up to 19.2k over cable distances of up to 50 ft.

RS366
EIA standard for interconnecting data terminal equipment and automatic calling equipment for data communications.

RS366 Dialing
Feature that allows H.320 codecs that are directly connected to the MCU via a V.35/RS449/EIA-530 interface to dial into multimedia, multipoint conferences. This allows endpoints to join conferences after the conference start time and to drop from conferences early. Also allows the aforementioned codecs to place and receive spontaneous 2B and multirate point-to-point calls with other endpoints directly connected to the MCU and other endpoints anywhere in the network. Finally, allows H.323 endpoints connected behind H.323 gateways to participate in multimedia, multipoint conferences for which there is a V.35/RS449/EIA-530 connection between the gateway and the MCU. The H.323 endpoints can dial into MCU conferences.

RSC
Regional Support Center.

S

scan time
See *rotation scan time*.

scheduling terminal
Also known as the MCU-ST. A 715 Business Communications Terminal (BCT) used for reserving conferences on the MCU and for checking the status of conferences and availability of MCU ports.

SCI
Switch Communication Interface.

SCO
System Control Office.

SDN
Software Defined Network.

site profile
A list of video endpoints that may participate in multipoint video conferences and their location, time zone, bandwidth, and network type.

SM
See *Speed Match*.

SPE
See *Switch Processing Element*.

Speed Match
Feature that enables endpoints with varying audio modes, video formats, and transfer rates to participate in the same MCU conference. The feature enables low-speed desktop video endpoints to interwork with high-speed group system video endpoints in a multipoint conference with full video and audio capabilities.

status
The current state of a conference. A reserved conference status can be active, in use, or complete. A file conference status is always inactive. A dedicated conference status is always active.

Switch Processing Element
A complex of circuit packs (including the processor, memory, and bus interface cards) that are mounted in a control carrier. The SPE serves as the control element for the MCU.

system administrator
The on-site telecommunications manager who administers the MCU trunks, maintenance alarm terminals, dial plan, passwords, MCU-extensions and numbers, and system time. The system administrator can request system reports to check on MCU performance and also has access to all conference-related forms.

T

T.120
The suite of standards defined by the ITU-T in support of data conferencing. The Lucent Technologies MCU includes support for the T.120 infrastructure recommendations, including T.123 (transport), T.122/T.125 (MCS), T.124 (GCC), T.126, and T.127.

TDM
See *time division multiplexing*.

Terminal Names
A feature that enables the MCU to poll and ascertain “naming/identification” parameters from endpoints and to pass the information onto other endpoints. Therefore, conference participants whose endpoints display this information can identify other conference participants.

time division multiplexing
Multiplexing that divides a transmission channel into successive time slots.

tones
See *Notification Tones*.

U

UCC
See *Universal Conference Control*.

UDP
See *Uniform Dial Plan*.

Uniform Dial Plan
A feature that allows a unique 4- or 5-digit number assignment between the MCU and another MCU or PBX.

Universal Conference Control
A customer feature that enables an endpoint to manage an H.320 conference out-of-band. UCC is intended to provide a number of powerful services similar to, and in lieu of, H.243 Chair Control. The controlling terminal for UCC *Star* is a regular touchtone telephone, such as that used by an Audio Add-On party. The terminal does not receive audio, video, or data from the conference it is managing, and it does not transmit this media to the H.320 conference multiplex. UCC *P&C* (Point and Click) provides an alternative user interface to UCC *Star*; namely, a graphical-user interface (GUI) on a PC.

V

VAS
See *Voice-Activated Switching*.

video endpoint
A video codec, with camera, speakers, screen, and other equipment required for multimedia conferencing.

Video-Switching Mode and Broadcaster Notification
A feature used by the MCU to announce to all the conference endpoints the current video-switching mode and broadcaster. The feature is also used whenever the endpoint initiates a See-Me (User-Activated Presentation) request that cannot be honored.

Voice-Activated Switching
A conference mode where the video image of the person speaking is seen by all other conferees. The speaker sees the video image of the previous speaker. When a new speaker begins talking, the video switches to the new speaker.

VTC
See *Lucent Technologies Video Technical Center*.

W

warning tone
A tone that sounds when only 10 minutes remain in a conference.

Web.Res.Now
Web-based conference reservation system that manages dedicated MCU conferences.

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