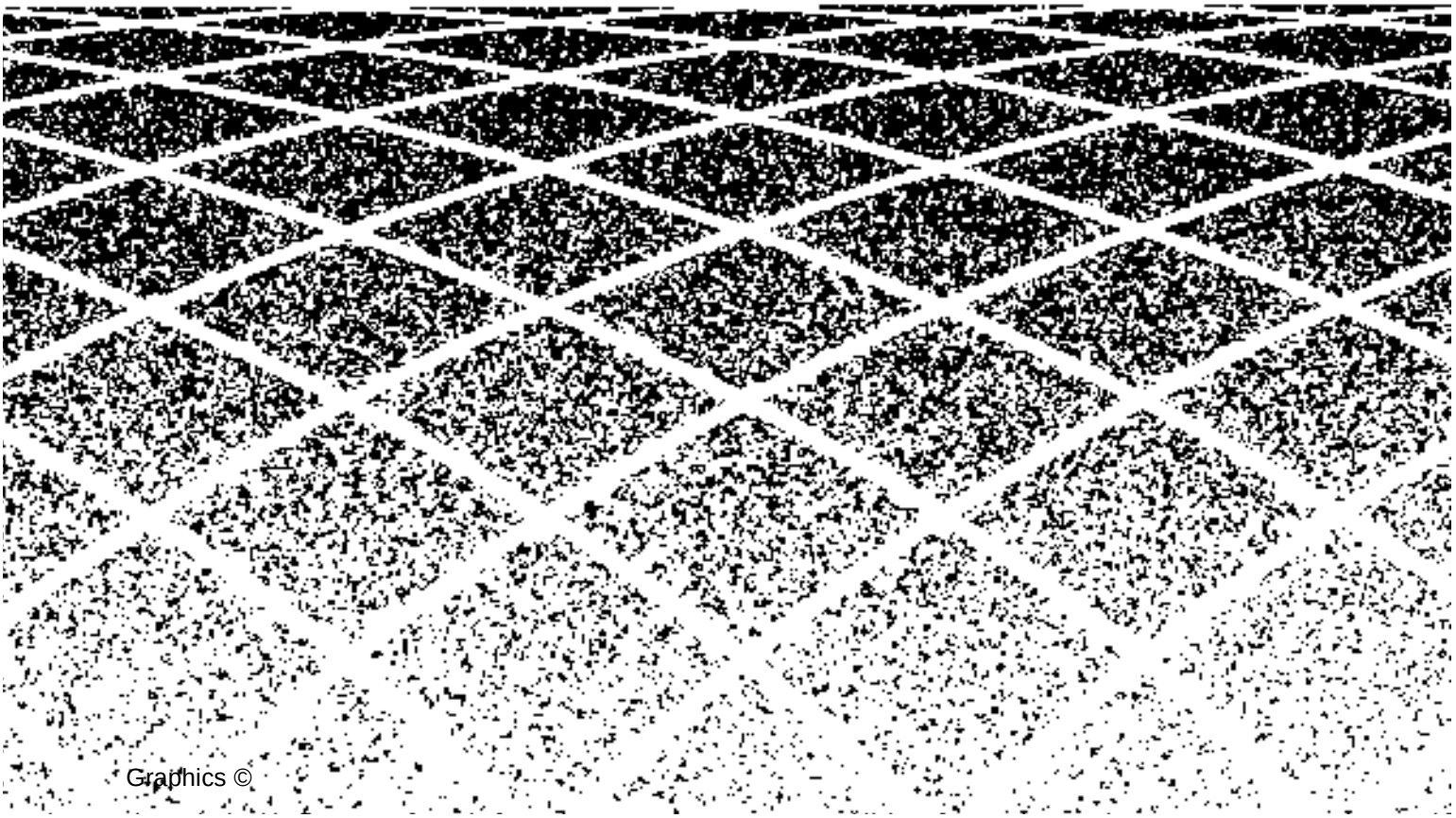
555-027-727
Issue 3
September, 1994

MultiPoint Control Unit System Administration and Reports



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About This Document

This manual describes the AT&T MultiPoint Control Unit (MCU) administration functions and reports available to a telecommunications manager or an on-site MCU system administrator.

Prerequisites

It is assumed that before you use this manual 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 manual in conjunction with the following AT&T MCU documents and information sources:

- *AT&T MultiPoint Control Unit Installation Quick Reference,* 555-027-723

  Provides brief instructions on installing an AT&T MCU and components to enable you to do video conferencing.

- *AT&T MultiPoint Control Unit System Description,* 555-027-721

  Provides an introduction to the MCU, including descriptions of models, configurations, features, and facility requirements.

- *AT&T MultiPoint Control Unit Maintenance,* 555-027-724

  Provides information about tests, alarms, and the actions you need to take to maintain your AT&T MCU.

- *AT&T MultiPoint Control Unit Reservation Agent Manual,* 555-027-725

  Provides detailed instructions an AT&T MCU reservation agent or telecommunications manager can use for scheduling and reserving conferences and troubleshooting using the MCU Scheduling Terminal (MCU-ST).

- *AT&T MultiPoint Control Unit Seminar,* BM1068U

  A course offered by AT&T to provide you with an overview of the AT&T MCU administration and scheduling functions.

- *AT&T Conference Reservation System User's Manual,* 555-230-520

  Provided with the optional AT&T Conference Reservation System, this manual gives detailed information and procedures an AT&T CRS administrator and/or reservations agent can use to install CRS, set up and manage CRS databases, reserve conferences and perform CRS system administration.

Organization

This manual contains the following three chapters and two appendices:

- Chapter 1, "MultiPoint Conferencing" provides an overview of the technology that led to the development of the AT&T MCU. It also gives a description of the features of the MCU.
- Chapter 2, "MCU System Administration" explains how to perform ongoing administration procedures, such as adding trunks.
- Chapter 3, "Checking MCU Performance" provides a list of MCU reports you can display or print to track MCU performance and status of trunks.
- Appendix A, "P x 64 Interoperability" presents a list of the video endpoints that are supported by the MCU, along with any limitations in features or functionality. This list is current as of the print date for this document. For the most current list, consult your account executive.
- Appendix B, "Paper-Based Scheduling Forms" provides blank copies of the forms described in Chapter 2, "MCU System Administration".
- Appendix C, "Application Notes" describes how the MCU should be configured and administered for operation in each supported country. To this purpose, the required circuit packs are listed, and the administration of the relevant features and system parameters is described.

This manual also has a Glossary and Index.

Typographic Conventions

Several conventions are used to quickly convey information. 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) at 800 242-2121. If you are a customer in Australia or the United Kingdom, call the International Technical Access Center (ITAC) at 303 538 4666.

MultiPoint Conferencing

1

Overview

The AT&T MultiPoint Control Unit (MCU) Release 2.0 brings multiple remote video (and, optionally, audio-only) endpoints onto a single video conference call. It is a multimedia product since it combines audio and video. This chapter gives background information on video/audio conferencing and the tasks you will be performing.

Standard vs. Proprietary

Until recently, video endpoints, like the original fax machines, could make contact only with other video endpoints developed by the same manufacturer. This was due to the fact that the technology, both video telephony and video conferencing, was built upon a manufacturer's private protocol, known as proprietary algorithms.

In 1990, the international standards body ITU-T (formerly CCITT) adopted a set of standards or rules that allow video product manufacturers to develop products with the ability to communicate with each other. These standards are collectively known as the H-series or Px64.

Multipoint Conferencing

Multipoint video conferencing takes the Px64 standards a step beyond two video endpoints communicating with each other (known as point-to-point). Multipoint video conferencing brings together multiple video endpoints at geographically dispersed locations into a single conference call. This requires tracking of video images, sound, and data from multiple sites for a simultaneous multimedia presentation.

Unless the Px64 conference is cascaded (see "Cascading" on page 1-8), only one non-Px64 audio-only endpoint can be included in a Px64 conference on an AT&T MCU. A cascaded conference allows two such endpoints. However, an audio-only endpoint can send a call from an audio bridging service. Therefore, multiple audio-only endpoints can participate in the conference via this service.

NOTE:
You can include Plain Old Telephones (POTS) as non-Px64 Audio-only endpoints.

Meet-Me Numbers and MCU-Extensions

As with any endpoint on a network (such as a telephone, computer, and video codec), the MCU is assigned network numbers, known as Meet-Me numbers. These numbers are used by conferees to dial into the MCU to participate in a multipoint conference. Each MCU is assigned multiple Meet-Me numbers, thus allowing multiple video endpoints to connect to the MCU simultaneously.

Each Meet-Me number is mapped to an MCU-extension on the MCU. Conferees will not be aware of the MCU-extension being reached since they are dialing a Meet-Me number. However, you must be aware of the mapping of Meet-Me numbers to MCU-extensions in order to reserve conferences on the MCU.

NOTE:
See *AT&T MultiPoint Control Unit (MCU) System Description* for details on ordering Meet-Me numbers.

Administration of Meet-Me numbers on the MCU requires coordination between you and your network service provider. There are two primary steps in administering Meet-Me numbers. First, obtain the Meet-Me numbers from the network service provider and ensure the network DS1 trunks are properly administered to route the Meet-Me numbers to the MCU. As each call arrives at the MCU, the network signals the Meet-Me numbers dialed by the video endpoint to reach the MCU. The MCU uses these digits to determine which MCU-extension, and ultimately which conference, will receive the incoming call. You can manually map Meet-Me Numbers to their respective MCU-extension within the MCU-Extension form (see Chapter 2, "MCU System Administration" for details).

Typically, the network service provider is instructed to signal only the last four digits of the Meet-Me number to the MCU. MCU-extensions must be administered within the MCU to correspond to these digits signaled to the MCU by the network.

Secondly, you must administer the MCU dial plan to create MCU-extensions. MCU-extensions are used to correspond to MCU ports for a particular conference. The MCU offers a pool of MCU video ports. For each conference, the MCU dynamically allocates the appropriate number of video ports and assigns an MCU-extension to each video port.

When a reservation agent reserves a conference, an MCU-extension is added for each conferee. This process tells the MCU to reserve a video port for the duration of the conference and label it with a given MCU-extension. Once the conference is over, the video port is put back into the pool and the MCU-extension is again available for use.

Joining a Conference

Conferees can join a multipoint video conference by either dialing into the MCU (explained here) or receiving a call from the MCU (explained later in this chapter, "Dial-Out" on page 1-8).

For the former procedure, conferees use Meet-Me numbers. A Meet-Me number is usually a standard 10-digit phone number (often with 700 as the area code), a 7-digit PBX number or a 4- to 5-digit extension number that is part of a private dial plan.

NOTE:
Meet-Me numbers may be longer than 10 digits for international calls or for other configurations and access codes.

When a conference is scheduled to begin (and any time before the conference ends), conferees simply dial their Meet-Me number from their video endpoint to join in to the multipoint conference. Conferees need not join the conference in any particular order.

MCU Features

To assist conference conveners in configuring multipoint video conferences that maximize their conference experience, it is important to know how many MCU ports are available for conference calls and the features and capabilities offered to conferees. The following section describes the choices available.

NOTE:
The MCU is offered in different models with various options (see the *Multi-Point Control Unit System Description* for details). The full-featured EX

model is described here. Feature sets vary depending on the MCU model, options selected and capabilities of the participating endpoint.

Number of MCU Ports

An MCU port is a collection of MCU resources that allow a video endpoint using a supported transfer rate (bandwidth) to connect to the MCU and participate in a multipoint conference. It includes trunking resources (trunks), video codec and conferencing resources and Px64 protocol termination resources.

Each MCU port can dynamically support various transfer rates. Transfer rate refers to the number of channels connected to a conference multiplied by the bandwidth of the channel. B-channel refers to a full-duplex, switched digital 56k/64k data channel provided by the network from a source (the video endpoint) to a destination (the MCU). A 2B-channel doubles the data channel bandwidth of the conference call from 56k/64k to 112k and 128k respectively. H0 refers to a transfer rate of one channel with a 384k bandwidth. The MCU supports video calls comprised of 2B-channel 56k or 64k. The MCU also supports the following one channel rates: 384k (H0), 768k, 1472k, and 1536k. In addition, bonding (discussed in depth later in this chapter) provides additional bandwidths.

The MCU is offered in configurations based on MCU port resources and the MCU model. The first number in the configuration is the total number of MCU ports; all can be used for 2B conferences. The second number (always equal to or less than the first number) represents the number of MCU ports that can be used for H0 or one channel conferences. Also, one non-Px64 audio-only endpoint (or two if cascading is enabled) may be included in either type of conference (this is not indicated in the configuration).

The smallest configuration is 4/4 with four MCU ports available for 2B or 1B conferences. A four port system accepts a maximum of one four-party conference. The largest configuration is a 64/32 port system. With a maximum single 2B conference size of 64, the 64/32 port configuration can handle many combinations of conferences (such as four 16-party conferences or eight 8-party conferences). With a 64/32 configuration, you can use up to 32 MCU ports for H0 conferences. The following configurations represent the maximum port capacity involving each of the other 1B bandwidths (indicated in parentheses): 64/28 (for 768k), 64/16 (for 1472k), 64/16 (for 1536k). Finally, the maximum bonded 336k/384k port capacity is 12.

Conference Modes

Meetings are conducted in a variety of ways. Some meetings require interaction among all conferees; others focus on one speaker who provides information to other conferees. Conference modes allow you to adjust the MCU to operate according to the type of meeting being held. A mode controls who is being seen at a given moment in a conference.

Conference modes are available according to the MCU model. Accordingly, some MCU models may feature conference modes that may not be available on other models.

The following conference modes are available:

Voice-Activated Mode (Automatic Control)

With voice-activated conference mode, the video image of the person speaking is seen by all other conferees except the speaker who continues to see the previous speaker. There is a minimal length of time between when a speaker stops talking and the video switches to the new speaker. When a new speaker becomes a video source, the MCU waits a moment before switching again to prevent a “ping-pong” appearance. A ping-pong appearance is caused by switching the video source too rapidly during a conversation that involves frequent changes of speaker. This conference mode is ideal for meetings where all participants share information. This automatic mode is a part of the MCU functionality and requires no special features from the video endpoint.

Chair Person Mode (User Control)

For this mode, the chair person determines which video sending endpoint is broadcast to all the other conferees. The chair person can change the video broadcaster at any time. Each conferee can hear audio from any other conferee. Any conferee can acquire the data token and initiate a data broadcast. Chair Person Control requires endpoint support per ITUT-T H.243 standards.

Broadcast Mode with Auto Scan (Advanced Control)

In this mode, everyone in the video conference sees and hears one location continuously and unchanging. The location being viewed (the broadcaster) does not receive audio from any viewing location. The broadcaster does see video from each of the conferees on a continuous rotating basis (the rotation scan time can be adjusted when the conference is reserved). If the conferees join the conference before the broadcaster, they are added to the conference but view nothing until the broadcaster joins. If applicable, conferees will hear entry tones as other conferees join the conference. The advanced mode is part of the MCU functionality and requires no special features from the video endpoint.

Presentation Mode (Advanced Control)

Presentation mode allows one location to be viewed constantly and unchanging by all conferees, but unlike broadcast mode, all conferees including the presenter have audio from all sites. When anyone at the participating locations asks a question or makes a comment, the presenter hears and views that location. The other conferees hear the question or comment but continue to view the presenter. This conference mode is ideal for classroom training and instructional meetings where there is one primary speaker and limited interaction from the conferees. If the conferees join the conference before the presenter, they are conferenced in but view nothing until the presenter joins. If applicable, conferees will hear entry tones as other conferees join the conference. This advanced mode is part of the MCU functionality and requires no special features from the video endpoint.

Video Quality

You can adjust the video quality according to the capabilities of the participating endpoints and the network interfaces used by both the endpoints and the MCU. Video quality is controlled by the amount of transfer rate and bandwidth allocated to video and the absolute amount of transfer rate selected for a given conference. As mentioned earlier, the transfer rate is based on the number of B-channels associated with a conference call.

The bandwidth for a video call is factored by the number of channels selected for a conference. The broader the bandwidth, the better the video quality. Therefore, for 2B transfer rates, 112k (two 56k channels) is the basic video quality; 128k (two 64k channels) provides better video quality. The one channel transfer rate of 1536k provides the maximum bandwidth and thus the finest video image available with the MCU.

In most cases, the MCU will not allow an endpoint to participate fully (audio and video) in a conference if it does not support the selected (or higher) bandwidth of the conference. However, rate adaptation (discussed later in this chapter) allows an endpoint calling with a 56k bandwidth to join a conference arranged with a 64k bandwidth and participate as a “full” participant.

The best video available for a 56k/64k video conference is achieved using a method of speech compression known as G.728 Low Delay Codebook Excited Linear Prediction (LD-CELP). G.728 uses 16k audio. Basically, the MCU and the video endpoints take the total amount of bandwidth on a conference call (regardless of whether it is 56k or 64k) and use a larger percentage of the bandwidth for video. The reduction in the audio bandwidth is indiscernible to most people. With maximum bandwidth for video, video quality is enhanced.

The highest quality audio you can arrange for a video conference is known as G.711 Pulse Code Modulation (PCM). G.711 is 56k/64k audio. Essentially, this selection maximizes the bandwidth allocated to audio. Therefore, the video quality is not as good as that which is offered with G.728.

NOTE:
Not all supported video endpoints offer G.728. For a multipoint conference to operate using G.728, all participating endpoints must support G.728. If one does not, the MCU automatically adjusts the conference to G.711.

Notification Tones

Notification tones are used during a conference to alert conferees about the status of a conference. Each tone has a different length and frequency so it can be distinguished from each other. Tones are selected on a per conference basis at the time a conference is reserved.

Entry and Exit Tones

When a conferee joins a conference, an entry tone alerts other conferees already on the conference that another endpoint has joined. Similarly, when a conferee disconnects from a conference, an exit tone notifies the other conferees that an endpoint has dropped.

Warning Tone

This tone sounds when only 10 minutes remain in a conference. This tone gives advance notice to conferees to either conclude the conference or request additional time to extend the conference.

Billing Information

In cases where the conference convener wants to track the reason for a conference and its duration for billing purposes, you can assign a billing identification number (up to 15 characters). The billing number associates the conference with a particular client, project or whatever information is necessary for proper accounting.

Audio Add-On

Audio Add-on is a customer option that allows the MCU to support one non-Px64 Audio-only endpoint participant per conference. This option is enabled via the Optional Features form. Audio Add-on participants hear the same tones (when applied) as Px64 endpoints. You can increase Audio Add-on capacity in increments of two.

Audio Add-on port capacity is considered supplementary to Px64 Multimedia port capacity. For example, if there are four Audio Add-on ports on a 24 port system, the system is viewed as one containing 24 Px64 ports plus four Audio Add-on ports. Accordingly, the expression “four port conference optioned with an Audio

Add-on port," for example, implies that the conference involves four Px64 ports plus one Audio Add-on port.

Audio Add-on functions only with Meet-Me endpoints (and not with the Dial-out feature). This protects against security violations. Also, Audio Add-on provides an optional Digital Tone Multifrequency (DTMF) password security capability. Finally, an Audio Add-on party may participate in a bonding conference by dialing into the AT&T MCU.

Dial-Out

Dial-out allows the MCU to originate calls and dial to the appropriate endpoints. Therefore, conference participants can be joined to conferences by receiving a call from the MCU. Each conference participant who joins a conference in this manner is identified as a Dial-out participant. This identification is made during conference reservation.

At conference start time, the MCU automatically originates a call to each Dial-out participant. A Dial-out call can use two different telephone numbers per endpoint. Once the Dial-out participant entry is submitted, the feature immediately tries to "set up" the call by initiating it to the endpoint. If the call fails, the MCU retries the call either up to four times by default or up to nine times via administration of the Feature-Related System Parameters form, which contains a Dial-out Post Answer Failure Retry Limit field. The MCU does not consider a call to be "set up" until the MCU has fully established a Px64 call to an endpoint (that is, the network is established and the protocol handshaking is completed).

The MCU provides Dial-out failure alarms via the DCP maintenance terminal. Such an alarm is required whenever the MCU is unable to establish a connection to the endpoint. Alarming via the telephone alerts the appropriate person (for example, the Reservations Agent) to investigate the problem by using the status conference command.

Mixed Conference Setup Mode

The MCU can support both Meet-me and Dial-out endpoints in the same conference. During conference reservation, each participant in a conference is identified as either a Meet-Me (Dial-in) or Dial-out participant.

Cascading

Cascading allows one MCU to connect to another MCU on a per conference basis. As such, this feature enables endpoints on each MCU to participate in a joint conference. The endpoints may be bonding or non-bonding endpoints. As

the first sentence implies, only one cascade per conference is allowed. However, note that each MCU can support up to 24 simultaneous cascades (for 24 different conferences).

Cascaded MCUs are connected to each other by an inter-MCU or cascade link. Each such link is unique to only one cascaded conference. In every case, the link uses a single conference port that is reserved via administration. This port is called the inter-MCU link port or cascade port. Cascade links are supported as endpoints by the bonding feature.

The total number of ports allotted for each cascaded conference includes a tally for the cascade port. For example, an MCU that is participating in a cascade with four endpoints requires five ports for the conference (the additional port is used by the cascade link). A maximum of 23 Px64 parties can be included in a conference on a single cascaded MCU. Joining two MCUs together via a cascade enables a maximum conference size of 46 Px64 parties.

Each cascaded MCU can support just one non-Px64 Audio-only endpoint. Therefore, a cascaded conference can support a maximum of two Audio-only endpoint participants (one per MCU).

At conference start time, the dial-out (originating) MCU establishes the cascade link by dialing a call to the dial-in (receiving) MCU.

NOTE:
The Reservation Agent determines the dial-out MCU and the dial-in MCU during administration of the cascade link.

NOTE:
The originating MCU also immediately attempts to establish a cascade link call whenever the Reservation Agent adds a cascade link record to a conference already in progress.

The MCU provides a recovery and retry procedure for automatic calls that result in failed call attempts. Accordingly, the MCU retries each original call either four times (by default) or (for post-answer failures only) nine times via administration of the Feature-Related System Parameters form.

For failed call originations, the MCU provides inter-MCU link failure alarms via the MCU Maintenance Alarm Terminal. For example, a Reservations Agent with a co-located MCU Maintenance Alarm Terminal could respond to an alarm on the terminal's "dial-out button" by using the status conference command to investigate the problem.

You can use cascading to join two conferences on a single MCU via an external MCU-link call. Therefore, a single MCU can cascade with itself via the network.

Cascading is optioned via the Optional Features form.

Dynamic Conference Resizing

This feature allows the Reservation Agent to add and remove bonding or non-bonding conference participants before the conference starts as well as during the conference. The feature is applicable to all types of conference participants (including Meet-Me, Dial-out, Audio Add-on, and Cascade links). This feature can add to a conference as many participants as is allowed according to the established limits for both conference participant types and available MCU resources. For example, resources permitting, the MCU never allows more than one Audio Add-on port to a conference. On the other hand, the number of allowable Px64 endpoints within a conference is restricted only by the number of available Px64 ports for the duration of the conference.

Also, the feature allows you to remove and change dial-out numbers and to change the stop time on a reserved conference. Finally, the feature allows you to change dial-out numbers to dial-in numbers (and vice versa), and it allows you to change a primary MCU to a secondary MCU (and vice versa).

Dynamic Conference Resizing is optioned via the Optional Features form.

Bonding

Bonding (Bandwidth on Demand) allows the AT&T MCU to establish conferences at bandwidths greater than 128k without the use of ISDN-PRI and Wideband H0 channels. This feature is a standard protocol that provides the channel aggregation necessary to combine and synchronize individually placed B-channels to produce various bandwidths. Specifically, bonding can place separate 56k/64k calls to provide bandwidths of up to 336k/384k. For example, a 384k “bonded” call is comprised of six separate 64k (6 x 64) calls that route separately through the network and are aggregated together at the destination point. Similarly, a 336k bonded call is comprised of six separate 56k (6 x 56) calls that are handled the same way. The AT&T-MCU can support calls from bonding compliant inverse multiplexers, and it can place bonded calls to inverse multiplexers.

The maximum bonding port capacity is 12. The multiple calls associated with a specific bonding call (for example, 6 calls for a 384k bandwidth) may arrive over different DS1 facilities for this capacity. Also, the AT&T MCU allows the interworking of H0 (384k) calls and bonded 384k calls. This means that the endpoint(s) for each call type can participate in the same conference. The AT&T MCU treats bonded calls at different rates as separate calls.

The DS1/MMI cable or the TDM bus transmits the bonding information from the DS1 circuit pack to the bonding MMI circuit pack. Once the bonding protocol is

processed, the H.221 multiplex is transmitted via the TDM bus to the H.221 MMI circuit pack.

The AT&T MCU supports two bonding modes of operations, as follows:

- **Mode 1** provides user data rates that are multiples of the bearer rate (56k or 64k). The overhead octets are removed after the call is synchronized. Therefore, this mode does not provide an in-band monitoring function. Error conditions on one or more of the channels that disturb overall synchronization are not recognized after the call is in an active state.
- **Transparent Mode** is invoked when bonding framing is not detected. The network channels are connected to the application and bypass the bonding channel aggregation functionality. This mode is necessary whenever one endpoint supports bonding and the other endpoint does not support bonding. This mode allows 112k or 128k conferees to join a bonded conference only as a Px64 audio-only endpoint.

Bonding supports the following features: dial-in, dial-out, dynamic conference resizing, and cascading. Also, an audio-only party may participate in a bonding conference by dialing into the AT&T MCU. Finally, bonding conferences can receive entry, exit, and end-of-conference warning tones.

Bonding is optioned via the Optional Features form.

Meet-Me Number—MCU-Extension Correspondence

This feature allows you to see the Meet-Me Number listed next to its corresponding MCU-extension (provided these entries are correctly administered). This listing is available on the MCU-Extension form, the Conference Record form, and the Status of Conference form. Therefore, the feature facilitates troubleshooting and general system usage.

Rate Adaptation

The AT&T MCU uses rate adaptation to interwork endpoints operating with 2B-channels that are on 56k networks and 64k networks. Rate adaptation provides for a simpler conferencing arrangement between such endpoints.

A 2B rate adaptation conference is originally administered as a 2B 64k conference optioned for rate adaptation. Each 64k endpoint that subsequently joins the conference operates at 64k until a 56k endpoint is cut through to the conference. Thereafter, once Scheduled Conference Mode (SCM) decisions are made, the 56k endpoint causes the conference to “adjust” to an effective rate of 56k in each B-channel to all endpoints. Those 64k endpoints that properly follow the change to 56k mode qualify as full audio and video participants. However, those endpoints

that do not respond to the mode change qualify only as an audio source (and not as a video source).

If one or more 64k endpoints join the conference after the conference has been adjusted (to 56k), the MCU signals that it is conducting the conference in 56k mode and that it expects the endpoint to reduce its effective rate to 56k.

Whenever all the 56k endpoints leave a 2B rate adaptation conference, the conference continues to operate in 56k mode. Also, in cases where a 64k conference has been adjusted to a 56k conference, the “effective rate” is reset to 64k whenever all the endpoints have disconnected from the conference.

A mixture of 56k and 64k calls from any one endpoint is blocked. Such an endpoint is reduced to a 1B audio-only endpoint until either a second B-channel call with the same speed as the first call is received or the endpoint disconnects and redials.

Conference Scheduling Methods

There are two ways to schedule and reserve conferences on the MCU. Each MCU is administered to accommodate only one conference scheduling method. The methods include the following:

- PC-Based Reservation System

  The AT&T Conference Reservation System (CRS) is an optional PC-based reservation system that automates the scheduling tasks, performs conflict resolution and ensures that selected endpoints have the ability to participate in a conference. It also allows you to reserve time on the MCU in advance of the scheduled conference date and automatically reserves the conference on the day it is scheduled to begin.

- Paper-Based Scheduling System

  You may arrange conferences from the MCU Scheduling Terminal (MCU-ST) on an as-needed basis or within a 24-hour window using the internal reservation system. On-demand conferencing can be used to dedicate resources of the MCU around-the-clock. You reserve MCU ports so selected locations always have the ability to convene a conference at any time without making a reservation.

  You can also create a paper-based scheduling system to track the number of MCU ports available and reserve MCU time in advance (more than 24 hours ahead of the planned start time) of the conference date.

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 tasks include:

- Changing passwords
- Setting the system clock
- Performing video endpoint inventories
- Administering international-based parameters
- Administering trunks
- Administering maintenance alarm terminals
- Administering dial plans
- Administering Security Violation Notification (SVN)
- Administering Meet-Me numbers and MCU-extensions
- Administering cascades
- Performing system backups

NOTE:
Diagnostic and maintenance procedures are described in the *AT&T MultiPoint Control Unit Maintenance Manual*. Conference trouble-

shooting procedures are provided in the *AT&T MultiPoint Control Unit Reservation Agent Manual*.

Using the MCU-MT

The MCU-MT is a 715 Business Communications Terminal (BCT) 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.

Keys and Functions

The keyboard of the MCU-MT is equipped with cursor keys, transaction keys and editing keys to enable you to send and receive data from the MCU. These keys and their functionality are described below:

- **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 you add or change information in that field.

- **Editing Keys**

  BACKSPACE key and CLEAR are used to edit data in a field. BACK-SPACE 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-ST use the following customer logins (shown in hierarchical order).

Table 2-1. 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 Meet-Me extensions Administer features Administer conference records Perform maintenance commands on trunks, circuit packs, and maintenance alarm terminals Perform system resets
System administrator using remote MCU-ST	rcust	Same as cust login
Reservation agent using remote MCU-ST	agent	Display administration data Administer Meet-Me numbers Administer conference records
Display only (using either MCU-MT or remote MCU-ST)	browse	Display administration and maintenance data

Logging In from the MCU-MT

Perform the following procedure to log in on the MCU-MT:

1. Verify the MCU-MT displays: *Login:*
2. Enter **cust**.
3. Verify 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 715, 513, 4410, or 4425): [715]* press RETURN.
7. The screen should now display: *command:*

Logging In Remotely from the MCU-ST

You can perform administration and report functions from the remote MCU-ST. Perform the following procedure to log in to the MCU-ST:

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 the MCU-ST displays: *Login:*
4. Enter **rcust**.
5. Verify 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 715, 513, 4410, or 4425): [715]* press RETURN.
9. The screen should now display: *command:*

Changing Passwords

Perform the following procedure to log in to the MCU and to change the associated passwords:

1. Login as **cust** on the MCU-MT.
2. The screen 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 the screen 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 will need to 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 CRS is installed, the date and time on the MCU and CRS must be synchronized.

The system clock is only affected 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. Use the following procedure to set the date and time:

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 is12, 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 screen 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 Px64 (H.320) standards. In some cases, such as with the AT&T Group Video System 4000, additional Px64 (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 Px64 (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:

See Appendix A, "P x 64 Interoperability", for a list of video endpoints supported by the MCU. 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. AT&T will have no knowledge or control over any such changes made by other manufacturers in their Px64 (H.320) implementation. As such, AT&T cannot guarantee the interoperability of our ITU-T implementation with untested versions of ITU-T compliant endpoints.

See the "Collecting Endpoint Data" on page 2-8 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 (blank form available in Appendix B, "Paper-Based Scheduling Forms") for the reservation agent to use.

Table 2-2. 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	1536k	private	1150	A. Snowdon/ 619 555-2222

Collecting Endpoint Data

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

- the type of video endpoint (such as, AT&T Group Video System, AT&T TeleMedia Personal Video System, PictureTel™ Group Video System)
- the model name or number (such as Model 400, Model 70)
- the options installed
- the time zone where the video endpoint is located
- the encoding algorithms supported (Px64, proprietary)
- the specific software version (such as 4.1, 1.1)
- the bandwidths supported
- the network access (public through direct connection or private through a PBX)
- the network services supported (such as, AT&T 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.

- the name of the location or person associated with the video endpoint
- the name and number of the person responsible for setup and maintenance
- the 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 RBS network access trunks are restricted to originating and receiving only 56k calls. Endpoints originating 64k calls are blocked if the MCU only has access 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 capacity by both the MCU and each participating video endpoint. Also, 768k, 1472k, and 1536k calls require ISDN-PRI.

Recording Meet-Me Numbers and MCU- Extensions

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

The form should show 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 should also include 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 "Administering MCU-Extensions and Meet-Me Numbers" on page 2-49 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:

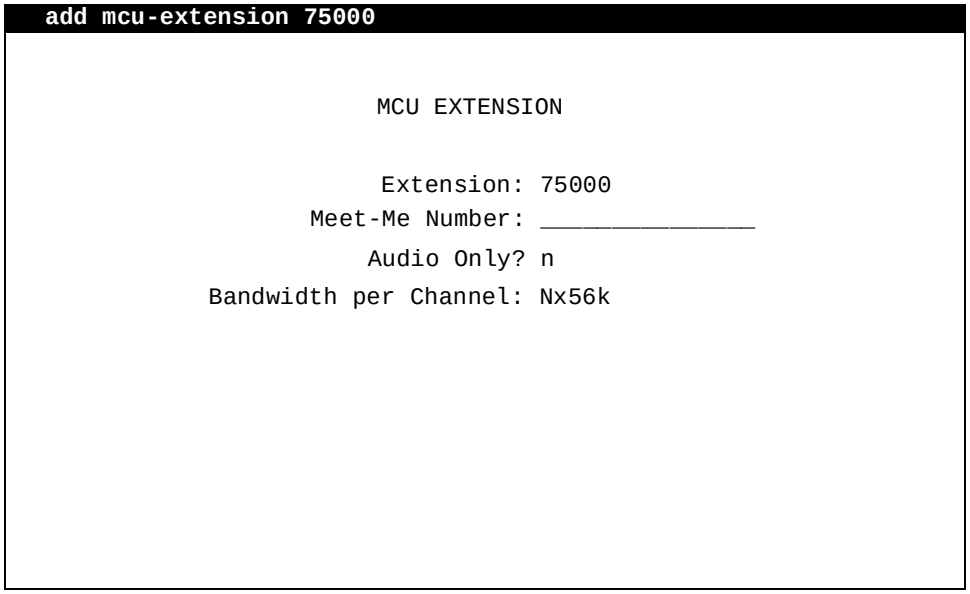

Figure 2-1. MCU-Extension Form for Add/Change/Remove/Display

NOTE:
In the previous figure, the entry “Nx56k,” where “N”=6, in the previous figure is used to accommodate 336k bonding.

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 Only

72009       (700)737-5009         all            n
75000       Port 1                Nx56k          n
75001       (700)737-5001         all            n
75002       (700)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 2-2. MCU-Extension Form for the List Command

Administering International-Based Parameters

You can implement and administer parameters associated with specific international call characteristics by completing the System Parameters Country-Options form. Table 2-3 lists the commands that can access the form.

Table 2-3. 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.

Use the following procedure to administer international-based parameters:

1. At the *command:* prompt, enter **change *system-parameters country-options***.
2. Verify the screen displays the *System Parameters Country-Options Form*.
3. Use TAB and RETURN to advance to the fields you want to change (see "System Parameters Country-Options Form" on page 2-12 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: 10             Version of Digital Loss Plan: B


TONE DETECTION PARAMETERS
 Tone Detection Mode: 4      Dial Tone Validation Time (msec): 600
    Interdigit Pause: short
```

Figure 2-3. 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 2-4. System Parameters Country Options Form (Page 2)

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

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

- *Companding Mode*

  Enter **A-law** (U.K. or Australia) or **Mu-law** ([default] U.S. or Canada) to identify the companding algorithm to be used by system hardware.

- *Base Tone Generation Set*

  Enter a country code (either **1** [default] for the U.S., **2** for Australia, or **10** for the U.K.) to identify the base tone generator set to be used.

- *440Hz MCU-Dial Tone*

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

- *440Hz Secondary-Dial Tone*

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

- *Digital Loss Plan*

  Enter a country code (either **1** [default] for the U.S., **2** for Australia, or **10** for the U.K.) to identify the digital loss plan to be used by the system.

- *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).

The next two fields control tone detection on the system:

- *Tone Detection Mode*

  Enter a code (either **1** [default] for the U.S., **2** for Australia, **3** for the U.K., **4** or **5**) to specify the type of tone detection used on a Tone Detector circuit pack.

- *Dial Tone Validation Timer*

  This popup 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. The default is **600**.

- *Interdigit Pause*

  Enter short (5 to 30 ms) or long (20 to 20 ms) to specify the maximum length of the inter-digit 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
  — call-wait-ringback — conference
  — confirmation — disable-dial
  — hold — intercept
  — intrusion — mcu-dial
  — recall-dial — reorder
  — rep-confirmation — ringback
  — secondary-dial

- *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.

- 350/-17.25
- 350+425/-4.0
- 350+440/-13.75
- 375+425/-15.0
- 404/-11.0
- 404/-16.0
- 404+425/-11.0
- 404+450/-11.0
- 425/-4.0
- 425/-11.0
- 425/-17.25
- 440/-17.25
- 440+480/-19.0
- 480/-17.25
- 480+620/-24.0
- 525/-11.0
- 620/-17.25
- 697/-8.5
- 770/-8.5
- 852/-8.5
- 941/-8.5
- 1000/0.0
- 1000/+3.0
- 1004/0.0
- 1004/-16.0
- 1209/-7.5
- 1336/-7.5
- 1400/-11.0
- 1477/-7.5
- 1633/-7.5
- 2025/-12.1
- 2100/-12.1
- 2225/-12.1
- 2804/-16.0
- goto
- silence

Administering Trunks

All video calls connecting to the MCU are transported on 56k or 64k 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, network access facility and provisioning call routing. See the *AT&T MultiPoint Control Unit System Description* for details.

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 DS1 or UDS1 trunks. Based on the number of ISDN-PRI trunks configured, ISDN-PRI NFAS is supported domestically, but not in Australia and the United Kingdom, and D-channel backup features are supported both domestically and internationally.

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:
The TN2207 circuit pack does not support robbed bit signaling.

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 utilizing the DS1 (TN767D) 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 TN767D DS1 (TN2207 UDS1) circuit pack must be used in conjunction with a TN765 Processor Interface circuit pack. 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 31 56k or 64k trunks for video transmission and one trunk for signaling.

- **DS1 tie trunks**

  The TN767D 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.

Administering DS1 or UDS1 Circuit Packs

Table 2-4 provides the commands that relate to the DS1 or UDS1 circuit pack.

Table 2-4. DS1 Circuit Pack Administration Commands

Action	Object	Qualifier
add	ds1	location
change	ds1	location
display	ds1	location
list	measurements ds1	location
remove	ds1	location

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

Use the following procedure to administer parameters for the TN767D DS1 or TN2207 UDS1 circuit pack:

1. At the *command:* prompt, enter **add ds1** ***location*** where ***location*** is the location of the TN767D DS1 or TN2207 UDS1 circuit pack in the carrier.
2. Verify the screen displays the *DS1 Circuit Pack Form*.
3. Use TAB and RETURN to advance to the fields you want to change (see "DS1 Circuit Pack Form" on page 2-19 for a description of the fields and values).
4. Submit the form.
5. To verify the form, type **display ds1** ***location***.

DS1 Circuit Pack Form

The DS1 Circuit Pack form is used to administer parameters on the TN767D DS1 or TN2207 UDS1 circuit pack. The circuit pack must be installed before it can be administered. The following list identifies the fields on the DS1 Circuit Pack form and the possible values:

- *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.

- *Bit Rate*

  Enter 1.544 for the 24-channel DS1 or UDS1 rate. Enter 2.048 for the 32-channel DS1 rate.

- *Line Coding*

  This entry indicates which line coding format is used to ensure the data meets T1- or E1-carrier requirements. For all bandwidths of 64k or greater, enter **b8zs** for bipolar eight zero suppression.

  If the MCU is connected to a PBX that does not support B8ZS line coding, administer the DS1 or UDS1 circuit pack to support only 56k calls. For a DS1 circuit pack, enter **zcs** (AMI also known as zero code suppression) in this field; for a UDS1 circuit pack, enter **hdb3**. The selection must match the method used on the other end of the link.

- *Line Compensation*

  Enter the number **1** or **2** as appropriate for the cable length. The cable length is one half the distance from the MCU TN767D DS1 or TN2207 UDS1 circuit pack to the CSU.

  1 = 000 to133 feet

  2 = 133 to 266 feet

- *Framing Mode*

  The network diagram 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.**

- *Signaling Mode*

  Enter **robbed-bit, isdn-pri** or **isdn-ext,** or **CAS** to match the method used on the other end of the link. For 56k, enter **robbed-bit**. For ISDN signaling, use **isdn-pri** if the DS1 trunk contains a D-channel or enter **isdn-ext** if the D-channel is external (this is possible if NFAS is being used). CAS signaling can be administered for the TN2207 circuit pack with bit rate 2.048 and with country code **2** (Australia), **4** (5ESS) or **10** (United Kingdom). CAS signaling is the common channel signaling for E1.

- *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**.

- *Interface*

  This field appears only when the *Connect* field value is **pbx**. **Network** or **user** are the appropriate values. Selecting **network** sets the MCU to use the network end of the protocol. Entering **user** will set the MCU to use the protocol of the opposite end.

- MMI Cabling Board

  This field displays the code for the MMI cabling board (for example, **1A10**).

- Country Protocol

  This field appears if **isdn-pri** or **CAS** appears for *Signaling Mode*. Value **1** (U.S.) is the default value. Other current reserved values are **2** (Australia), **4** (5ESS), and **10** (United Kingdom).

- CRC

  Enter **y** to indicate that a cyclic redundancy check is to be performed on trasmissions that the board 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.

- *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).

- *Remote Loop-Around Test*

  Enter **y** to allow testing when the DS1 or UDS1 is connected to appropriate equipment. The response should be **n** unless testing is extremely important.

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. 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 facilities. Loss of the reference signal, excessive phase changes, and variation in timing frequency can cause loss of data or dropped data connections.

Table 2-5 provides the commands that relate to the synchronization plan.

Table 2-5. 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 the screen displays the *Synchronization Plan* form.
3. Use TAB and RETURN to advance to the fields you want to change (see "Synchronization Plan Form" on page 2-21 for a list of fields and values).
4. Submit the form.
5. To verify the form, enter **display synchronization**.

Synchronization Plan Form

The following list provides the fields on the Synchronization Plan form and the related values:

- *Stratum*

  Enter the stratum level being used; **4** for Stratum 4 is the default.

  NOTE:
  Stratum 3 is not supported with the MCU.

- *Primary*

  Specify the four-character DS1 circuit pack location for the DS1 circuit pack that will function as the primary synchronization source.

- *Secondary*

  Specify the four-character DS1 circuit pack location for the DS1 circuit pack that will function as the secondary circuit pack if the primary circuit pack has a failure.

- *Location*

  Displays the location of each DS1 circuit pack administered on the DS1 Circuit Pack form.

- *Name*

  Displays the name of each DS1 circuit pack administered on the DS1 Circuit Pack form.

- *Slip*

  Displays the slip detection parameter for each circuit pack administered on the DS1 Circuit Pack form.

NOTE:
The DS1 Circuit Pack location, name and slip display continues onto page 2 if necessary.

Administering ISDN-PRI 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.

Processor Interface Data Module

The processor interface data modules are integrated into the TN765 Processor Interface circuit pack to provide the ISDN-PRI interface. Connection to this interface is provided via a DS1 interface. One direct EIA connection is available. When used, the physical channel assignment is **01**.

Table 2-6 gives the commands that relate to the data modules.

Table 2-6. 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

Use the following procedure to add a data module to provide an ISDN-PRI interface:

1. At the *command:* prompt, enter **add data-module *xxxx***.
2. Verify the screen displays the *Data Module* form (see "Processor Interface Data Module Form" on page 2-24 for a list of the fields and their 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 data-module *xxxx***.

Processor Interface Data Module Form

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.

- *Type*

  Enter **procr-intf**.

- *Physical Channel*

  Enter the two-digit circuit number of the processor interface port (**01** through **04** for the ESCC, **01** through **08** for the MCC).

- *Name*

  Optional field that identifies the name of the system such as MCU.

- *COS*

  Enter the desired Class of Service. The default is **1**.

- *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.

Processor Interface Links

The Processor Interface Links form identifies, describes, and enables processor interface links for ISDN-PRI between the MCU and ISDN. When you are adding an ISDN-PRI trunk, you begin by setting the *Enable* field to **n** to disable the link until you complete the administration of the trunk. You then administer the processor channels and signaling group before coming back to this form to enable the links (use **change communication-interface links** then enter **y** in the *Enable* field).

Table 2-7 provides the commands that relate to the processor interface link.

Table 2-7. Processor Interface Links Administration Commands

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

Use the following procedure to change the Processor Interface Links form:

1. At the *command:* prompt, enter **change communication-interface links**.
2. Verify the screen displays the *Processor Interface Links* form (see "Interface Links Form" on page 2-26 for a list of the fields and their 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 communication-interface links**.

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 another node in the ISDN network.

- *Enable*

  Enter **y** to enable the link.

  NOTE:
  All fields on this form may be initially entered except the *Enable* field which must be set to **n**. After administering the processor channels and signaling groups, you can enter **y** in this field.

- *Est Conn*

  Enter **y** is 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.

- *Brd*

  Enter the TN767D DS1 or TN2207 UDS1 circuit pack address that has been administered as using ISDN signaling.

- *DTE/DCE*

  No entry is required. Use to specify when the connection is to a DTE or a DCE.

- *Identification*

  Enter a name for the link (up to 15 characters).

Processor Channel Assignments

The Processor Channel Assignments form assigns channels to the MCU interface links.

Table 2-8 lists the commands that relate to the processor channels.

Table 2-8. Processor Channels Administration Commands

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

Use the following procedure to change the Processor Channels form:

1. At the *command:* prompt, enter **change communication-interface processor-channels**.
2. Verify the screen displays the *Processor Channels* form (see "Processor Channels Form" on page 2-27 for a list of the fields and their 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 communication-interface-processor channels**.

Processor Channels Form

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

- *Proc Chan*

  This display-only field indicates the 64 channels that are available.

- *Appl*

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

- *Link*

  Enter a number (**1** through **4** for ESCC; **1** through **8** for MCC) to identify the interface link/channel pair used to establish a connection.

- *Chal*

  This field, along with the *Link* field identifies the link and channel pair that is associated with the local processor channel. This field should remain blank.

- *Priority*

  Enter **h** for high or **l** for low to indicate the priority of this channel. Assignments should be made based on the operational speed of the links.

- *Remote Proc Chan*

  Leave this field blank.

- *Machine ID*

  Leave this field blank.

Administering Signaling Groups

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. The NFAS feature 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 Australia and the United Kingdom).

The Signaling Group form is administered to show how B-channels and D-channels are related. Table 2-9 lists the commands that can be used with the Signaling Group form.

Table 2-9. Signaling Group Commands

Action	Object	Qualifier
add	signaling-group	1-8 or next
change	signaling-group	1-8
display	signaling-group	1-8, print, or schedule
list	signaling-group	1-8, print, or schedule
remove	signaling-group	1-8

Use the following procedure to specify a signaling group:

1. At the *command:* prompt, enter **add signaling-group *x*** where ***x*** is the signaling group number (**1** through **8**) or **next**.
2. Verify the screen displays the *Signaling Group* form (see "Signaling Group Form" on page 2-29 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 signaling-group *x***.

Signaling Group Form

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 above commands.

- *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 Australia and the United Kingdom).

- *Max Number of NCA TSC*

  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.

- *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.

- *Trunk Group for NCA TSC*

  This field is not used.

- *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.

- *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 0, *Trunk Brd* 1 which should be the primary D-channel. If a secondary D-channel is used, it should be *Trunk Brd* 2 ID 1.

NOTE:
Pages 2 through 5 are not used.

NOTE:
Once you administer the processor channels and signaling groups, enter **y** into the *Enable* field of the Interface Links form.

Administering Trunk Groups

Use the following procedures to administer ISDN-PRI and tie trunks for the MCU. Table 2-10 shows the commands that are used to administer trunk groups using the Trunk Group form.

Table 2-10. ISDN-PRI Trunk Group Commands

Action	Object	Qualifier
add	trunk-group	1-16 (next)
change	trunk-group	1-16 or TAC X
display	trunk-group	1-16 or TAC X, print, or schedule
list	trunk-group	print or schedule
remove	trunk-group	1-16 or TAC X
status	trunk-group	1-16 or TAC X

Use the following procedure to add a ISDN or tie trunk group:

1. At the *command:* prompt, enter **add trunk group *x*** where ***x*** is a trunk group number (**1** through **99**) or **next**.
2. Verify the screen displays the Trunk Group form (see "Administering ISDN-PRI Trunk Groups" on page 2-31 or "Administering Tie Trunk Groups" on page 2-35 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 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:

Trunk Group Form, Page 1 of 10

- *Group Number*

  This display-only field shows the trunk group number.

- *Group Type*

  Enter **isdn** to identify the trunk group.

- *SMDR Reports*

  Enter **y** to track incoming calls made on all trunks in the trunk group.

- *Group Name*

  Enter a name up to 15 characters to describe this trunk group. The default is **OUTSIDE CALL.**

- *COR*

  Leave at the default of **0**.

- *TAC*

  Enter the trunk access code that must be dialed to access the trunk group. A different TAC must be assigned to each trunk group. SMDR uses the TAC to identify the trunk group on Call Detail Record (CDR) reports. Allowable entries must match the dial plan.

- *Direction*

  This display-only field is set to **incoming, outgoing,** or **two-way**.

- *Outgoing Display*

  If some calls are outgoing calls, this field should be set to **y**. Otherwise, the field should be set to **n**.

- *Dial Access*

  This field can be set to either **y** or **n**. The entry **y** enables trunk dialing instead of ARS dialing.

  NOTE:
  Setting this field to **y** may cause toll fraud.

- *Service Type*

  Indicates the service for which this ISDN-PRI trunk group will be dedicated. The valid entries include **accunet**, **cbc**, **public-ntwk**, **sddn**, **sdn**, or **tie** and any user-defined or selected service as administered on the Network Facilities form.

  NOTE:
  The service type selected must match whatever is administered at the other end.

- *Test Call ITC*

  Leave at the default of **rest** for restricted.

- *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. For example, for an ISDN tandem trunk, the far-end test number must be a 7-digit Electronic Tandem Network (ETN) number. Up to 15 digits may be entered in this field.

- *TestCall BCC*

  Leave this field at its default setting of **4** for DCP Mode 0.

- *Display, TCM, Lookahead Codeset*

  This field is not used.

- *Max Message Size to Send*

  Leave this field set at the default of **260** byte messages.

- *Trunk Hunt*

  For 384k bandwidth, set this field to **ascend** or **descend** as provisioned on the network.

- *Connected To*

  This field is defaulted to **main**.

- *Connected to Toll*

  This field is set to **n**.

- *DTT to DCO Loss*

  Leave this field set at the default of **normal**.

- *Synchronization*

  This field is not used.

- *Duplex*

  This field is not used.

Trunk Group Form, Page 2 of 10

- *ACA Assignment*

  Indicates whether or not Automatic Circuit Assurance (ACA) measurements are taken for this trunk group. The default is **n**. When ACA measurements are enabled, three related fields are shown where you need to specify the *Long Holding Time* (in hours), *Short Holding Time* (in seconds) and *Short Holding Time Threshold* (number of short holding time occurrences before alarming). ACA must also be enabled on the Feature-Related System Parameters form (see "Automatic Circuit Assurance Reports" on page 3-12 in Chapter 3).

  NOTE:
  Since video conference calls generally last several hours, be sure to set the *Long Holding Time* high enough to accommodate the average length (up to 10 hours). Descriptions of long holding time, short holding time and short holding time threshold are found under "Automatic Circuit Assurance Reports" on page 3-12 in Chapter 3.

- *Wideband Support*

  Indicates whether or not Wideband is supported on this trunk group. For 384k 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**. if this field is set to **y**, it prevents generation of tones on a data call that cause erroneous data transmission.

- *NCA-TSC Signaling Group*

  This field is not used.

- *Send CPN*

  This field is not used.

- *Send Name*

  This field is not used.

- Maintenance Tests

  Leave this field in its default setting of **y**.

Trunk Group Form, Page 3 of 10

- *Wideband Support Options*

  For 384k bandwidth and service providers supporting NxDS0, enter **y** for **H0**. For 1536k bandwidth, enter **y** for **H11**.

- *Contiguous*

  This field should be set to **y**.

Trunk Group Form, Page 4 of 10

- *Port*

  Enter the trunk port number (up to six characters).

  NOTE:
  Use **list configuration all** to determine the trunk port number.

- *Name*

  Enter the name of the member of the trunk group (up to 10 characters). The name can be the seven-digit telephone number or a 10-digit trunk circuit identification number (provided by the trunk supplier).

- *Sig Grp*

  Enter the signaling group number (**1** through **8**). 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. If a DS1 circuit pack appears in more than one signaling group, this field must be completed manually.

  NOTE:
  The fields and values of page 4 are continued on pages 5 through 10.

Administering Tie Trunk Groups

The following list identifies the fields on the Trunk Group form and their values entered when administering tie trunk groups:

Trunk Group Form, Page 1 of 9

- *Group Number*

  This is a display-only field shows the trunk group number.

- *Group Type*

  Enter **tie** to identify the trunk group.

- *SMDR Reports*

  Enter **y** to track incoming calls made on all trunks in the trunk group.

- *Group Name*

  Enter a name up to 15 characters to describe this trunk group. The default is **OUTSIDE CALL.**

- *COR*

  Leave this field at the default of **0**.

- *TAC*

  Enter the Trunk Access Code (TAC) that must be dialed to access the trunk group. A different TAC must be assigned to each trunk group. Station Message Detail Recording (SMDR) uses the TAC to identify the trunk group on Call Detail Record (CDR) reports. Allowable entries must match the dial plan.

- *Direction*

  This display-only field is set to **incoming, outgoing,** or **two-way.**

- *Outgoing Display*

  If some calls are outgoing calls, this field should be set to **y**. Otherwise, the field should be set to **n**.

- *Dial Access*

  Leave this field set to **n**.

- *Comm Type*

  This field is set either to **rbavd** for Robbed Bit Audio Voice and Data or to **avd** for DAS DS1 trunks. To use the latter entry, the DS1 signal mode must be set to **CAS**.

- *BCC*

  This field indicates barrier capabilities. Possible entries include **0** (voice), **1** (56k robbed-bit), **2** (up to 19.2 [modem is set up]), **3** (restricted 64k), or **4** (64k clear [CAS]).

- *ITC*

  Leave this field at the default setting of **rest** for restricted.

- Trunk Type (in/out):

  Specify the type of trunk. The default is **wink/wink**.

- *Incoming Rotary Timeout (sec)*

  Specify the number of seconds before an incoming rotary call is timed out. The default is **5**.

- *Outgoing Dial Type*

  This field should be set to **tone**.

- *Incoming Dial Type*

  This field should be set to **tone.**

- *Disconnect Timing*

  Specify in milliseconds the disconnect time. The default is **500** milliseconds.

- *Digits*

  Leave this field blank.

- *Digit Treatment*

  If inserting or deleting digits at the trunk group level, set this field to either **insert** or **delete** as appropriate. If **insert** is selected, you must enter the digits to be inserted.

- *DTT to DCO Loss*

  This field should be left at the default of **normal**.

- *Connected to Toll*

  Leave this field set to **n**.

- *Incoming Dial Tone*

  Leave this field set to **y**.

- *Synchronization*

  This field is not used.

- *Duplex*

  This field is not used.

Trunk Group Form, Page 2 of 9

- *ACA Assignment*

  Indicates whether or not Automatic Circuit Assurance (ACA) measurements are taken for this trunk group. The default is **y**. When ACA measurements are enabled, three related fields are shown where you need to specify the *Long Holding Time* (in hours), *Short Holding Time* (in seconds) and *Short Holding Time Threshold* (number of short holding time occurrences before alarming). ACA must also be enabled on the Features-Related System Parameters form (see "Automatic Circuit Assurance Reports" on page 3-12 in Chapter 3).

  NOTE:
  Since video conference calls generally last several hours, be sure to set the *Long Holding Time* high enough to accommodate the average length (up to 10 hours). Descriptions of long holding time, short holding time and short holding time threshold are found under "Automatic Circuit Assurance Reports" on page 3-12 in Chapter 3.

- *Data Restriction*

  Leave this field set to **n**. When set to **y** this prevents generation of tones on a data call that cause erroneous data transmission.

- *Suppress # Outpulsing*

  Leave this field set to **n**.

- *Maintenance Tests*

  Leave this field set to **n**.

Trunk Group Form, Page 3 of 9

- *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-digit telephone number or a 10-digit trunk circuit identification number (provided by the trunk supplier).

- *Sig Grp*

  Enter the signaling group number (**1** through **8**). 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. If a DS1 circuit pack appears in more than one signaling group, then this field must be completed manually.

  NOTE:
  The fields and values on page 3 are continued on pages 4 to 9.

Administering Maintenance Alarm Terminals

Use the Station form to assign alarm buttons to the maintenance alarm terminal. One maintenance alarm terminal (model 7444D) is supplied with the MCU; up to one additional maintenance alarm terminal is supported.

The following commands can be used to administer the maintenance alarm terminal.

Table 2-11. 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

Use the following procedure to add a maintenance alarm terminal:

1. At the *command:* prompt, enter **add station *x*xxx** where ***xxxx*** is an extension number or **next**.
2. Verify the screen displays the *Station* form (see "Station Form" on page 2-39 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

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

- *Extension*

  A display-only field that shows the extension number of the maintenance alarm terminal being administered.

- *Type*

  This field shows the model alarm terminal assigned to this extension.

  NOTE:
  The 7444D and 7444D-33 are the only type of alarm terminals supported by the MCU.

- *COR*

  The Class of Restriction assigned to this maintenance alarm terminal. This should be assigned to the same COR as any other maintenance alarm terminal.

- *Port*

  Specify the MCU port that is associated with this maintenance alarm terminal. This port is located on a TN754 Digital Line circuit pack.

- *COS*

  The Class of Service assigned to this maintenance alarm terminal should be the same as the COS assigned to any other maintenance alarm terminal.

- *Name*

  A label that identifies this maintenance alarm terminal.

- *Data Module*

  When **y** is entered, a 7400B data module must be administered for this maintenance alarm terminal.

- *Display Module*

  Set to **y** if the maintenance alarm terminal has a display (this is the correct setting for the 7444D or 7444D-33).

Station Form, Page 2 of 5

- *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 maintenance alarm 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. Table 2-12 lists the recommended maintenance alarm terminal button assignments for buttons 1 through 10.

Table 2-12. Maintenance Alarm Terminal Button Assignments (1 through 10)

Button	Assignment	Button	Assignment
1	call-appr	6	ac-alarm
2	call-appr	7	smdr1-alrm
3	call-appr	8	pr-sys-alm
4	lsvn-call	9	trk-ac-alm
5	aca-call	10	verify

Station Form, Page 3 of 5

- *Button Assignments*

  Use this continuation of screen 2 to assign features to the appropriate buttons. Table 2-13 lists the recommended maintenance alarm terminal button assignments for buttons 11 through 34.

Table 2-13. Maintenance Alarm 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	

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- *Display Button Assignments*

  Assign the display button appearances to the appropriate buttons as shown in Table 2-14.

Table 2-14. 7444D Display Button Assignments

Button	Assignment
1	normal
2	inspect
3	trk-id
4	trunk-name

Table 2-14. 7444D Display Button Assignments

Button	Assignment
5	date-time
6	timer
7	

Station Form, Page 5 of 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 end-point's ITC. The 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 Dial Plans

To accommodate the dial-out feature, you should administer the dial plan(s) appropriate for your site. These plans include 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 you can administer the forms necessary to enable AAR, either the Private Networking or Uniform Dialing Plan option on the System-Parameters Customer-Options form must be enabled.

Use the following procedures to administer AAR:

1. At the prompt, enter *display dialplan* to access the Dial Plan Form.
2. Populate the *Area Code* field.
3. Enter "fac" (feature access code) next to *8:* in the column headed by *First Digit. First Digit Table* in the form defines the MCU dial plan.
4. Submit the Dial Plan Form.
5. At the prompt, enter *change feature-access-codes* to display the Feature Access Code Form.
6. On Page 1 of the form, enter "8" in the *Auto Alternate Routing (AAR) Access Code* field.
7. Submit the form.
8. At the prompt, enter *change aar analysis* to access the AAR Analysis Table. This form is used to associate dialed or modified strings to routing patterns.
9. In the first available space in the column headed by *Dialed String,* enter a number of up to 18 digits (for example, "557").
10. In the corresponding space within the column headed by *Min,* enter the minimum number of digits needed to validate and route the call (for example, "7").
11. In the corresponding space within the column headed by *Max,* enter the maximum number of digits that can be used to validate and route the call (for example, "7").
12. In the corresponding space within the column headed by *Call Type*, enter "ars/aar."
13. Submit the form.

14. At the prompt, type **change route-pattern** *x*, where *x* is the routing pattern number you want to access. In step 12, we used “11” as a routing pattern. Therefore, replace *x* with “11.” The *Pattern Number: 11* screen appears.
15. In the first available space within the column headed by *Grp. No.,* enter the desired trunk group number to specify which trunk will be asssociated with the entries on the same line of the form.
16. In the corresponding space within the column headed by FRL, enter the Facility Restricition Level (FRL) to be assigned to the line of parameters on the form. Choices range from “0” (least restrictive) to “7” (most restricitive).
17. In the corresponding space within the column headed by *No. Del. Digits,* enter the total number of digits to be deleted from the dialed string when the corresponding trunk group is selected for call routing. The default is blank.
18. In the corresponding space within the column headed by *Inserted Digits*, enter the actual digits to be inserted. Up to 52 digits may be outpulsed.
19. For a 384k conference, ensure that “y” appears in the appropriate space within the *W* (Wideband) column under *BCC Value*. The BCC value is used to determine compatibility when non-ISDN-PRI facilities are connected to ISDN facilities.
20. In the corresponding space within the column headed by *ITC* (Information Transfer Capability), administer the type of traffic (either restricted, unrestricted, or both) that is allowed to be carried by this routing preference. “Rest” (restricted) allows calls that are originating only from restricted endpoints to access the routing preference. “Unre” (unrestricted) allows calls that are originating only from unrestricted endpoints to access the routing preference. “Both” allows calls originating from both restricted and unrestricted endpoints to access the routing preference.
21. If “both” is administered in the previous field, create the Information Transfer Capability codepoint in the Bearer Capability IE of the SETUP message. Do this in the corresponding space within the column headed by *BCIE* (Bearer Capability Information Element).
22. Submit the form.

Automatic Route Selection (ARS)

Automatic Route Selection (ARS) allows you to route calls over the public network according to the preferred (usually the least expensive) route available at the time you place the call. Before you can administer the forms necessary to enable ARS, the ARS option on the System-Parameters Customer-Options form must be enabled.

Use the following procedures to administer ARS:

1. At the prompt, enter *display dialplan* to access the Dial Plan Form.
2. Populate the *Area Code* field.
3. Enter "fac" (feature access code) next to *9:* in the column headed by *First Digit. First Digit Table* in the form defines the PBX dial plan.
4. Submit the Dial Plan Form.
5. At the prompt, enter *display feature-access-codes* to display the Feature Access Code Form.
6. On Page 1 of the form, enter "9" in the *Auto Route Selection (ARS) - Access Code 1*: field.
7. Submit the form.
8. At the prompt, enter *change ars analysis* to access the ARS Digit Analysis Table. This form is used to associate dialed or modified strings to routing patterns.
9. In the first available space in the column headed by *Dialed String,* enter a number of up to 18 digits (for example, "700").
10. In the corresponding space within the column headed by *Min,* enter the minimum number of digits needed to validate and route the call (for example, "10").
11. In the corresponding space within the column headed by *Max,* enter the maximum number of digits that can be used to validate and route the call (for example, "10").
12. In the corresponding space within the column headed by *Route Pat*, enter the routing pattern number (choose from 1 through 254) on a pointer to an associated Remote Home Number Plan Area (RHNPA) form to be used to route the call once all identifying digits have been received (for example, "7").
13. In the corresponding space within the column headed by *Call Type*, enter "fnpa."
14. Submit the form.

15. At the prompt, type **change route-pattern** *x*, where *x* is the routing pattern number you want to access. In step 12, we used “7” as a routing pattern. Therefore, replace *x* with “7.” The *Pattern Number: 7* screen appears.
16. In the first available space within the column headed by *Grp. No.,* enter the desired trunk group number to specify which trunk will be asssociated with the entries on the same line of the form.
17. In the corresponding space within the column headed by FRL, enter the Facility Restricition Level (FRL) to be assigned to the line of parameters on the form. Choices range from “0” (least restrictive) to “7” (most restricitive).
18. In the corresponding space within the column headed by *NPA,* enter the Numbering Plan Area (NPA) of the distant end.
19. In the corresponding space within the column headed by Prefix Mark, enter the appropriate number (0 through 4) to indicate the case(s) in which “1” is outpulsed before the dialed string.
20. In the corresponding space within the column headed by *Toll List*, enter a number that references the ARS Toll Table associated with the terminating NPA of the trunk group.
21. In the corresponding space within the column headed by *No. Del. Digits,* enter the total number of digits to be deleted from the dialed string when the corresponding trunk group is selected for call routing. The default is blank.
22. In the corresponding space within the column headed by *Inserted Digits*, enter the actual digits to be inserted. Up to 52 digits may be outpulsed.
23. For a 384k conference, ensure that “y” appears in the appropriate space within the *W* (Wideband) column under *BCC Value*. The BCC value is used to determine compatibility when non-ISDN-PRI facilities are connected to ISDN facilities.
24. In the corresponding space within the column headed by *ITC* (Information Transfer Capability), administer the type of traffic (either restricted, unrestricted, or both) that is allowed to be carried by this routing preference. “Rest” (restricted) allows calls that are originating only from restricted endpoints to access the routing preference. “Unre” (unrestricted) allows calls that are originating only from unrestricted endpoints to access the routing preference. “Both” allows calls originating from both restricted and unrestricted endpoints to access the routing preference.
25. If “both” is administered in the previous field, create the Information Transfer Capability codepoint in the Bearer Capability IE of the SETUP message. Do this in the corresponding space within the column headed by *BCIE* (Bearer Capability Information Element).
26. Submit the form.

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), Main/Satellite/Tributary configurations, and Distributed Communications System (DCS) 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. Before you can administer the forms to enable UDP, the optional UDP feature must be enabled on the System-Parameters Customer-Options form.

Use the following procedures to administer UDP:

1. At the prompt, enter *change dialplan* to access the Dial Plan Form.
2. On Page 1, enter “y” in the the *Uniform Dialing Plan?* field.
3. Per your requirements, enter “4” or “5” in the *Plan Length* field.
4. Go to Page 2.
5. In the first available space within the column headed by *CODE*, enter at least the first digit of the UDP number (for example, “3”). If you want to restrict the destination even further, add another digit or two.
6. Insert “n” in the corresponding space within the column headed by *LCL* (Local).
7. In the corresponding space within the column headed by *RNX*, enter the MCU-code for the destination MCU (for example, “472”). Ensure that the first digit in this code is not “0.”
8. Submit the Dial Plan Form.
9. At the prompt, enter *change aar analysis* to access the AAR Digit Analysis Table. This form is used to associate dialed or modified strings to routing patterns.
10. In the first available space in the column headed by *Dialed String,* enter the PBX-code created in the Dial Plan Form (for example, “472”).
11. In the corresponding space within the column headed by *Min,* enter the minimum number of digits needed to validate and route the call (for example, “10”).
12. In the corresponding space within the column headed by *Max,* enter the maximum number of digits that can be used to validate and route the call (for example, “10”).

13. In the corresponding space within the column headed by *Route Pat*, enter the routing pattern number (choose from 1 through 254) on a pointer to an associated Remote Home Number Plan Area (RHNPA) form to be used to route the call once all identifying digits have been received (for example, “7”).
14. In the corresponding space within the column headed by *Call Type*, enter “fnpa.”
15. Submit the form.
16. At the prompt, type **change route-pattern** *x*, where *x* is the routing pattern number you want to access. In step 12, we used “7” as a routing pattern. Therefore, replace *x* with “7.” The *Pattern Number: 7* screen appears.
17. In the first available space within the column headed by *Grp. No.,* enter the desired trunk group number to specify which trunk will be asssociated with the entries on the same line of the form.
18. In the corresponding space within the column headed by FRL, enter the Facility Restricition Level (FRL) to be assigned to the line of parameters on the form. Choices range from “0” (least restrictive) to “7” (most restricitive).
19. In the corresponding space within the column headed by *No. Del. Digits,* enter the total number of digits to be deleted from the dialed string when the corresponding trunk group is selected for call routing. The default is blank.
20. In the corresponding space within the column headed by *Inserted Digits*, enter the actual digits to be inserted. Up to 52 digits may be outpulsed.
21. For a 384k conference, ensure that “y” appears in the appropriate space within the *W* (Wideband) column under *BCC Value*. The BCC value is used to determine compatibility when non-ISDN-PRI facilities are connected to ISDN facilities.
22. In the corresponding space within the column headed by *ITC* (Information Transfer Capability), administer the type of traffic (either restricted, unrestricted, or both) that is allowed to be carried by this routing preference. “Rest” (restricted) allows calls that are originating only from restricted endpoints to access the routing preference. “Unre” (unrestricted) allows calls that are originating only from unrestricted endpoints to access the routing preference. “Both” allows calls originating from both restricted and unrestricted endpoints to access the routing preference.
23. If “both” is administered in the previous field, create the Information Transfer Capability codepoint in the Bearer Capability IE of the SETUP message. Do this in the corresponding space within the column headed by *BCIE* (Bearer Capability Information Element).
24. Submit the form.

Administering MCU-Extensions and Meet-Me Numbers

Calls gain access to the MCU via DS1 trunks. Once the network delivers a call across the DS1 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).

The number of MCU ports is determined by the software administration of the1B, 2B, H0, 768k, 1472k, and 1536k 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.

A 56k MCU-extension can be assigned to a 64k conference via rate adaptation. 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.

Network Services

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.

Table 2-15 provides an overview of the most commonly used digital services provided by AT&T.

Table 2-15. AT&T 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 56k, 64k, 384k 56k, 64k, 384k	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	56k, 64k, 384k	ISDN-PRI
Software Defined Data Network (SDDN)	NNN-NNNN Nxx-xxxx N0/1x-Nxx-xxxx	56 56k, 64k, 384k 56k, 64k, 384k	RBS network access ISDN-PRI ISDN-PRI
SDDN-Virtual Off-Net	700-56x-xxxx 700-737-xxxx	56k 56k, 64k, 384k	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

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 MCU is preset with a 5-digit dial plan. Each video port is identified by a unique 5-digit extension.

The commands in Table 2-16 can be used to access the Dial Plan Record form.

Table 2-16. Dial Plan Administration Commands

Action	Object	Qualifier
change	dialplan	
display	dialplan	print or schedule

Use the following procedure to change the dial plan:

1. At the *command:* prompt, enter **change dialplan**.
2. Verify the screen displays the *Dial Plan Record* form (see "Dial Plan Form" on page 2-51 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 dialplan**.

Dial Plan Form

The following list shows the fields on the Dial Plan Record and related values.

Dial Plan Form, Page 1 of 6

- *Area Code*

  Enter the area code number where the MCU is located.

- *Local PBX*

  Reserved for future use.

- *ARS Prefix 1 Required*

  Enter **y** if a prefix of "1" is required.

- *Uniform Dialing Plan*

  If UDP is administered, enter **y.** UDP provides a common four or five digit dial plan that can be shared among the MCU and a PBX.

- *Plan Length*

  When UDP is administered, enter the number of digits in the dial plan (**4** or **5**).

- *First Digit Table*

  This table consists of fields for each possible first digit and dialed number length. If the first digit of an extension number is **3** and the length of the extension number is **5** digits, enter **extension** in the third row, fifth column. Likewise, if trunk access codes (TAC) begin with a **1** and are **3** digits long, enter **tac** in the first row, third column.

Dial Plan Form, Page 2 through 6 of 6

- *Code*

  Enter the number representing the first one, two, three, or four digits of a 4- or 5-digit extension.

- *LCL*

  Enter **n** if the code above is located on a remote PBX.

- *RNX*

  When UDP is selected, the code yields the associated RNX which is used to select a Routing Pattern for the call.

- *ID*

  Leave this field blank.

Assigning MCU-Extensions

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. You obtain network numbers from Network Services or, if the MCU is an SDDN or private network location, the Meet-Me numbers can be selected and assigned per your existing dial plan. After you obtain the Meet-Me numbers, you need to associate each number 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**.

Table 2-17 provides the commands that can be used to administer MCU-extensions.

Table 2-17. Meet-Me Extension Administration Commands

Action	Object	Qualifier
add	mcu-extension	extension or next
change	mcu-extension	extension
display	mcu-extension	extension, count, print, or schedule
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.

Use the following procedure to add an MCU-extension:

1. At the *command:* prompt, enter **add mcu *xxxx*** where ***xxxx*** is an extension number or **next**.
2. Verify the screen displays the *MCU-Extension* form (see "MCU-Extension Form" on page 2-54 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 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.”

MCU-Extension Form

The following list shows the fields on the MCU-Extension form and their related values.

- *Extension*

  A display-only field that shows the MCU-extension number.

- *Meet-Me Number*

  Enter the network number (the number the conferee dials to join a conference) that corresponds to this MCU-extension.

  NOTE:
  There is no check that the correct network number is administered in the *Meet-Me Number* field**.** Accordingly, any alphanumeric string of characters is accepted.

- *Audio-Only*

  Enter **yes** or **no** (default).

- *Bandwidth per Channel*

  Enter the bandwidth the Meet-Me Number supports.

  NOTE:
  If the Meet-Me Number supports all the bandwidths, enter **all.**

  NOTE:
  If *Audio-Only* (previous field) is **yes**, the bandwidth does not appear.

Administering Cascades

Cascading is the linking of two MCUs to allow both MCUs to participate in the same conference. You can program one MCU 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, you must establish one MCU as the dial-out MCU and the other MCU as the dial-in MCU. You must also establish one MCU as the primary MCU and the other MCU as the secondary MCU. The appropriate procedures are discussed in the following sections.

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.

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 in order 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. You can establish both MCU types 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, type “CAS” (cascade) in the first available space within the column headed by **Type**.
3. In the corresponding space within the column headed by **Dial-Out Number 1**, type a number that reaches the dial-in MCU. If this is a 2B conference, type the same number in the corresponding space within the column headed by **Dial-Out Number 2.**

The following figure is an example of Page 2 of a completed Conference Record form for a dial-out MCU:

```
add conference 123                                    Page  2 of  3
                              CONFERENCE RECORD

CONFEREES
                MCU
   Type         Ext    Meet-Me Number  Dial-Out Number 1   Dial-Out Number 2

 1: CAS          5000                    705-555-1234        705-555-1234
 2: P64          5001                    ____________        ____________
 3: P64          5002                    ____________        ____________
 4: P64          5003                    ____________        ____________
 5: ____         _____                   ____________        ____________
 6: ____         _____                   ____________        ____________
 7: ____         _____                   ____________        ____________
 8: ____         _____                   ____________        ____________
 9: ____         _____                   ____________        ____________
10: ____         _____                   ____________        ____________
11: ____         _____                   ____________        ____________
12: ____         _____                   ____________        ____________
13: ____         _____                   ____________        ____________
14: ____         _____                   ____________        ____________
15: ____         _____                   ____________        ____________
```

Figure 2-5. 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* column 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 column.

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, type “CAS” (cascade) in the first available space in the column headed by **Type**.

3. In the corresponding space within the column headed by **MCU-EXT**, type the extension (usually the last four digits) of one of the CAS-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:

```
add conference 123                                    Page  2 of  3
                                CONFERENCE RECORD

CONFEREES
                MCU
   Type         Ext    Meet-Me Number  Dial-Out Number 1   Dial-Out Number 2

 1: CAS          1234                   ____________________  ____________________
 2: P64          6001                   ____________________  ____________________
 3: P64          6002                   ____________________  ____________________
 4: P64          6003                   ____________________  ____________________
 5: ____  ______                        ____________________  ____________________
 6: ____  ______                        ____________________  ____________________
 7: ____  ______                        ____________________  ____________________
 8: ____  ______                        ____________________  ____________________
 9: ____  ______                        ____________________  ____________________
10: ____  ______                        ____________________  ____________________
11: ____  ______                        ____________________  ____________________
12: ____  ______                        ____________________  ____________________
13: ____  ______                        ____________________  ____________________
14: ____  ______                        ____________________  ____________________
15: ____  ______                        ____________________  ____________________
```

Figure 2-6. Conference Record Form for a Dial-In MCU (Page 2)

4. Save the Conference Record form.

Establishing the Primary and Secondary MCUs

In a cascaded conference, one MCU is administered as the primary MCU, the other as the secondary MCU. This is done to make the MCU feature interactions as transparent as possible to the user. Generally, you can perform this administration 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, type “primary” in 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, type “secondary” in the **Cascade Mode** field on Page 1, and submit the form.

Procedure for Cascaded Conferences in Broadcast with Auto-Scan or Presentation Mode

You must always ensure that a Px64 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, you must also configure the secondary MCU by accessing the Conference Record form for this MCU, typing the MCU-extension of the cascade link in the **Ctrl Ext** field on Page 1 of the form, and then submitting the form.

Performing System Backups and Saving Translations

The save translation 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.

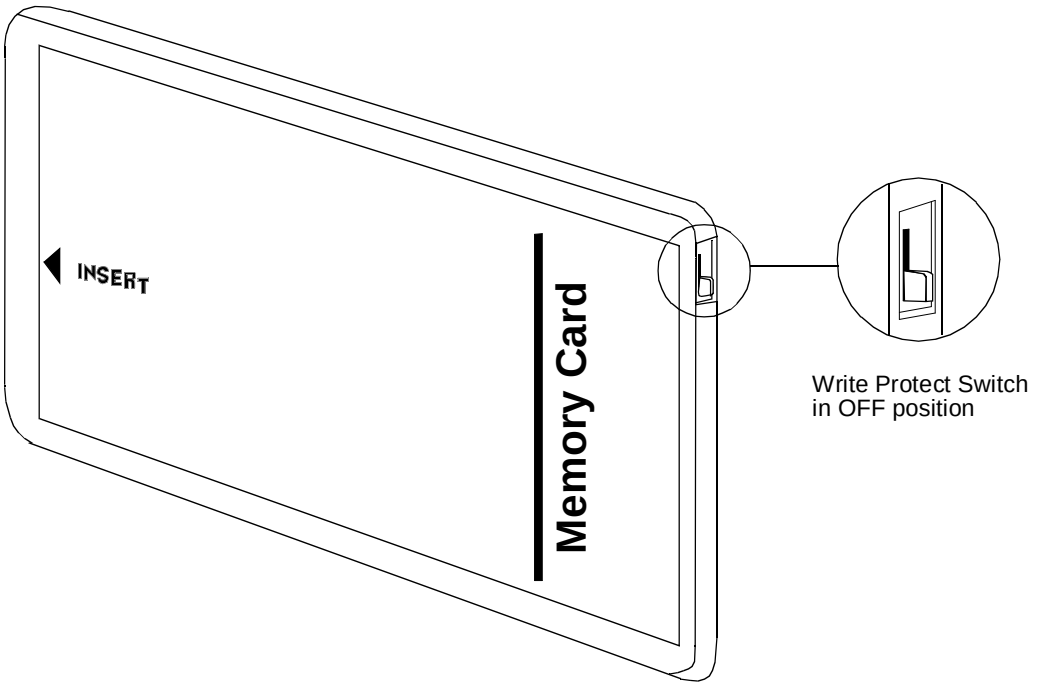

Figure 2-7. Memory Card

Use the following procedure to save translations on the original system memory card and to make a backup memory card:

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. Login 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 AT&T Video Technical Center (VTC) at **800 242-2121** or the International Technical Access

Center (ITAC) at **303 538-4666**. Select **Video** from the voice menu and a VTC or ITAC engineer will assist you.

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, then swap the memory cards. Place the backup memory card in a safe place.

Checking MCU Performance

3

Overview

You can request a variety of reports about the MCU. These reports provide details about 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, you can request to print individual reports 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 allows you to specify when each report is printed (**immediate**, **deferred**, or **scheduled**). The **immediate** option prints the report on-demand and is the default.

The **deferred** option allows you to schedule 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* so you can 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 you remove it 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 you want to change a scheduled report 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 you to view or print a history of the most recent administration and maintenance changes. 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

There are three reports that provide traffic measurements for trunk groups and Call-By-Call (CBC) reports. The Trunk Group Call-By-Call 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 Call-By-Call 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, you can optimize, within certain limits, the CBC trunk group without involving any of the Inter-Exchange Carrie/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 you can analyze this report, you must know the intent of the UAP.

The following list identifies the data collected by the Trunk Group Measurements Call-By-Call 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 in order to run tests.

- *Usage Allocation Plan Used*

  A list of the UAP numbers followed by a list of the respective amount of time (in minutes), the duration, that each plan was in effect during the measurement period. The *Number* field can display up to a maximum of six plan numbers; a maximum of three plans may be defined for each trunk group.

- *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 AT&T, 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 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 in order 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:
In order to generate this report you must first enter the number of trunks you want measured hourly 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 in order 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.

Automatic Circuit Assurance Reports

The ACA Parameters Report lists all trunk groups in the system and displays the current definitions for long and short holding times. It identifies possible malfunctioning trunks by providing an alerting mechanism that monitors the occurrences of an excessive number of short holding time calls and abnormally long holding time calls. The report sets time ranges for each trunk group.

NOTE:
The *ACA Assignment* field must be set to **y** on the Trunk Group form (see "Administering Trunks" on page 2-16 in Chapter 2). ACA must also be enabled on the Feature Related System Parameters form described in this section.

The ACA Measurements Report displays the audit trail of any short or long holding time that exceeds the established limits. It displays up to 64 of the most recent entries.

Table 3-7 lists the commands related to obtaining ACA measurements.

Table 3-7. Automatic Circuit Assurance Administration Commands

Action	Object	Qualifier
change	system-parameters features	
list	aca-parameters <number>	count, print, or schedule
list	measurements aca	print or schedule

Features-Related System Parameters Form

The following list shows the fields on the Feature-Related System Parameters form that must be set use to the ACA feature:

- *Automatic Circuit Assurance (ACA) Enabled*

  Enter **y** to activate the ACA features to allow the MCU to take trunk group measurements and report possible malfunctions.This field must be set to **y** if you want to collect data for the ACA reports.

- *ACA Referral Calls*

  Displayed when ACA is enabled, specifies where the ACA referral calls are generated. The default and the correct setting is local.

- *ACA Referral Destination*

  Enter the extension number of the maintenance alarm terminal that will alert when a malfunction is reported.

- *ACA Short Holding Time Originating Extension*

  Enter an unassigned extension number that is used to identify when the referral is the result of too short holding times. A short holding time warning is sent to the maintenance alarm terminal specified as the ACA referral destination and displayed as “ACA CALL(SHORT).”

- *ACA Long Holding Time Originating Extension*

  Enter an unassigned extension number that identifies when the referral is the result of excessively long holding times. A long holding time warning is sent to the maintenance alarm terminal specified as the ACA referral destination and displayed as “ACA CALL(LONG).”

ACA Parameters Report

The following list shows the data collected by the ACA Parameters Report:

- *Group Number*

  A unique number assigned during administration that identifies each trunk group.

- *TAC*

  The Trunk Access Code (TAC) assigned during administration for the trunk group. An ISDN trunk cannot have a TAC assigned to it.

- *Group Type*

  The type of trunk associated with the accumulated data (**isdn-pri** or **tie**).

- *Group Name*

  The trunk group identification that is administered on the Trunk Group form.

- *ACA On?*

  Indicates whether or not the trunk group is being monitored by ACA. When you do not have a specific reason to monitor a trunk group, set this field to **n**.

- *Short Hold Time*

  The maximum number of seconds that a call is considered a short holding time call. A holding time longer than this value is considered normal up until the long holding time is exceeded. The short holding time value is specified on the Trunk Group form.

- *Short Threshold*

  The system maintains a running total of calls that have short holding times. The count is increased by one for each call that meets that criteria and decreased by one for each call that does not. When this count reaches a designated threshold, an entry is made on the ACA Measurements Report and a referral call is placed. The threshold value is administered on the Trunk Group form.

- *Long Hold Time*

  The minimum time of seizure, in hours, that the MCU will consider a call as having a long holding time. This number is specified on the Trunk Group form. Since video conference calls generally run for several hours, this number should account for the longer duration. A referral call is placed as soon as a single long holding call is detected. Subsequently, a referral call is placed at the top of the hour until the problem is resolved or the trunk becomes idle.

ACA Measurements Report

The following list shows the data collected for the ACA Measurements Report:

- *Day & Time of Referral*

  The day and time at which either the threshold for the short holding time was exceeded or long holding time call was reached and a referral call was entered (shown as day/hh:mm). Referral calls are attempted only if the referral button is enabled. A referral call is completed if the call is answered. A call that is not answered is attempted again when a new ACA call is received.

- *Trunk Group No.*

  Number of the trunk group in which the referral occurred.

- *Trunk Access Code*

  The Trunk Access Code (TAC) number administered for the trunk group.

- *Trunk Member*

  One specific trunk in the group that incurred the referral. This information can be used with other maintenance tests to identify the equipment location (circuit pack) of the trunk group member.

- *Type of Referral*

  Indicates whether or not the referral occurred as the result of too many short holding times or long holding times. Depending upon local arrangements, it may be appropriate to notify maintenance personnel if they have not already been alerted.

DS1 Measurements

There are two performance measurements for DS1 links. The DS1 Link Performance Measurements Summary Report provides an indication of the quality of a DS1 link that connects to a DS1 interface circuit pack. The DS1 Link Performance Detailed Log Report lists errored events for the past 24 hours. The errored event records are listed for each 15-minute interval. This shows 96 records (for each 15-minute interval) from the current time back 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 the *AT&T MultiPoint Control Unit Reservation Agent Manual* for troubleshooting details).

Table 3-8. DS1 Measurements Administration Commands

Action	Object	Qualifier
clear	measurements-ds1	
list	measurements ds1	location, print, or schedule
list	measurements ds1-log	location, print, or schedule

Where location is **NNCSSF** where **NN** is the port network number, **C** is the carrier, **SS** is the slot and **F** is the facility number.

DS1 Link Performance Measurements Summary

The following list shows the data collected on the DS1 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 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 Link Performance Detailed Log Report

The following list shows the data collected on the DS1 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 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 3-9. 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 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.

- *Queue Ovfl.*

  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 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 3-10. 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 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 3-11. System Status Reports Administration Commands

Action	Object	Qualifier
monitor	system view1	
monitor	system view2	

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" on page 3-26 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 3-12. 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 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 first10 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 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*

  Since 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 allow you to determine usage, available capacity and help to diagnose certain customer-reported problems.

Table 3-13 lists the commands that relate to occupancy measurements.

Table 3-13. 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 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.

- *Intcom Atmpt*

  This field is not used.

- *Out Atmpt*

  This field is not used.

- *Pnet Atmpt*

  The number of incoming seizures that are made over private network facilities.

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.

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.

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 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 3-14. Tone Receiver Measurements Administration Command

Action	Object	Qualifier
list	measurements tone-receiver	last-hour, today-peak, yesterday-peak, print, or schedule

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 touch-tone 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.

P x 64 Interoperability

A

The Px64//H320 standards have been implemented on the AT&T MultiPoint Control Unit (MCU). in conformance with the ITU-T standards on video teleconferencing. AT&T has tested, and will continue to test interoperability of its Px64 implementation with the video products of other manufacturers. AT&T will disclose the products that are found to be interoperable with the MCU. The current results on AT&T Px64 interoperability testing are provided in the table below. Tests are conducted on-site in a laboratory environment with purchased or loaned video endpoints and the tests will carry the distinction of AT&T certification of results.

NOTE:
The ITU-T standards are implemented on a self-certification basis. 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 and add features or options to their endpoints, their products may change. AT&T will have no knowledge or control of any such changes made by other manufacturers in their Px64 implementation. As such, AT&T cannot guarantee the interoperability of our ITU-T implementation with untested versions of ITU-T compliant endpoints.

Table A-1 shows the AT&T and non-AT&T video endpoints that are supported by the MCU.

NOTE:
This table is current as of the print date for this document. For the most current version of the table, consult your Account Executive.

Table A-1. MCU Supported Video Endpoints

Endpoint Vendor/ Software Version	FCIF	QCIF	PCM	LD-CELP	112 Kbps	128 Kbps	384 Kbps	Data Module & MCU Version
AT&T/NCR PVS 70 Version 1.01, 1.00.12	Yes	Yes	Yes	Yes	Yes	Yes	NA	Direct DEF BRI MCU 1.0
AT&T GVS 4000 Version 3.0P.03	Yes	Yes	Yes	NA	Yes	Yes	Yes	ABC 2.3 MCU 1.0
AT&T GVS 1000 Version 1.0	Yes	Yes	Yes	Yes	Yes	Yes	NA	Direct DEF BRI MCU 1.0
AT&T GVS 4000 Version 4.1V	Yes	Yes	Yes	NA	Yes	Yes	Yes	ABC 2.3 MCU 1.0
CLI Rembrandt II/VP Version 7.94	Yes	Yes	Yes	NA	Yes	Yes	Yes	ABC 2.3 MCU 1.0
CLI Eclipse Version 1.1	NA	Yes	Yes	Yes	Yes	Yes	NA	Direct DEF BRI MCU 1.0
GPT Version 2.70	Yes	Yes	Yes	NA	Yes	Yes	Yes	HSL/DEFG3RV2 MCU 1.0
PictureTel S4000 Version 3.0P.03	Yes	Yes	Yes	NA	Yes	Yes	Yes	ABC 2.3 MCU 1.0
PictureTel S4000 Version 4.1V	Yes	Yes	Yes	NA	Yes	Yes	Yes	ABC 2.3 MCU 1.0
PictureTel S1000 Version 1.0	Yes	Yes	Yes	Yes	Yes	Yes	NA	Direct DEF BRI MCU 1.0
SONY PCS-2000A/ 2000AP PCS-Version 1.04	Yes	Yes	Yes	NC	NC	NC	Yes	ABC 2.3 MCU 1.0

Yes—Endpoint passed a voice-activated-switching test with the MCU by dialing in from AT&T premises and participating in a conference.

NA—Feature is not applicable (the endpoint does not support the feature).

NC—Test was not conducted.

Paper-Based Scheduling Forms

B

The blank forms on the following pages can be copied for use with the paper-based scheduling system described in the *AT&T MultiPoint Control Unit Reservation Agent Manual.*

Included in this appendix are the following forms:

- Site Profile
- MCU-Extensions and Numbers to Dial

Use the following two forms to prequalify video endpoints and record available Meet-Me numbers for multipoint video conferences.

NOTE:
The following forms are just examples of what you might use. Be sure to modify the forms to satisfy the requirements at your site.

Site Profile					
Location Name	**Time Zone**	**Bandwidths Available**	**Network Type**	**Room Location**	**Contact Name/Phone**

MCU-Extensions and Numbers to Dial							
		Number to Dial				Number to Dial	
Ext.	**Bandwidth**	**Public**	**Private**	**Ext.**	**Bandwidth**	**Public**	**Private**

Application Notes

C

This appendix discusses how the MCU system should be configured and administered for operation in each supported country. Also included is information about how certain features (for example, ARS) should be used. Additionally, potential problems that might be encountered in supported countries are discussed.

Only administration and hardware items that must be changed from their default configuration for the MCU to meet in-country type approval requirements and provide basic interworking with the public telephone network are discussed.

Also, only a limited set of applications are addressed by the configurations and administration values specified in this appendix. However, by following this appendix, a system setup that allows digital calls and video conferencing on DS1 or UDS1 trunks and meets type approval requirements in each country will result. To implement the system fully, refer to the appropriate sections of this manual for in-depth information.

The appendix is organized into two parts:

- A general overview of hardware configurations and certain systemwide options (such as tone plan country codes) presented in table format.
- Specific sections covering each of the countries supported by the MCU.

Each country-specific section provides a brief background on key aspects of that country's Public Network (PN), such as major digital standards. This is followed by subsections dealing with the administration of the MCU for connection to in-country digital Public Network Trunks (PNTs), selection of proper MCU-provided call-progress tones, and private network connections.

Hardware Overview

This section provides a listing of the various circuit packs supported by the MCU as well as the hardware administration of the Country Code assignments.

Circuit Pack Support

The following tables list the circuit packs available in MCU installations. If a circuit pack is listed without a specific vintage, you may use any vintage. If a vintage or suffix is listed for the circuit pack, you may use a circuit pack with a greater vintage or suffix.

Table C-1 contains a summary of the circuit packs supported in the appropriate countries.

Table C-1. MCU Circuit Pack Support Per Country

Country	A or μ Encoding	Ton Det	Tone Gen Clock	MMI	VC
U.S. and Canada	μ	TN748D	TN768	TN787C	TN788
Australia	A	TN420C	TN780	TN787C	TN788
U.K.	A	TN420C	TN780	TN787C	TN788

Tables C-2 and C-3 list the trunk and line circuit packs supported for each country by the MCU.

Table C-4 contains a summary of circuit pack functionality for all trunk and line circuit packs supported by the MCU.

Table C-5 lists the processing and environmental circuit packs supported by the MCU.

Table C-2. Trunk Circuit Packs Supported by the MCU

Country	Tie	ISDN CO	Encoding A/μ
U.S. and Canada	TN767D	TN767D	μ
Australia	TN2207	TN2207	A
U.K.	TN2207	TN2207	A

Table C-3. Line Circuit Packs Supported by the MCU

Country	Dig Line	Encoding A/μ
U.S. and Canada	TN754B	μ
Australia	TN754B	A
U.K.	TN754B	A

Table C-4. Trunk/Line Circuit Pack Functionality Summary

Circuit Pack Name	Description	# Ports	Supports Administrable Timers?	Encoding A/μ
TN754B	Digital Line	8	N/A	Cμ*
TN767D	DS1	24	Yes	μ
TN2207	UDS1	24/32	Yes	μ or *

* Software controlled — μ-Law default

Table C-5. Processing/Environmental Circuit Packs per Country

Country	Processor Circuit Pack	AC Power Supply	DC Power Supply	Network Control
U.S. and Canada	TN765 w/CPP1*	†	676B	TN777B
Australia	TN765 w/CPP1*	†	676B	TN777B
U.K	TN765 w/CPP1*	PEC3981 UK	676B	TN777B

*CPP1 is the memory expansion circuit pack.

†The following circuit packs are used: WP91153 L2, 644A, 645B.

Hardware Administration Country Codes

The MCU supports the administration of circuit pack operating parameters through the selection of *country codes*. Each country code corresponds to a pre-defined set of (one or more) programmable hardware and/or firmware affecting attributes.

Country code assignments are listed in the following table:

Table C-6. Country Code Assignments

Administered Code	Country
1	U.S. and Canada
2	Australia
3	Japan
4	Italy
5	Netherlands
6	Singapore
7	Mexico
8	Belgium

Table C-6. Country Code Assignments *(continued)*

Administered Code	Country
9	Saudi Arabia
10	U.K.
11	Spain
12	France
13	Germany
14	Czechoslovakia
15	CIS
16	Argentina
17	Greece

NOTE:
Country Codes 3 through 9 and 11 through 17 are not currently supported by the MCU.

If no country code has been assigned to a particular country, an administrator must use the following table to determine the appropriate code for each feature area in that country when configuring the system.

Table C-7. Recommended Country Codes to be Entered by Administrators

Country	TONE GEN	ISDN	DIG XMIT	TONE DET
U.S. and Canada	1	1	1	1
Australia	2	2	2	2
U.K. OTR001	10	10	10B	3
U.K. Non-OTR001	10	10	10A	3

Country-Specific Configurations and Administration

The following sections present information on how the MCU should be administered for each of the supported countries. The values shown are those you should actually enter into the fields on the specified forms.

United States and Canada

All the domestic circuit packs listed in the following table may be used with the MCU.

Table C-8. Supported Domestic Port and Service Circuit Packs

Circuit Pack Code	Function	Notes
TN748D	Tone Detector	
TN754B	Digital Line	
TN767D	DS1 Interface	
TN768	Tone Clock	Stratum 3
TN787B	Multimedia Interface	
TN788	Voice Conditioner	

Administration

All timers and option selections default to the values appropriate for operation in the U.S. Whenever a country code is requested, use Code **1**.

Australia

The recommended and available circuit packs for Australia are listed in the following table:

Table C-9. Recommended and Available Circuit Packs for Australia

Circuit Pack Code	Function	Notes
TN420C	Tone Detector	V2 circuit pack to be sold
TN754B	Digital Line	
TN780	Tone Gen/Clock	
TN787B	Multimedia Interface	
TN788	Voice Conditioner	
TN2207	ISDN	

Feature Administration

Only those feature-related parameters that may have a type approval or basic connectivity implication for a particular country are included in the following list:

- ARS/AAR Administration

  Be sure to check all ARS/AAR defaults. These defaults are intended for U.S. operation; therefore, certain values are likely to be inappropriate for international use. In particular, you should usually not use the following call types (which appear as ARS/AAR defaults) in non-North American ARS/AAR tables:

  - **fnpa**: North American numbers with an area code
  - **hnpa**: North American numbers without an area code
  - **svc**: North American numbers of the form ‘‘x11’’

  It is recommended that you use only the following call type in Australia:

  - **int** (for all international numbers)

System Parameter Administration

Only those feature-related parameters that may have a Type Approval or basic connectivity implication for a particular country are listed here.

- **Features**
  - Dial-out Post Answer Failure Retry Limit: **4**
  - AAR/ARS Dial Tone Required?: **y**
  - Automatic Circuit Assurance (ACA) Enabled? **n**
  - SMDR Parameters:
    - Primary Output Layout: **expanded**
    - Primary Output Destination: **eia**
    - Use ISDN Layouts?: **y**
    - EIA Device Bit Rate: **9600**
    - Disconnect Information in Place of Facility Restriction Level?: **n**
    - Suppress SMDR for Ineffective Call Attempts?: **y**
    - SMDR Date Format: (user supplies in the form *day/month*)
    - Privacy—Digits to Hide: **0**
  - System Printer Parameters:
    - Printer Extension: (User supplies)
    - EIA Device Bit Rate: **9600**
    - Lines Per Page: **60**
  - Security Violation Notification Parameters:
    - SVN Login Violation Notification Enabled?: **n**
- **Country Options**
  - Companding Mode: **A-law**
  - Base Tone Generator Set: **2**
  - 440Hz MCU-Dial Tone?: **n**
  - 440Hz Secondary-Dial Tone?: **n**
  - Digital Loss Plan: **2**
  - Tone Detection Mode Parameters
    - Tone Detection Mode: **2**
    - Interdigit Pause: **short**
    - Tone Name: (blank)

- — Tone (Frequency/Level): (blank)

Digital Trunk Administration

Not all possible valid administrable combinations are listed in this section; only the most common or standard combination that is compatible with public network or Type Approval standards for each country is presented.

- Tie Trunks
  - — Non-ISDN Signaling Example (DS1 Administration Screen)
    - Bit Rate: **2.048**
    - Interface Companding: **A-law**
    - Line Coding: **HDB3**
    - Signaling Mode: **CAS**
    - Country Protocol: **2**
    - CRC?: **no**
    - Idle Code: **11111111**
  - — ISDN-PRI (Private Network) Signaling

    This example assumes the use of Australia Option 2 with facility associated signaling. Other feature options require changes in one or more administered items.
    - DS1 Administration Form
      - — Bit Rate: **2.048**
      - — Interface Companding: **A-law**
      - — Line Coding: **HDB3**
      - — Signaling Mode: **isdn-pri**
      - — Country Protocol: **2**
      - — Connect: **pbx**
      - — Interface: **user**
      - — CRC: **no**
      - — Idle Code: **11111111**

- Signaling Group Form
  - — Associated Signaling: **Yes**
  - — Primary D-Channel: **xxxx16**
    (xxxx=>depends on location of circuit pack)
- Trunk Group Administration Form
  - — Group Type: **isdn-pri**
  - — Service Type: **tie**

- ISDN-PRI (Public Network)
  - — DS1 Administration Form
    - Bit Rate: **2048**
    - Interface Companding: **A-law**
    - Line Coding: **HDB3**
    - Signaling Mode: **isdn-pri**
    - Country Protocol: **2**
    - Connect: **Network**
    - CRC: **No**
    - Idle Code: **11111111**
  - — Signaling Group Form
    - Associated Signaling: **Yes**
    - Primary D-Channel: **xxxx16** (xxxx=>depends on the physical location of the circuit pack)
  - — Trunk Group Administration Form
    - Group Type: **isdn-pri**
    - Service Type: **public_ntwrk**

United Kingdom

The recommended and available circuit packs for the United Kingdom are listed in the following table:

Table C-10. Recommended and Available Circuit Packs for the United Kingdom

Circuit Pack Code	Function	Notes
TN420C	Tone Detector	V2 circuit pack to be sold
TN754B	Digital Line	
TN780	Tone Gen/Clock	
TN787B	Multimedia Interface	
TN788	Voice Conditioner	
TN2207	ISDN	Needs an external converter for ETSI to DAS II/DPNSS

Feature Administration

Only those feature-related parameters that may have a type approval or basic connectivity implication for a particular country are included in the following list:

- ARS/AAR Administration

  Be sure to check all ARS/AAR defaults. These defaults are intended for U.S. operation; therefore, certain values are likely to be inappropriate for international use. In particular, you should usually not use the following call types (which appear as ARS/AAR defaults) in non-North American ARS/AAR tables:

  - **fnpa**: North American numbers with an area code
  - **hnpa**: North American numbers without an area code
  - **svc**: North American numbers of the form ‘‘x11’’.

  It is recommended that you use only the following call types in the United Kingdom:

  - **int** (for all international numbers)
  - **natl**: (for all national PN numbers)

System Parameter Administration

Only those feature-related parameters that may have a Type Approval or basic connectivity implication for a particular country are listed here.

- **Features**
  - Dial-out Post Answer Failure Retry Limit: **4**
  - AAR/ARS Dial Tone Required?: **y**
  - Automatic Circuit Assurance (ACA) Enabled? **n**
  - SMDR Parameters:
    - Primary Output Layout: **expanded**
    - Primary Output Destination: **eia**
    - Use ISDN Layouts?: **y**
    - EIA Device Bit Rate: **9600**
    - Disconnect Information in Place of Facility Restriction Level?: **n**
    - Suppress SMDR for Ineffective Call Attempts?: **y**
    - SMDR Date Format: (user supplies in the form *day/month*)
    - Privacy—Digits to Hide: **0**
  - System Printer Parameters:
    - Printer Extension: (User supplies)
    - EIA Device Bit Rate: **9600**
    - Lines Per Page: **60**
  - Security Violation Notification Parameters:
    - SVN Login Violation Notification Enabled?: **n**
- **Country Options**
  - Companding Mode: **A-law**
  - Base Tone Generator Set: **10**
  - 440Hz MCU-Dial Tone?: **n**
  - 440Hz Secondary-Dial Tone?: **n**
  - Digital Loss Plan: **10**
  - Version of Digital Loss Plan: **B**
  - Tone Detection Mode Parameters
    - Tone Detection Mode: **3**
    - Interdigit Pause: **short**

- — Tone Name: (blank)
- — Tone (Frequency/Level): (blank)

Digital Trunk Administration

Not all possible valid administrable combinations are listed in this section; only the most common or standard combination that is compatible with public network or Type Approval standards for each country is presented.

- Tie Trunks
  - — Non-ISDN Signaling Example (DS1 Administration Screen)
    - Circuit Pack: TN2207 (or TN464C from upgrades)
    - Bit Rate: **2.048**
    - Interface Companding: **A-law**
    - Line Coding: **HDB3**
    - Signaling Mode: **CAS**
    - Country Protocol: **10**
    - CRC?: **no**
    - Idle Code: **01010100**
  - — ISDN-PRI (Private Network) Signaling

    This example assumes the use of U.K. Option 10 with facility associated signaling. Other feature options require changes in one or more administered items.
    - DS1 Administration Form
      - — Bit Rate: **2.048**
      - — Interface Companding: **A-law**
      - — Line Coding: **HDB3**
      - — Signaling Mode: **isdn-pri**
      - — Country Protocol: **10**
      - — Connect: **pbx**
      - — Interface: **user**
      - — CRC: **no**
      - — Idle Code: **01010100**

- Signaling Group Form
  - — Associated Signaling: **Yes**
  - — Primary D-Channel: **xxxx16**
    (xxxx=>depends on location of circuit pack)
- Trunk Group Administration Form
  - — Group Type: **isdn-pri**
  - — Service Type: **tie**

- ISDN-PRI (Public Network)
  - — DS1 Administration Form
    - Bit Rate: **2048**
    - Interface Companding: **A-law**
    - Line Coding: **HDB3**
    - Signaling Mode: **isdn-pri**
    - Country Protocol: **10**
    - Connect: **Network**
    - Idle Code: **01010100**
  - — Signaling Group Form
    - Associated Signaling: **Yes**
    - Primary D-Channel: **xxxx16** (xxxx=>depends on the physical location of the circuit pack)
  - — Trunk Group Administration Form
    - Group Type: **isdn-pri**
    - Service Type: **public_ntwrk**

Glossary

A

active
A reserved conference status when the conference is scheduled to begin but no conferees have joined. A dedicated conference status is active at all times.

advanced mode
A feature of the MCU that offers presentation and broadcast with auto scan conference modes.

AT&T Conference Reservation 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.

AT&T MultiPoint Control Unit
Equipment that provides high-quality multimedia conferencing with video endpoints that communicate via the ITU-T Px64 standards.

AT&T Video Technical Center
A video technical support group that is available to help you with questions about the AT&T MCU R1.0.

audio add-on
Feature that allows the MCU to support one non-Px64 Audio-only endpoint participant per conference.

audio port
The port used for an audio add-on party to join a conference. See also *port*.

automatic mode
A feature of the MCU that offers voice-activated conference mode.

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), 384k (H0) bandwidth, and 1-channel bandwidths of 768k, 1472k, and 1536k. Also, bandwidths such as 336k and 384k are available through the bonding feature.

bandwidth on demand
Feature that allows the AT&T MCU to establish conferences at bandwidths greater than 128k without the use of ISDN-PRI and Wideband H0 channels.

bonding
See “bandwidth on demand.”

bonding ports
The ports in an MCU conference that use the bonded bandwidth 336k or 384k.

broadcaster
A video endpoint is considered the broadcaster when it sends the same video, audio and data signal to two or more locations.

broadcast w/scn
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.

C

cascading
Two or more AT&T MCUs connected via a Px64 communications link. This feature allows endpoints connected on different MCUs to participate in the same conference.

chair control mode
A conference mode where one site controls which endpoints are added and/or dropped during a conference.

CCITT
See ITU-T.

CIF
Common Intermediate Format (also known as FCIF or Full CIF). An industry-standard means for encoding video signals for high quality pictures. QCIF MPI specifies a picture interval numbers.

class
A Conference Record has one of three classes: reserved, dedicated or file.

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.

convener
The person who schedules a conference and is responsible for distributing the Meet-Me numbers.

CRS
See AT&T Conference Reservation System.

D

dedicated
A conference class that reserves a specified number of MCU ports for a multipoint video conference at anytime.

dial-out
Feature that allows Px64 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.

dynamic conference resizing
Feature that allows the user to add and remove conference participants before the conference starts as well as during the conference.

DX
An MCU model that features up to 20 ports and the automatic conference mode.

E

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.

EX
An MCU model that features up to 28 ports, notification tones and automatic and advanced conference modes.

exit tone
A tone that alerts conferees on a conference that a conferee has left the conference.

F

far-end
When troubleshooting problems on the MCU, the video endpoint is often referred to as being on the far-end of the connection.

FCIF
See CIF.

file
A conference class that saves a copy of a Conference Record for future use.

FX
An MCU model that features up to 24 ports and automatic and advanced conference modes.

G

G.711
See PCM.

G.728
See LD-CELP.

H

H0
A one channel 384k bandwidth call.

H.320
The multipoint conferencing standard adopted by the ITU-T (also known as Px64). 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-series
See H.320.

handshake
An expression relating to the establishment of communications between the MCU and a video endpoint.

I

Inactive
A conference status when a conference is reserved but has not begun and when a conference has a class of file.

In-use
A conference status when a reserved conference has begun and at least one conferee has joined the conference.

ITU-T
An international group known as International Telecommunications Union-Telecommunications Formerly CCITT) established in 1993 to develop standards to allow video endpoints to communicate with each other.

L

LD-CELP
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.

login
A keyword assigned to a user to access the MCU.

login password
A codeword that can be customized by the user that corresponds with the user's login.

M

Management Terminal
A 715 BCT used primarily for administration and maintenance functions.

MCU

See AT&T MultiPoint Control 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-MT
See Management Terminal.

MCU-ST
See Scheduling Terminal.

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.

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.

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.

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.

notification tones
The entry, exit, and warning tones available with some of the MCU models.

P

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 AT&T Conference Reservation System to accept reservations and qualify video endpoints to participate in a multipoint video conference.

PCM
Pulse Code Modulation. A means of encoding audio which uses less bandwidth for video than LD-CELP so the picture quality is not as good as LD-CELP.

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.

Port and Extension Usage Chart
A chart that is used to record the number of MCU ports and MCU-extensions reserved for a particular day.

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 whom is speaking. Any conferee can be a speaker. The video endpoint that is the presenter views endpoints determined by voice activated switching.

protocol
A set of international industry-standard rules governing the exchange of data between two entities.

Px64
Also known as H-series. The standards adopted by the ITU-T committee that allow video endpoints that comply with the standards to communicate with each other.

Q

QCIF
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 AT&T MCU to interwork endpoints operating at 2B that are on both 56k networks and 64k networks.

recurring meeting
A meeting that occurs more than once in a definite pattern, such as every Thursday.

reserved
A conference class that indicates a conference will begin and end within the next 24 hours.

reservation agent
The person responsible for accepting multipoint conference reservations, making reservations on the MCU-ST, or CRS and resolving minor problems.

rotation scan time
The number of seconds that a broadcaster views each location before viewing the next location in the rotation.

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.

Site Profile
A list of video endpoints that may participate in multipoint video conferences and their location, time zone, bandwidth, and network type.

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.

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

tones
See Notification Tones.

V

video endpoint
A video codec, with camera, speakers, screen and other equipment required for multimedia conferencing.

VTC
See AT&T Video Technical Center.

voice-activated
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.

W

warning tone
A tone that sounds when only 10 minutes remain in a conference.

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