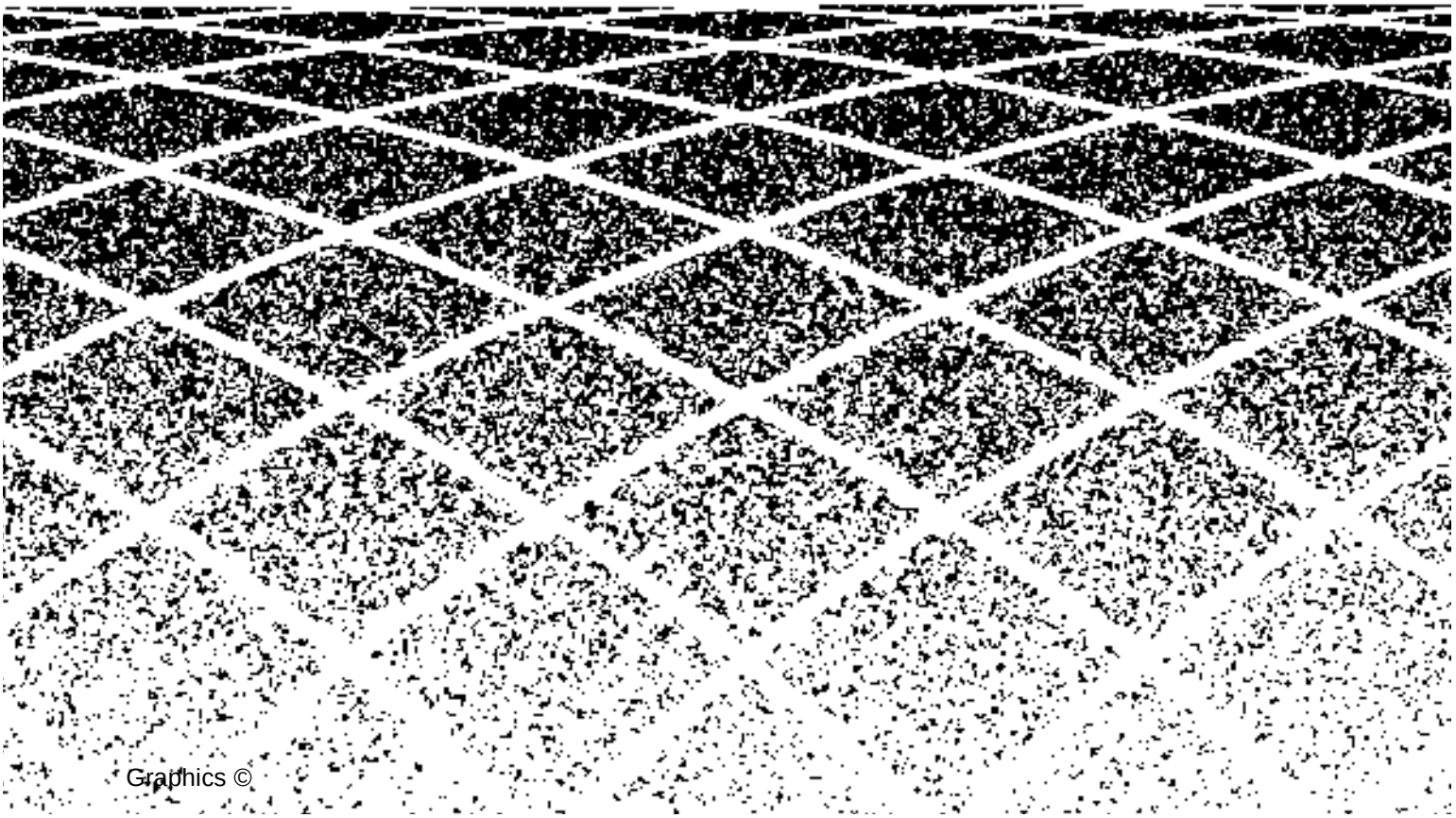
555-027-727
Issue 4
April, 1995

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 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 the tests, alarms, and maintenance of an AT&T MCU.

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

  Provides detailed instructions that an AT&T MCU reservations 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

  An AT&T course offering that provides 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 that 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.

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

  Discusses security risks and measures that help prevent external telecommunications fraud involving AT&T products, including the R1V3 switch and various terminals.

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

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

- *AT&T GBCS Toll Fraud Overview,* 015-338-100

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

Organization

This manual contains the following three chapters and three 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 that can be displayed or printed to track MCU performance and status of trunks.
- Appendix A, "Px64 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.
- 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 In dex.

Typographic Conventions

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

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

WARNING:

emphasizes information that is important to your safety.

- The following icon:

CAUTION:

indicates information you need to prevent equipment damage.

- The following icon:

NOTE:

identifies additional information pertinent to the text preceding it.

Getting Help

If you need help with the procedures or other information in this document, and if you are a customer in the United States or Canada, be sure to call the Video Technical Center (VTC) at 800 242-2121. If you are a customer in Australia or the United Kingdom, call the Global Technical Access Center (GTAC) at 303 538-4666.

MultiPoint Conferencing

1

Overview

The AT&T MultiPoint Control Unit (MCU) Release 3.0 brings multiple remote video (and, optionally, audio add-on) and data endpoints onto a single video conference call. It is a multimedia product because it combines audio, video, and data. In this chapter, background information is given on video/audio/data conferencing and the tasks needed to perform it.

Standard vs. Proprietary

Until recently, video endpoints, such as 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 International Telecommunications Union- Telecommunications (ITU-T) 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/data 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 because they are dialing a Meet-Me number. However, you must be aware of the mapping of Meet-Me numbers to MCU-extensions 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 the system administrator and the appropriate 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. Meet-Me Numbers can be mapped manually 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, the system administrator 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 reservations 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 "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 Private Brance Exchange (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 *MultiPoint 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 consisting of 2B-channel 56k or 64k. The MCU also supports the following 1-channel rates: 128k, 192k, 256k, 320k, 384k (H0), 768k, 1472k, 1536k, and 1920k. In addition, BONDing (discussed in-depth later in "BONDing" on page 1-10) 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 1-channel conferences. Also, up to six non-Px64 Audio Add-On endpoints (or up to 12 such endpoints if cascading is enabled) may be included in either type of conference (this is not indicated in the configuration).

The smallest MCU configuration contains four dedicated 2B 56k ports. A four-port system accepts a maximum of one four-party conference. The largest configuration is a 64/52 port system. With a maximum single 2B conference size of 64, the 64/52 port configuration can handle many combinations of conferences (such as four 16-party conferences or eight 8-party conferences). With a 64/52 configuration, up to 52 MCU ports can be used for H0 conferences. The following configurations represent the maximum port capacity involving each of the other 1-channel bandwidths (indicated in parentheses): 64/64 (for 128k or 192k), 64/60 (for 256k), 64/52 (for 384k), 64/36 (for 768k), 64/20 (for 1472k), and 64/20 (for 1536k), and 64/16 (for 1920k). The maximum BONDed 336k/384k port capacity is 16. Finally, the maximum number of Multipoint Communication Service/Multilayer Protocol (lMCS/MLP) data ports per MCU or per conference is 24.

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 the MCU to be adjusted 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 speakers. 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.

Contributor Mode (Automatic Control)

This mode (also known as the *See-Me* function) allows high-resolution video still images to be sent to the other endpoints. It also allows a broadcast to be sent to all the conferees in a conference. The relevant endpoints must support the ITU-T H.243 standard.

Chairperson Mode (User Control)

For this mode, the chairperson determines which video sending endpoint is broadcast to all the other conferees. The chairperson 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. Chairperson 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

The video quality can be adjusted 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 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 B-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 1-channel transfer rate of 1920k provides the maximum bandwidth and, therefore, the finest video image available with the MCU.

In most cases, the MCU will not allow an endpoint to participate fully (audio, video, and data) 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 (LD-CELP) uses 16k audio. For a conference without data, 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. For a conference involving data, the MCU supports MLP (described later) only with G.728 (LD-CELP). With G.728 (LD-CELP), MLP gets 56k or 64k bandwidth. The bandwidth beyond the initial channel is used for video. Audio modes are discussed in detail later in this chapter.

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. Entry and exit tones are a customer option.

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, a billing identification number (up to 15 characters) can be assigned. 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. Audio Add-On capacity can be increased 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.

Audio Add-On is optioned via the System Parameters Customer-Options form.

Dial-Out

Dial-Out is a customer option that allows the MCU to originate calls and dial out 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.

NOTE:
A Dial-Out retry can be forced by setting the Redial flag on the Conference Record form to **y**.

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. An alarm via the telephone alerts the appropriate person (for example, the Reservations Agent) to investigate the problem by using the status conference command.

Dial-Out is optioned via the System Parameters Customer-Options form.

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 is a customer option that 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, 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 reservations 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 reservations 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.

Cascading can be used 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 System Parameters Customer-Options form.

Dynamic Conference Resizing

This feature allows the reservations 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, Universal Conference Control, and Cascade links) and also to the “Basic/Enhanced,” “Application Compliant,” and “Redial” flags on the Conference Record. 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.

Also, the feature allows dial-out numbers to be removed and changed and the stop time on a reserved conference to be changed. Other capabilities include changing dial-out numbers to dial-in numbers (and vice versa) and changing a primary MCU to a secondary MCU (and vice versa).

Dynamic Conference Resizing is optioned via the System Parameters Customer-Options form.

Multirate Bandwidths

MCU conferences support the following Integrated Services Digital Network (ISDN) multirate bandwidths: 128k, 192k, 256k, 320k, 384k (H0), 768k, 1472k, 1536k, and 1920k. These are 1-channel bandwidths, and they use an Integrated Services Digital Network-Primary Rate Interface (ISDN-PRI) facility from an endpoint that provides the appropriate size bandwidth.

Endpoints that join a conference via multirate ISDN facilities interwork with BONDed endpoints in the same conference and at the same bandwidth. Also, one Audio Add-On endpoint may be added on a per-conference basis.

BONDing

BONDing (Bandwidth on Demand) allows the AT&T MCU to establish conferences at bandwidths of between 112k/128k and 336k/384k 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 comprises 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 comprises 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 multiple calls associated with a specific BONDing call (for example, six calls for a 384k bandwidth) may arrive over different Digital Signal Level-1 (DS1) facilities for this capacity. Also, the AT&T-MCU allows the interworking of H0 (384k) calls and BONDed 384k calls. This allows the endpoint(s) for each call type to participate in the same conference. The AT&T-MCU treats BONDed calls at different rates as separate calls.

The Digital Signal Level-1/Multimedia Interface (DS1/MMI) cable or the Time-Division Multiplexing (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, Cascading, Universal Conference Control, Dedicated Access, and G.722 (7kHz). An Audio Add-On party may participate in a BONDing conference by dialing into the AT&T-MCU. BONDing conferences can receive entry, exit, and end-of-conference warning tones. Finally, BONDed ports support a form of Low-/High-Speed Interworking (see the Low-/High-Speed Interworking section for more details).

BONDing is optioned via the System Parameters Customer-Options form.

Meet-Me Number—MCU-Extension Correspondence

This feature allows the Meet-Me Number to be seen listed next to its corresponding MCU-extension (provided these entries are correctly administered). This listing is available on the “add” MCU-Extension form, the “list” 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 audio/video endpoints operating at 2B that are on 56k 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). However, these endpoints can view other conference participants.

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.

Rate adaptation is administrable for cascaded conferences.

Multipoint Communication Service (MCS)/ Multilayer Protocol (MLP)

The MCU supports a multipoint data conferencing feature based on the H.221 MLP as specified in ITU-T Recommendations T.122, T.123, and T.125 for the MCS. Only the “variable” rate MLP is supported. The Expansion Services Module (ESM) is required to support this feature.

“Variable” MLP takes the capacity that remains in the initial channel (56k/64k) after it deducts the framing signals (1.6k) and the audio bit rate (16k, 48k, or 56k). Although MLP may coexist with any of the audio modes supported by the MCU, only MLP with G.728 (LD-CELP) is supported. When G.728 (LD-CELP) is used, MLP receives either 56k or 64k bandwidth. The bandwidth beyond the initial channel is used for video.

The Selected Communication Mode (SCM) of a conference specifies whether or not the conference is using MLP. Whenever the MLP option is off, no MLP data channels are opened for the conference; this is true regardless of whether the endpoints support MLP. Whenever the MLP option is in effect, an MLP channel is opened only to the endpoints that support both var-MLP and G.728 (LD-CELP). Endpoints that do not declare a suitable MLP capability become audio-only secondary endpoints upon joining the conference. Endpoints that declare an acceptable MLP capability but do not declare G.728 (LD-CELP) also become audio-only secondary endpoints.

The MLP feature works in both stand-alone and cascading MCU environments.

Application Compliant Flag

The assignment of MCS channels is coordinated to prevent different applications from colliding on the same MCS channel. This is done in the absence of higher-level protocol that allows endpoints to negotiate what application to run and what MCS channel to use for a data collaboration session. However, this "coordination approach" does not work for all applications. Such applications are considered to be "non-compliant."

To resolve this problem, the Application Compliant Flag in the Conference Record can be used. With this per-user flag, certain endpoints can be chosen to use non-compliant applications. The MCU is designed to open an MCS/MLP channel to such endpoints, but it does not send data to them or receive data from them. Therefore, the endpoints can still participate in the audio/video portion of the conference without damaging the data collaboration session.

Conference Redial Flag

The Conference Redial flag in the Conference Record allows an endpoint to be "redialed" during a conference (therefore, this flag is in effect for dial-out calls). This can be done by changing the field entry from blank to *y*. An endpoint may need to be redialed if the user does one of the following during a conference:

- Fails three password validations and wants to try again
- Misses the password entry window
- Disconnects from the conference and wants to rejoin

The Conference Redial Flag can be set via Dynamic Conference Resizing.

Terminal Names

The MCU supports the passing of terminal names and identification as defined in the ANSI243 (ANSI version of H.243) recommendation. Endpoint participation in accordance with the ANSI243 recommendation is required.

This feature enables the MCU to poll and ascertain “naming/identification” parameters from endpoints and to pass the information on to other endpoints. Therefore, conference participants whose endpoints display this information can identify other conference participants.

A terminal name can contain up to 15 characters. The MCU uses the “Audio-Only” string as the terminal name for an Audio Add-On port. Finally, this feature can be used in a cascaded environment.

Basic/Enhanced Service Flag

Endpoints from several vendors tend to crash whenever these endpoints receive certain standard BAS commands/caps. To prevent this problem, set the Basic/Enhanced Service Flag in the Conference Record to *basic* (default). Doing so will disable commands/caps that are known to cause problems. If no such problems are apparent, set the flag to *enhanced*, where appropriate.

Per-Conference Password

This feature applies to dial-in, dial-out, and audio add-on calls. The password for a conference is provided by the customer at reservation time. Conferees are prompted for the password when they join the conference. Except for the conferee on the Audio Add-On port, each conferee has three chances to enter the password correctly (the Audio Add-On conferee has just one chance). If the conferee fails the password validation, the MCU issues an ns-command to notify the endpoint of the failure. The endpoint is then dropped, and the event is logged internally. The Status Conference form shows “invalid password” as the drop reason for that endpoint.

For a dial-out call, if the conferee is not there but the endpoint has turned on auto-answer, the MCU queries for the password once the call is answered. The query times out after a systemwide administrable interval (30-300 seconds), and the MCU reprompts for the password. After three timeouts, the endpoint is disconnected, and the event is logged internally. The Status Conference form shows “UIN/password timeout” as the drop reason for that endpoint. The only way for the dial-out conferee to join the conference after the link is disconnected is to call the reservations agent.

Audio Quality

The highest quality audio that can be arranged for a video conference is known as G.722 (7kHz) audio. G.722 provides a 7kHz audio bandwidth. This bandwidth allows for a more natural sounding voice conference. The MCU supports G.722-conferenced audio at 48k or 56k, as defined in ITUT-T Recommendation G.722.

G.728 Low Delay Codebook Excited Linear Prediction (LD-CELP) uses 16k audio and is a method of speech compression designed to enhance video quality.

G.711 Pulse Code Modulation (PCM) uses 56k/64k audio. Essentially, this selection maximizes the bandwidth allocated to audio. Therefore, the video quality is not so good as that offered by G.728 (LD-CELP).

If an endpoint with G.711 (PCM) audio joins a conference with G.722 (7kHz) audio, the conference retains the G.722 (7kHz) audio mode due to interworking. Also, G.728 (LD-CELP) is not allowed in a G.722 (7kHz) conference because the video rates do not match.

For 2B conferences, if *Audio Mode* in the Conference Record is administered as "auto," and if the MLP option is off, G.728 (LD-CELP) is the preferred audio mode. For conferences higher than 2B, if *Audio Mode* is administered as "auto," G.722 (7kHz) is the preferred audio mode.

The MCU interworks a non-H.320 Audio-only endpoint so that the endpoint can participate in the audio portion of the conference.

Selected Communications Mode (SCM) Upgrade

Selected Communications Mode (SCM) is used to configure multipoint conferences as endpoints are joined to the MCU (both Meet-Me and Dial-Out endpoints qualify). Highest Common (HC), the default conference mode, adaptively selects the audio mode, either the Common Intermediate Format (CIF) or Quarter Common Intermediate Format (QCIF) video resolution, and Minimum Picture Interval (MPI) to provide multimedia service to the maximum number of endpoints. The HC audio and video format algorithms begin by assuming that the SCM of the conference is a "high-quality" audio mode (G.728, for example) and video format (CIF at 30 frames per second, for example). If, as endpoints join the conference and provide the MCU with their capability information, an endpoint that does not support the "higher-quality" audio mode, video resolution, and/or MPI joins the conference, the MCU changes the SCM of the conference and all the endpoints to a "lower-quality" audio mode (G.711, for example) and/or video format (QCIF at 15 frames per second, for example). However, SCM Upgrade allows that conference to return automatically to the "higher-quality" audio mode (G.728, in this case) once the last "lower-quality" audio endpoint (G.711, in this case) disconnects from the conference. Similarly, once the last "low-resolution" video format endpoint (QCIF at 15 frames per second, in this case) disconnects from the conference, the remaining endpoints return to the "higher-quality" video format (CIF at 30 frames per second, in this case).

All MCU features except for cascading and rate adaptation support SCM Upgrade.

Low-/High-Speed Interworking

This feature allows 56k or 64k endpoints to join higher-speed conferences as Audio Add-On endpoints via conventional dial-in Px64 ports. Accordingly, the user is required to place *only one call* to the MCU to use this feature correctly. Multiple Px64 ports in a conference can implement this feature simultaneously.

The MCU does not support this feature in 2x56/64k conferences due to several factors, including: the availability of both 2x56K and 2x64k conferencing in the core product, rate adaptation, the presumed availability of 56k service for all 64k bandwidths, and the time required to limit the number of calls answered at either 56k or 64k if there is a bandwidth mismatch.

Whenever a 56k or 64k endpoint dials a Px64 (non-BONDed) port, the MCU signals support of only a 1B channel to the endpoint, and it answers just one call. Note that if more than one call is simultaneously placed, the MCU rejects the additional call attempts; this causes some endpoints to drop the entire connection.

The following points also apply to this feature:

- "Low-speed" endpoint does not cause any changes to the SCM of the conference.
- CDR record(s) for the call indicate the bandwidth used by the call (and not the conference bandwidth).
- Because Px64 ports that are used for this feature appear to the system the same as all the other Px64 ports in the conference, adequate resources to hold the conference must be available. The MCU verifies that sufficient resources for the conference bandwidth are available before it allows the conference to be scheduled. Therefore, for example, although two ports of a 384k conference dial in at 1x56, these ports consume two 384k port's worth of system capacity and resources.

NOTE:
BONDed ports support a form of Low-/High-Speed Interworking. For example, an endpoint can place a 1x64 or 2x64 call to a 256k BONDed port and, once the BONDing process times out and enters Transparent Mode, the endpoint becomes an Audio Add-On participant. If the originating endpoint places two calls, the two calls are connected and billed. If only one call is placed, only one call is connected and billed.

Video Switching Mode and Broadcaster Notification

The MCU uses a proprietary ns-command to announce to all the conference endpoints the current video switching mode and broadcaster. The command is also sent to an endpoint whenever the endpoint initiates a See-Me (User Activated Presentation) request that cannot be honored.

Real-Time Conference Status

The **monitor conference** administration command displays, on the MCU Management Terminal, the real-time status of a conference and its associated conferees. Once the command is issued, the display is automatically updated approximately once each minute. Accordingly, this feature is intended for troubleshooting purposes.

Manufacturer and Product Information Exchange

The MCU exchanges ns-caps with each endpoint that joins the conference. Besides information regarding capabilities, ns-cap messages contain manufacturer and product information for the MCU or endpoint. Such information received by the MCU from the endpoint is displayed on the Status Conference form.

Audio Add-On Echo Control

A single Audio Add-On conferee who dials into a conference through an analog circuit may cause all other conferees to hear an echo. Echoes occur at the 2- to 4-wire conversion (hybrid) in the network or in PBXs and are caused by impedance mismatches at those points. These mismatches cause delays and, therefore, echoes that are perceptible to the user. To prevent this problem, an echo canceler function is implemented in the MCU, specifically, within the TN788 circuit pack.

Retry Parameters

An overall call retry counter with the range 0 through 200 is administrable. This counter applies to both pre-answer and post-answer failure cases. In addition, a retry interval with a range from 1 to 5 minutes (in intervals of one minute) is administrable. The overall failure retry counter and the existing post-answer failure counter operate on a per-call basis. The counters for the second channel do not affect the counters for the first channel.

Dedicated Access

This feature enables endpoints to participate in multipoint audio and video conferences via non-signaled T1 or E1 facilities. For this feature, the MCU connects to a Multiplexer (MUX), a Digital Access Crosspoint System (DACS), or to H.320 endpoints via DS1 facilities. A maximum of 20 DS1 connections to the MCU are supported. The following list indicates other points relevant to Dedicated Access:

- 2B and Wideband endpoints are supported.
- All DS0s relevant to an endpoint are located on the same facility and are contiguous.

- Dedicated Access endpoints and switched endpoints interwork within the same conference and at the same bandwidth.
- Dynamic Conference Resizing is supported.
- MCU ports are universal ports (via MCU readministration).
- Audio Add-On, cascaded, and BONDed endpoints can participate in a conference at the same bandwidth as the feature; however, such endpoints must be accessed via switched facilities.
- An SCM upgrade occurs on a downgraded conference consisting of Dedicated Access endpoints and switched endpoints if the downgrade is caused by switched endpoints and if those switched endpoints drop from the conference.

BRI/DCP Direct Connect Interface

This feature allows the user to connect Basic Rate Interface (BRI) or Digital Communications Protocol (DCP) endpoints directly to the MCU without involving a public or private network, PBX, or MUX. The feature allows up to 12 BRI stations or up to four DCP group systems to connect directly to the MCU.

H.261 Annex D Still Image

This feature allows a still image to be transmitted in the video channel of the H.221 format based on procedures specified in H.261 Annex D. A still image can be transmitted at up to four times its CIF resolution. This feature interacts with H.320 terminals that declare H.261 still-image capability in the cap set. An endpoint usually uses this feature in conjunction with the “See-Me” feature (Contributor mode) by first requesting “See-Me” control, then waiting a half a second, sending the still image, and finally relinquishing “See-Me” control. This specific process transmits a complete still image to other endpoints in a Voice-Activated Switching (VAS) conference and avoids disruptions.

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 time on the MCU to be reserved in advance of the scheduled conference date and automatically reserves the conference on the day it is scheduled to begin.

- Paper-Based Scheduling System

Conferences may be arranged 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. With reserved MCU ports, selected locations always have the ability to convene a conference at any time without making a reservation.

In addition, a paper-based scheduling system can be created 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.

Suggested forms for paper-based scheduling are provided in Appendix B, "Paper-Based Scheduling Forms".

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 system features
- Administering international-based parameters
- Administering trunks
- Administering ISDN D-channels
- Administering the ESM
- Administering maintenance alarm terminals
- Administering dial plans
- Administering Meet-Me numbers and MCU-extensions
- Administering cascades
- Administering Dedicated Access endpoints
- Administering Direct Connect endpoints
- Performing system backups
- Checking the status of the MMI circuit packs

NOTE:
Diagnostic and maintenance procedures are described in the *AT&T MultiPoint Control Unit Maintenance Manual*. Conference troubleshooting procedures are provided in the *AT&T MultiPoint Control Unit Reservations Agent Manual*.

Using the MCU Maintenance Terminal

The MCU Maintenance Terminal (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 data to be sent and received from the MCU. These keys and their functionality are described in the following list:

- **Cursor Keys**

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

- **Editing Keys**

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

- **Transaction Keys**

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

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

Logins and Permissions

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

Table 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 MCU-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
Reservations 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

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

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

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

5. The software version is displayed.
6. When the screen displays: *Terminal Type (Enter 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. To log in to the MCU-ST, follow these procedures:

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

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

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

Changing Passwords

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

1. Log in as **cust** on the MCU-MT.
2. The 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 on page 2-3).

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

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

7. Reenter the new password in the *New Password (enter again)* field and press ENTER.
8. Verify that the 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 must be adjusted by one hour twice a year (advance one hour for daylight savings time; back one hour for standard time).

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

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

Set Date and Time Form

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

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

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

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

8. Press ENTER.
9. When the 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, "Px64 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 its ITU-T implementation with untested versions of ITU-T compliant endpoints.

See "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 reservations 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	224/256k	private	1150	A. Snow/619 555-2222

Collecting Endpoint Data

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

- Type of video endpoint (such as, AT&T Group Video System, AT&T Vistium™ Personal Video System, PictureTel™ Group Video System)
- Model name or number (such as Model 400, Model 70)
- Options installed
- Time zone where the video endpoint is located
- Encoding algorithms supported (Px64, proprietary)
- Specific software version (such as 4.1, 1.1)
- Bandwidths supported
- Data capabilities of the system
- Network access (public through direct connection or private through a PBX)
- Network services supported (such as 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.

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

Currently, most video endpoints have the ability to select between originating a 2B-channel 56k call or a 2B-channel 64k call. In general, video endpoints with ISDN-PRI access to the network are not constrained on the types of calls they can make. However, endpoints with robbed-bit signaling (RBS) network access trunks are restricted to originating and receiving only 56k or BONDed 112/128k, 168/192k, 224/256k, or 280/320k calls. Endpoints originating 64k calls are blocked if the MCU has access only to 56k facilities or when all 64k facilities are in use when the call arrives. Currently, 384k calls to the MCU require ISDN-PRI H0 capacity by both the MCU and each participating video endpoint. Also, 128k, 192k, 256k, 320k, 384k, 768k, 1472k, 1536k, and 1920k calls require ISDN-PRI.

Recording Meet-Me Numbers and MCU- Extensions

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

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

The MCU-Extensions Form also includes the bandwidth per channel available with each MCU-extension and the Meet-Me numbers categorized by network type, either private or public. It should be updated whenever a Meet-Me number is added or removed (see "Recording Meet-Me Numbers and MCU- Extensions" on page 2-9 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:

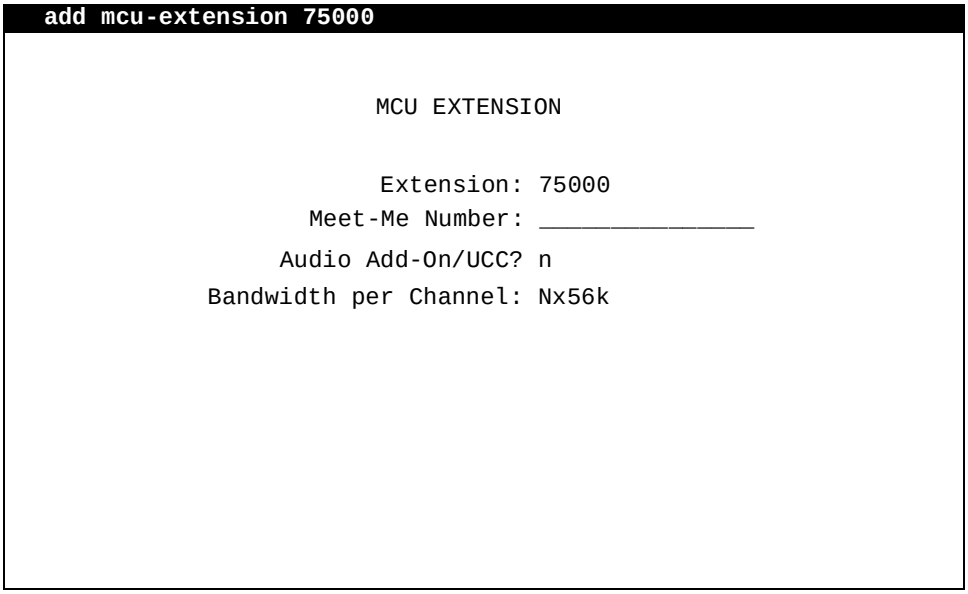

Figure 2-1. MCU-Extension Form for Add/Change/Remove/Display

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 Add-On/UCC?

   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 the Call Detail Recording (CDR) Link

Call Detail Recording (CDR) allows specific information to be recorded about the multimedia conference. To enable CDR, the CDR link must be administered. This is done by populating the *Primary Output Layout* and *Primary Output Destination* fields on page 2 of the Feature-Related System Parameters form and then submitting the form (see the next section for details).

Administering System Features

The Feature-Related System Parameters form sets the parameters for various MCU features. The following table provides the relevant commands.

Table 2-3. System Parameters Features Administration Commands

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

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

1. At the *command:* prompt, type **change system-parameters features** and then press RETURN.
2. Verify that the screen displays the Feature-Related System Parameters screen.
3. From the *AAR/ARS Dial Tone Required* field, use TAB and RETURN to advance to the fields you want to change.
4. When the form is completed, verify that the screen displays: *command successfully completed* followed by *command:*
5. To verify the form, type **display system-parameters features** and then press RETURN.

Features-Related System Parameters Form

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

```
change system-parameters features                          Page 1 of 3

                    FEATURE-RELATED SYSTEM PARAMETERS

              Conference Password Query Timeout (seconds): 60
                 Dial-Out Failure Retry Interval (minutes): 1
                      Dial-Out Overall Failure Retry Limit: __
                  Dial-Out Post Answer Failure Retry Limit: 9

                                 AAR/ARS Dial Tone Required? y









                 Automatic Circuit Assurance (ACA) Enabled? y
                                        ACA Referral Calls: local
                                  ACA Referral Destination: _____
            ACA Short Holding Time Originating Extension: _____
             ACA Long Holding Time Originating Extension: _____
```

Figure 2-3. Feature-Related System Parameters Form (Page 1)

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

Feature-Related System Parameters Form, Page 1 of 3

- *Conference Password Query Timeout (seconds)*

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

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

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

- *Dial-Out Overall Failure Retry Limit*

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

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

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

- *AAR/ARS Dial Tone Required*

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

- Automatic Circuit Assurance (ACA) Enabled

  Enter **y** if ACA measurements are being taken.

- ACA Referral Calls

  Specify where the ACA referral calls are generated—**local**, **remote** or **primary**.

- ACA Referral Destination

  Enter the extension that is to receive the local ACA referral call.

- ACA Short Holding Time Originating Extension

  Enter an unassigned extension number in this field and the long holding time field. Do not enter the same extension number in both this field and in the field described in the next bullet item.

- ACA Long Holding Time Originating Extension

  Enter an unassigned extension number in this field and the short holding time field. Do not enter the same extension number in both this field and in the field described in the preceding bullet item.

Feature-Related System Parameters Form, Page 2 of 3

- Primary Output Layout

  Only a primary CDR output device is allowed. Possible values are **printer** and **expanded** layouts.

- Primary Output Destination

  Enter the extension number assigned to the CDR device. This is the extension of the Modular Processor Data Module (MDPM) or the Electronic Industries Association (EIA) interface (the DCE connector).

- Use ISDN Layouts

  Displayed only when ISDN-PRI is selected on the System Parameters Customer Options form. Enter **y** if the CDR port outputs data in ISDN format.

- EIA Device Bit Rate

  Enter the operating speed of the EIA interface (**300, 1200, 2400, 4800**, or **9600**).

- Disconnect Information in Place of Facility Restriction Level

Enter **y** to allow call disconnect data to be recorded in the place of an FRL data on the CDR report. The call disconnect data is printed on the CDR report instead of the FRL data. An **n** response enables FRL field data to be recorded on the CDR report.

- Suppress CDR for Ineffective Call Attempts

  Enter **y** to suppress recording unsuccessful call attempts.

- *CDR Date Format*

  Shows the date in a numerical “month/day” or “day/month” format.

- Privacy-Digits to Hide

  This field is set to **n**. It is not used.

Feature-Related System Parameters Form, Page 3 of 3

- Printer Extension

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

- EIA Device Bit Rate

  Enter the required printer speed setting (**1200**, **2400**, **4800**, or **9600**).

- Lines Per Page

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

Administering International-Based Parameters

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

Table 2-4. International-Based Parameters Commands

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

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

To administer international-based parameters, follow these procedures:

1. At the *command:* prompt, enter **change *system-parameters country-options***.
2. Verify that the screen displays the *System Parameters Country-Options Form*.
3. Use TAB and RETURN to advance to the fields you want to change. (See "System Parameters Country-Options Form" on page 2-16 for a description of the fields and values.)
4. Submit the form.
5. To verify the form, type **display *system-parameters country-options***.

System Parameters Country-Options Form

The System Parameters Country-Options form appears as follows:

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


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


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

Figure 2-4. 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 2-5. 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** (UK or Australia) or **Mu-law** (US or Canada) to identify the companding algorithm to be used by system hardware. Default is **Mu-law**.

- *Base Tone Generation Set*

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

- *440Hz MCU-Dial Tone*

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

- *440Hz Secondary-Dial Tone*

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

- *Digital Loss Plan*

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

- *Version of Digital Loss Plan*

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

The next two fields control tone detection on the system:

- *Tone Detection Mode*

  Enter a code (either **1** for the US, **2** for Australia, or **3** for the UK) to specify the type of tone detection used on a Tone Detector circuit pack. Default is **1**.

- *Dial Tone Validation Timer*

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

- *Interdigit Pause*

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

- *Tone Name*

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

Acceptable keywords include the following:

— 1-call-wait
— 2-call-wait
— 3-call-wait
— busy
— 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 T1 or E1 trunks. Based on the number of

ISDN-PRI trunks configured, ISDN-PRI NFAS is supported for T1 trunks but not for E1 trunks, and D-channel backup features are supported 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 RBS.

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

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

- **ISDN-PRI**

  When the DS1 (UDS1) interface provides ISDN-PRI trunk group service, a TN767E DS1 (TN2207 UDS1) circuit pack 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 30 56k or 64k trunks for video transmission and one trunk for signaling.

- **DS1 tie trunks**

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

Administering DS1 or UDS1 Circuit Packs

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

Table 2-5. DS1 (UDS1) Circuit Pack Administration Commands

Action	Object	Qualifier
add	ds1 (uds 1)	location*
change	ds1 (uds 1)	location*
display	ds1 (uds 1)	location*
list	measurements ds1 (uds 1)	location*
remove	ds1 (uds 1)	location*

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

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

1. At the *command:* prompt, enter **add ds1** ***location*** where ***location*** is the location of the TN767E DS1 or TN2207 UDS1 circuit pack in the carrier.
2. Verify that the screen displays the *DS1 Circuit Pack Form*.
3. Use TAB and RETURN to advance to the fields you want to change (see "DS1 (UDS1) Circuit Pack Form" on page 2-22 for a description of the fields and values).
4. Submit the form.
5. To verify the form, type **display ds1** ***location***.

DS1 (UDS1) Circuit Pack Form

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

```
change ds1 a15                                              Page 1 of 1
                              DS1 CIRCUIT PACK
             Location: 1A15                              Name: __________
             Bit Rate: 2.048                    Line Coding: hdb3

       Signaling Mode: isdn-pri
              Connect: pbx                          Interface: user
   MMI Cabling Board: 1A10                 Country Protocol: 1
       MMI Interface: network
 Interface Companding: mulaw                             CRC? n
            Idle Code: 11111111



                           MAINTENANCE PARAMETERS
      Slip Detection? n             Remote Loop-Around Test? n
```

Figure 2-6. DS1 Circuit Pack Form (Option without ESM)

```
change ds1 a15                                              Page 1 of 1
                              DS1 CIRCUIT PACK
             Location: 1A15                              Name: __________
             Bit Rate: 2.048                    Line Coding: hdb3

       Signaling Mode: isdn-pri
              Connect: pbx                          Interface: user
   MMI Cabling Board: 1A10                 Country Protocol: 1
       MMI Interface: ESM
 Interface Companding: mulaw                             CRC? n
            Idle Code: 11111111

                           MAINTENANCE PARAMETERS
      Slip Detection? n             Remote Loop-Around Test? n

                              ESM PARAMETERS
      Sanity Timeout: 30                          Run State: 2
     Startup Timeout: 120
```

Figure 2-7. DS1 Circuit Pack Form (Option with ESM)

```
change ds1 a14                                              Page 1 of 1
                              DS1 CIRCUIT PACK
              Location: 1A14                          Name: __________
              Bit Rate: 1.544                 Line Coding: b8zs
     Line Compensation: 1                    Framing Mode: esf
        Signaling Mode: robbed-bit

     MMI Cabling Board: 1A11
         MMI Interface: network


             Idle Code: 11111111


                           MAINTENANCE PARAMETERS

        Slip Detection? n            Remote Loop-Around Test? n
```

Figure 2-8. DS1 Circuit Pack Form (Option with RBS)

```
change ds1 a16                                              Page 1 of 1
                              DS1 CIRCUIT PACK
              Location: 1A16                          Name: _______________
              Bit Rate: 2.048                 Line Coding: hdb3

        Signaling Mode: isdn-ext_____


     MMI Cabling Board: 1A12
         MMI Interface: network
  Interface Companding: mulaw
             Idle Code: 11111111


                           MAINTENANCE PARAMETERS

        Slip Detection? n            Remote Loop-Around Test? n
```

Figure 2-9. DS1 Circuit Pack Form (Option with External D-Channel)

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 TN767E 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** (US) is the default value. Other current reserved values are **2** (Australia), **4** (5ESS), and **10** (United Kingdom).

- *MMI Interface*

  Possible values include **ESM** and **network** (default). **ESM** indicates that the DS1 circuit pack is cabled to the ESM-MMI circuit pack, while **network** indicates that the DS1 circuit pack is cabled to the H.221-MMI circuit pack.

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

  - "ESM PARAMETERS" section (which includes *Sanity Timeout, Startup Timeout*, and *Run State*) appears at the bottom of the form.
  - *Bit Rate* must contain **2.048**.
  - *Line Coding* must contain **hdb3**.
  - *Signaling Mode* must contain **isdn-pri**.
  - *Connect* must contain **pbx**.
  - *Interface* must contain **network**.
  - MMI Cabling Board must be administered.

- *CRC*

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

- *Sanity Timeout*

  This field indicates the amount of time that the Remote Maintenance Board (RMB), if functional within the system, allows between the receipt of sanity messages. The value must be specified in 10-second increments and range between **30** (default) and **300** seconds. If the timeout expires, the RMB reboots the ESM.

- *Startup Timeout*

  This field indicates the amount of time the RMB waits after a reboot before the first sanity message must be received. The value must be specified in 10-second increments and range between **0** and **600** seconds. Default is **120**.

- *Run State*

  This field indicates whether the system is active or idle. Values include **0** through **9.** Default is **2**. System run states also control system rebooting and reinitialization.

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.

The following table provides the commands that relate to the synchronization plan.

Table 2-6. Synchronization Plan Administration Commands

Action	Object	Qualifier
change	synchronization	
display	synchronization	print or schedule

To change the synchronization assignments, follow these procedures:

1. At the *command:* prompt, enter **change synchronization**.
2. Verify that 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 2-28 for a list of fields and values).
4. Submit the form.
5. To verify the form, enter **display synchronization**.

Synchronization Plan Form

The Synchronization Plan form appears as follows:

```
change synchronization                                    Page 1 of 2

                          SYNCHRONIZATION PLAN
               SYNCHRONIZATION SOURCE (DS1 circuit pack location)

                     Stratum: 4
                     Primary: 1A15       Secondary:

                              DS1 CIRCUIT PACKS

          Location  Name   Slip       Location  Name   Slip
          1A15   n________  _         ____    ________  _
          1A16   n________  _         ____    ________  _
          1A14   n________  _         ____    ________  _
          ____   _________  _         ____    ________  _
          ____   _________  _         ____    ________  _
          ____   _________  _         ____    ________  _
          ____   _________  _         ____    ________  _
          ____   _________  _         ____    ________  _

   NOTE: TN722 DS1 source will result in stratum 4, type II synchronization
```

Figure 2-10. 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.

  NOTE:
  An ESM DS1 circuit pack (if present) should not be administered 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.

  NOTE:
  An ESM DS1 circuit pack (if present) should not be administered as the secondary synchronization source.

- *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 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 an ISDN-PRI or ISDN-BRI 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**.

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

Table 2-7. 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

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

1. At the *command:* prompt, enter **add data-module *xxxx***.
2. Verify that the screen displays the *Data Module* form (see "Processor Interface Data Module Form" on page 2-31 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 Processor Interface Data Module form appears as follows:

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

                                  DATA MODULE

Data Extension: 31000        Type: procr-intf        Physical Channel:
          Name: _____________  COS: 1                              COR: 1
            Maintenance Extension:









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

    Ext     Name
 1: ____    _______
```

Figure 2-11. 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*

  To provide ISDN-BRI connectivity via an ISDN-BRI data module, enter **7500**. Otherwise, enter **procr-intf**.

  NOTE:
  To enable ISDN-BRI connectivity via this field, the *Max. BRI Direct Connect Capacity* field in the Optional Features form must contain "12".

- *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. Default is **1**.

- *COR*

  Enter the desired Class of Restriction. 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** and then enter **y** in the *Enable* field).

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

Table 2-8. Processor Interface Links Administration Commands

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

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

1. At the *command:* prompt, enter **change communication-interface links**.
2. Verify that the screen displays the *Processor Interface Links* form (see "Processor Interface Links Form" on page 2-33 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**.

Processor Interface Links Form

The Processor Interface Links form appears as follows:

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

                              INTERFACE LINKS

                 Est    PI                Destination      DTE/
Link   Enable    Conn   Ext     Prot      Digits      Brd    DCE    Identification
 1:      y       y    70000    ISDN     1A15_____  _____   ___    pri to v5d2___
 2:      n       n    _____    BX25     DTE______  _____   ___    ______________
 3:      n       n    _____    BX25     DTE______  _____   ___    ______________
 4:      n       n    _____    BX25     DTE______  _____   ___    ______________
 5:      n       n    _____    BX25     DTE______  _____   ___    ______________
 6:      n       n    _____    BX25     DTE______  _____   ___    ______________
 7:      n       n    _____    BX25     DTE______  _____   ___    ______________
 8:      n       n    _____    BX25     DTE______  _____   ___    ______________
```

Figure 2-12. Processor 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.

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

- *Est Conn*

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

- *PI Ext*

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

- *Prot*

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

- *Digits*

  Leave this field blank.

- *Destination Brd*

  Enter the TN767E DS1 or TN2207 UDS1 circuit pack address that has been administered as using ISDN signaling.

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

- *DTE/DCE*

  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.

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

Table 2-9. Processor Channels Administration Commands

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

To change the Processor Channels form, follow these procedures:

1. At the *command:* prompt, enter **change communication-interface processor-channels**.
2. Verify that the screen displays the *Processor Channels* form (see "Processor Channel Assignment Form" on page 2-35 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 Channel Assignment Form

The Processor Channel Assignment form appears as follows:

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

                        PROCESSOR CHANNEL ASSIGNMENT

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

Figure 2-13. Processor Channel Assignment Form

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

- *Proc Chan*

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

- *Appl*

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

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

- Only one entry per form is permitted.
- Entry cannot be administered if there are no MCS/MLP ports administered on the System-Parameters Customer Options form.
- *Channel* and *Remote Proc Chan* fields must be blank.
- *Priority* field must be set to **high**.
- DS1 circuit pack administered in the *Destination Brd* field of the Interface Links form should be an ESM DS1 circuit pack.

- *Link*

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

- *Chan*

  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 Non-Facility Assocated Signaling (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 E1 trunk.)

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

Table 2-10. 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

To specify a signaling group, follow these procedures:

1. At the *command:* prompt, enter **add signaling-group *x*** where ***x*** is the signaling group number (**1** through **8**) or **next**.
2. Verify that the screen displays the *Signaling Group* form (see "Signaling Group Form" on page 2-38 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

Two variations of the Signaling Group form appear as follows:

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

                              SIGNALING GROUP

  Group Number: 2    Associated Signaling? y      Max number of NCA TSC: 0
                         Primary D-Channel: _____  Max number of CA TSC: 0
                                                   Trunk Group for NCA TSC: _
```

Figure 2-14. Signaling Group Form (Option without NFAS)

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

                              SIGNALING GROUP

 Group Number: 2      Associated Signaling? n     Max number of NCA TSC: 0
                         Primary D-Channel: _____  Max number of CA TSC: 0
                       Secondary D-Channel:     Trunk Group for NCA TSC: _

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

Figure 2-15. Signaling Group Form (Option with NFAS)

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

- *Group Number*

  This is a display-only field that provides the signaling group number used as a qualifier in the 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 all E1 trunks.)

- *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.

  NOTE:
  An ESM DS1 port (if present) should not be administered as a trunk 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.

  NOTE:
  An ESM DS1 port (if present) should not be administered as the secondary D-Channel.

- *Trunk Group for NCA TSC*

  This field is not used.

- *Trunk Brd*

  When using NFAS, enter a four-character DS1 interface circuit pack number that has trunk members belonging to this Signaling Group. The four characters are the port network number, carrier letter, and two-digit slot number (**01** through **18**). The default is blank.

  NOTE:
  An ESM DS1 circuit pack (if present) should not be administered as the primary D-Channel.

- *Interface ID*

  When using NFAS, enter an interface ID (**0** through **31**) for the corresponding DS1 interface circuit pack. The interface ID is referenced by both ends to determine which B-channel is to be used to establish a call; therefore, the interface ID numbers on both ends must be the same. The 4ESS™ switch must start with *Trunk Brd* 1 (which should be the primary D-channel) ID 0. If a secondary D-channel is used, it should be *Trunk Brd* 2 ID 1.

  NOTE:
  Pages 2 through 5 are not used.

  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. The following table shows the commands that are used to administer trunk groups using the Trunk Group form.

Table 2-11. 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

To add an ISDN or tie trunk group, follow these procedures:

1. At the *command:* prompt, enter **add trunk group *x*** where ***x*** is a trunk group number (**1** through **64**) or **next**.
2. Verify that the screen displays the Trunk Group form (see "Administering ISDN-PRI Trunk Groups" on page 2-41 or "Administering Tie Trunk Groups" on page 2-45 sections 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.

- *CDR Reports*

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

- *Group Name*

  Enter a name of 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. CDR uses the TAC to identify the trunk group on 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. 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 the alarm goes off). 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 or 1536k bandwidth, set this field to **y**. (When Wideband is supported, page 3 of this form needs to be completed.)

- *Data Restriction*

  Leave this field set to **n**. 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.

  NOTE:
  ESM DS1 ports (if present) should not be administered as members of any trunk group. These ports are hard-cabled to the ESM MMI ports.

- *Name*

  Enter the name of the member of the trunk group (up to 10 characters). The name can be the 7-digit telephone number or a 10-digit trunk circuit identification number (provided by the trunk supplier).

- *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 that shows the trunk group number.

- *Group Type*

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

- *CDR Reports*

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

- *Group Name*

  Enter a name of 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 TAC that must be dialed to access the trunk group. A different TAC must be assigned to each trunk group. CDR uses the TAC to identify the trunk group on 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 CAS 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. 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. 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 the alarm goes off). 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.

- *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, this field must be completed manually.

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

Administering the ESM from the MSM

To administer the ESM from the MSM, you must first establish the MSM-ESM link and then enable MCS/MLP data transfer within the conference. To complete these steps, you must access and populate several forms from the MCU-MT.

Establish the MSM-ESM link as follows:

1. Verify that the time setting on the MSM is correct by entering the **display time** command. If you need to correct the time, enter the **set time** command and complete the fields in the Set Date and Time form.
2. Access the System-Parameters Customer-Option form by entering **change system-parameters customer-options**. Enter the maximum number of MCS/MLP ports that can be allocated in the system at the same time into the *Max. MCS/MLP Port Capacity* field. Then submit the form.
3. Access the DS1 (UDS1) Circuit Pack form by entering **change uds1 *location***, where ***location*** is the location of the UDS1 circuit pack in the carrier. Ensure that the MMI cabling board is administered, and populate each field in the following list with the value specified:
   - *Bit Rate*: **2.048**
   - *Line Coding*: **hdb3**
   - *Signaling Mode*: **isdn-pri**
   - *Connect*: **pbx**
   - *Interface*: **network**
   - MMI Cabling Board: (location of this item; for example, **1A10**)
   - MMI Interface: **ESM**

   Submit the form.
4. Access the Interface Links form by entering **change communication-interface links**. Then populate each field in the following list with the value specified.
   - *Enable*: **n** (for now!)
   - *Est Conn*: **y**
   - *Prot*: **ISDN**
   - *Destination Brd*: (address of the relevant DS1 or UDS1 circuit pack)

   Submit the form.

5. Access the Processor Channel Assignments form by entering **change communication-interface processor-channels.** Populate each field in the following list with the value specified.
   - *Link:* (link number administered in the Interface Links form in the previous step)
   - *Appl:* **ESM**
   - *Channel*: blank
   - *Remote Proc Chan*: blank
   - *Priority*: **high**

   Submit the form.
6. Access the Signaling Group form by entering **add signaling-group *x***, where ***x*** is either the signaling group number (**1** through **8**) or **next**. Ensure that channel 16 is represented in the *Primary D-Channel* field (for example, **A0816**, where A08 is the location of the UDS1 circuit pack). Then complete the form according to the directions provided in the "Signaling Group Form" on page 2-38. Finally, submit the form.
7. Access once again the Interface Links form by entering **change communication-interface links.** Then enable the link by setting the *Enable* field to **y**. Finally, submit the form.

To enable MCS/MLP data transfer within the conference, access the Conference Record by entering **change conference *xxx***, where ***xxx*** is the conference number. Then populate each field in the following list with the value specified.

- *Audio Mode*: **ld-celp**
- *App Comp?*: **y** (for each endpoint)

Submit the form.

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 2-12. Station Administration Commands

Action	Object	Qualifier
add	station	extension or next
change	station	extension
display	station	extension
list	station	extension, type, count, port, print, or schedule
remove	station	extension

To add a maintenance alarm terminal, follow these procedures:

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

Station Form

Page 1 of the Station form appears as follows:

```
add station next                                          Page 1 of 4

                                    STATION

Extension: 31000
     Type: 7444D                                              COR: 1
     Port: _____________________                              COS: 1
     Name: _____________________
FEATURE OPTIONS




     Data Module? n
  Display Module? y
```

Figure 2-16. Station Form (Page 1)

The following list shows the fields on page 1 the Station form and their related values.

- *Extension*

  A display-only field that shows the extension number of the 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

```
add station next                                          Page 2 of 4

                                   STATION

NON-SWITCH DATA
     Room: _________                                    Headset? n
     Jack: _____
    Cable: _____




BUTTON ASSIGNMENTS
     1: call-appr                                  6: _________
     2: call-appr                                  7: _________
     3: call-appr                                  8: _________
     4: _________                                  9: _________
     5: _________                                 10: _________
```

Figure 2-17. Station Form (Page 2)

- *Room*

  Specify the room number where this maintenance alarm terminal is located.

- *Headset*

  This field is not used. Leave it set to **n**.

- *Jack*

  Specify the jack number where the 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. The following table lists the recommended maintenance alarm terminal button assignments for buttons 1 through 10.

Table 2-13. Maintenance Alarm Terminal Button Assignments (1 through 10)

Button	Assignment	Button	Assignment
1	call-appr	6	ac-alarm
2	call-appr	7	cdr1-alrm
3	call-appr	8	pr-sys-alm
4	lsvn-call	9	trk-ac-alm
5	aca-call	10	verify

Station Form, Page 3

```
add station next                                          Page 3 of 4

                                    STATION

BUTTON ASSIGNMENTS
  11: _________                      12: _________

  13: _________                      24: _________
  14: _________                      25: _________
  15: _________                      26: _________
  16: _________                      27: _________
  17: _________                      28: _________
  18: _________                      29: _________
  19: _________                      30: _________
  20: _________                      31: _________
  21: _________                      32: _________
  22: _________                      33: _________
  23: _________                      34: _________
```

Figure 2-18. Station Form (Page 3)

- *Button Assignments*

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

Table 2-14. 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	

Station Form, Page 4

```
add station next                                          Page 4 of 4

                                    STATION

DISPLAY BUTTON ASSIGNMENTS

1: _________
2: _________
3: _________
4: _________
5: _________
6: _________
7: _________
```

Figure 2-19. Station Form (Page 4)

- *Display Button Assignments*

  Assign the display button appearances to the appropriate buttons as shown in the following table.

Table 2-15. 7444D Display Button Assignments

Button	Assignment
1	normal
2	inspect
3	trk-id
4	trunk-name
5	date-time
6	timer
7	

Station Form, Page 5

```
add station next                                          Page 5 of 5

                                    STATION

DATA MODULE
     Data Extension: ____                  BCC: 2        ITC: restricted
               Name: _________________     COR: 1        COS: 1









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

Figure 2-20. Station Form (Page 5)

NOTE:
This screen displays only when **y** is entered in the *Data Module* field on the first screen.

- *Data Extension*

  Enter the extension assigned to the data module.

- *BCC*

  Leave this field set to **2.**

- *ITC*

  Information Transfer Capability used for routing calls according to the endpoint's ITC. Default is **restricted**.

- *Name*

  Optional name for identification.

- *COR*

  The Class of Restriction assigned to this extension.

- *COS*

  The Class of Service assigned to this extension.

- *Ext Name*

  The terminal with a data extension button for this data module.

Administering Dial Plans

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

To administer AAR, follow these procedures:

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 associated with the entries on the same line of the form.
16. In the corresponding space within the column headed by FRL, enter the Facility Restriction Level (FRL) to be assigned to the line of parameters on the form. Choices range from "0" (least restrictive) to "7" (most restrictive).
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 or 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 calls to be routed over the public network according to the preferred (usually the least expensive) route available at the time you place the call. Before the necessary forms can be administered to enable ARS, the ARS option on the System-Parameters Customer-Options form must be enabled.

To administer ARS, follow these procedures:

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 associated with the entries on the same line of the form.
17. In the corresponding space within the column headed by FRL, enter the Facility Restriction Level (FRL) to be assigned to the line of parameters on the form. Choices range from “0” (least restrictive) to “7” (most restrictive).
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 trunk groups that support bandwidths of 384k or higher, 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 or 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 the forms to enable UDP can be administered, the optional UDP feature must be enabled on the System-Parameters Customer-Options form.

To administer UDP, follow these procedures:

1. At the prompt, enter *change dialplan* to access the Dial Plan Form.

2. On Page 1, enter “y” in 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 associated with the entries on the same line of the form.
18. In the corresponding space within the column headed by FRL, enter the Facility Restriction Level (FRL) to be assigned to the line of parameters on the form. Choices range from “0” (least restrictive) to “7” (most restrictive).

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 or 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, and Connections

Dial-in 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).

NOTE:
As an alternative, the MCU can dial out calls to the appropriate endpoints. A Dial-Out call can use two different telephone numbers per endpoint.

The number of MCU ports is determined by the software administration of the 2B, 128k, 192k, 256k, 320k, 384k, 768k, 1472k, 1536k, 1920k, BONDed 112/128k, 168/192k, 224/256k, 280/320k, and 336/384k, UCC and MCS/MLP MCU ports and the hardware resources in the MCU. The MCU is shipped with a configuration of hardware resources appropriate for the model ordered.

As calls arrive at the MCU trunks, the network sends the dialed digits across the MCU access trunk. The Meet-Me number dialed is mapped to an MCU-extension, which in turn enables the MCU to allocate available resources for the call. If the incoming call is from a 2B-channel 56k/64k video endpoint, the MCU actually receives two incoming network calls for the same MCU-extension. On the other hand, if the incoming call is from a BONDed 336 (6 x 56k) video endpoint, the MCU receives six incoming calls for the same MCU-extension. Whatever the number of calls, the calls are joined by the MCU once they are placed, and the MCU assumes that they come from the same video endpoint.

An important function of call routing is the ability to manipulate the digits. This capability provides great flexibility when assigning numbers to users and connecting users to conferences. Digit manipulation can be done through the network, on the PBX (if the MCU is behind a PBX), on the MCU trunk group or in the MCU call routing table. This optional capability allows the MCU to handle any digit string matching the North American dial plan, a private network dial plan, or any multinational dial plan.

The Dedicated Access feature enables endpoints to participate in multipoint audio and video conferences via non-signaled T1 facilities. For this feature, the MCU connects to a multiplexer (MUX), a Digital Access Crosspoint System (DACS), or to H.320 endpoints via DS1 facilities. A maximum of 20 DS1 connections to the MCU are supported.

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.

Finally, Low-Speed/Higher-Speed Interworking allows 56k and 64k endpoints to join a conference of a higher bandwidth as Audio Add-On endpoints.

Network Services

The MCU is viewed as an endpoint by the public network. If the MCU resides behind a PBX, the PBX serves as the public network endpoint. Endpoint users must subscribe to a network service that provides a digital transmission service.

The following table provides an overview of the most commonly used digital services provided by AT&T.

Table 2-16. 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 1B*, 2B^, 1-channel multirate+, BONDed# 1B*, 2B^, 1-channel multirate+, BONDed#	RBS network access ISDN-PRI% ISDN-PRI%
ACCUNET Switched Digital Service (SDS) SW56	700-56x-xxxx	56k	RBS network access
ACCUNET SDS SW56/64	700-737-xxxx	1B*, 2B^, 1-channel multirate+, BONDed#	ISDN-PRI%
Software Defined Data Network (SDDN)	NNN-NNNN Nxx-xxxx N0/1x-Nxx-xxxx	56k 1B*, 2B^, 1-channel multirate+, BONDed# 1B*, 2B^, 1-channel multirate+, BONDed#	RBS network access ISDN-PRI% ISDN-PRI%
SDDN-Virtual Off-Net	700-56x-xxxx 700-737-xxxx	56k 1B*, 2B^, 1-channel multirate+, BONDed#	RBS network access ISDN-PRI%
Private Network (PBX)	1 to 7 digits		Based on PBX dial plan connecting to the MCU

x = any number from 0 through 9
N = any number from 2 through 9
* 56k, 64k
^112k, 128k
+128k, 192k, 256k, 320k, 384k, 768k, 1472k, 1536k, or 1920k
112/128k, 168/192k, 224/256k, 280/320k, or 336/384k
% NonBONDed

Administering the Dial Plan

Through administration of the dial plan, the MCU identifies calls from the network and routes them to available video ports. This is done through the Dial Plan Record form.

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

The commands in the following table can be used to access the Dial Plan Record form.

Table 2-17. Dial Plan Administration Commands

Action	Object	Qualifier
change	dialplan	
display	dialplan	print or schedule

To change the dial plan, followl these procedures:

1. At the *command:* prompt, enter **change dialplan**.
2. Verify that the screen displays the *Dial Plan Record* form (see "Dial Plan Form" on page 2-67 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

Page 1 of the Dial Plan form appears as follows:

```
change dialplan                                          Page 1 of 6


                                DIAL PLAN RECORD
   North American Area Code: 908                 Local PBX ID: __
      ARS Prefix 1 Required? y
          Uniform Dial Plan? y                    Plan Length: 5

FIRST DIGIT TABLE
First                                Length
Digit   -1-        -2-        -3-        -4-        -5-        -6-
  1: _________  _________  _________  _________  _________  _________
  2: _________  _________  _________  _________  _________  _________
  3: _________  _________  _________  _________  _________  _________
  4: _________  _________  _________  _________  _________  _________
  5: _________  _________  _________  _________  _________  _________
  6: _________  _________  _________  _________  _________  _________
  7: _________  _________  _________  _________  _________  _________
  8: _________  _________  _________  _________  _________  _________
  9: _________  _________  _________  _________  _________  _________
  0: _________  _________  _________  _________  _________  _________
  *: _________  _________  _________  _________  _________  _________
  #: _________  _________  _________  _________  _________  _________
```

Figure 2-21. Dial Plan Form (Page 1)

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

Dial Plan Form, Page 1 of 6

- *Area Code*

  Enter the area code number where the MCU is located.

  NOTE:
  Leave this field blank for an MCU in the United Kingdom or Australia.

- *Local PBX ID*

  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 between 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. Network numbers can be obtained from Network Services or, if the MCU is an SDDN or private network location, the Meet-Me numbers can be selected and assigned according to the existing dial plan. After the Meet-Me numbers are obtained, each number needs to be associated with an MCU-extension on the MCU.

NOTE:
Whenever possible, the last four digits of the MCU-extension should be the same as the last four digits of the corresponding Meet-Me number. This helps with troubleshooting so you can quickly relate a dialed Meet-Me number with the matching MCU-extension on the MCU. It is important to note, however, that an MCU-extension cannot begin with a **0**.

The following table provides the commands that can be used to administer MCU-extensions.

Table 2-18. MCU-Extension Administration Commands

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

NOTE:
An MCU-extension cannot be removed if it is assigned to a conference reserved on the MCU.

To add an MCU-extension, follow these procedures:

1. At the *command:* prompt, enter **add mcu *xxxx*** where ***xxxx*** is an extension number or **next**.
2. Verify that the screen displays the *MCU-Extension* form (see "MCU-Extension Form" on page 2-70 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 MCU-Extension form appears as follows:

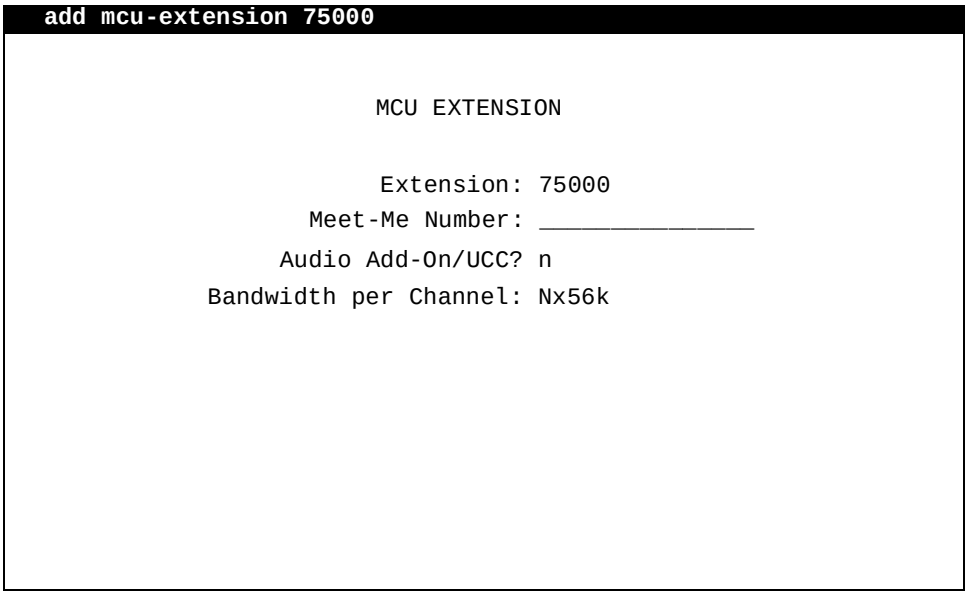

Figure 2-22. MCU-Extension Form

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 Add-On/UCC*

  Enter **yes** or **no.** The default is **no**.

- *Bandwidth per Channel*

  Enter the bandwidth that the Meet-Me Number supports.

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

NOTE:
If *Audio Add-On/UCC* (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. One MCU can be programmed to set up a link with another MCU by populating the appropriate fields in several forms. The resulting MCU-MCU link is called a cascade link. Once the cascade link is established, it is maintained by one of the MCUs in the cascaded MCU pair for the duration of the conference.

To enable cascading, one MCU must be established as the dial-out MCU and the other MCU as the dial-in MCU. Also, one MCU must be established as the primary MCU and the other MCU as the secondary MCU. 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 to establish a cascade link. The MCU that launches the call is called the dial-out MCU. The MCU that receives the call is called the dial-in MCU. Both MCU types can be established by populating the appropriate fields on page 2 of the Conference Record form for each MCU.

To establish the dial-out MCU, do the following:

1. Access the Conference Record form for the MCU that is to serve as the dial-out MCU.
2. On page 2 of the form, type “CAS” (cascade) or “BCAS” (BONDed cascade) in the first *Type* field (for Conferee 1).
3. In the *Dial-out # 1* field for Conferee 1, type a number that reaches the dial-in MCU. If this is a 2B conference, type the same number in the corresponding *Dial-out # 2* field**.**

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

```
change conference 123                                    page  2 of 8

                              CONFERENCE RECORD
CONFEREES
1          Type: CAS_ MCU-Ext: 5000_ Meet-Me Number: xxxxxxxxxxxxxx
         Redial? _   App Comp? _          Basic/Enh: Basic
   Dial-Out # 1: 705-555-1234_________ Dial-Out # 2: 705-555-1234______
  Password Reqd? _

2          Type: P64_ MCU-Ext: 5001_ Meet-Me Number: xxxxxxxxxxxxxx
         Redial? _   App Comp? _          Basic/Enh: Basic
   Dial-Out # 1: ____________________ Dial-Out # 2: ________________
  Password Reqd? _

3          Type: P64_ MCU-Ext: 5002_ Meet-Me Number: xxxxxxxxxxxxxx
         Redial? _   App Comp? _          Basic/Enh: Basic
   Dial-Out # 1: ____________________ Dial-Out # 2: ________________
  Password Reqd? _

4          Type: P64_ MCU-Ext: 5003_ Meet-Me Number: xxxxxxxxxxxxxx
         Redial? _   App Comp? _          Basic/Enh: Basic
   Dial-Out # 1: ____________________ Dial-Out # 2: ________________
  Password Reqd? _
```

Figure 2-23. Conference Record Form for a Dial-Out MCU (Page 2)

4. Save the Conference Record form.

NOTE:
If you access the Conference Record form by entering the **add conference** command (as is appropriate for administering cascading), the *Meet-Me Number* field is blank. However, if you access the form by entering the **change conference** command, and if the Meet-Me Number(s) corresponding to the appropriate MCU-extension are administered on the MCU-Extension form, the Meet-Me Number(s) appear(s) within the *Meet-Me Number* field.

To establish the dial-in MCU, do the following:

1. Access the Conference Record form for the MCU that is to serve as the dial-in MCU.
2. On Page 2 of the form, type “CAS” (cascade) or “BCAS” (BONDed cascade) in the first *Type* field (for Conferee 1).
3. In the *MCU-Ext* field for Conferee 1, type the extension (usually the last four digits) of one of the CAS- or BCAS-type dial-out numbers assigned to the corresponding dial-out MCU.

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

```
change conference 123                                    page  2 of 8

                              CONFERENCE RECORD
CONFEREES
1          Type: CAS_ MCU-Ext: 1234_ Meet-Me Number: xxxxxxxxxxxxxx
         Redial? _   App Comp? _          Basic/Enh: Basic
   Dial-Out # 1: _____________________ Dial-Out # 2: ____________________
  Password Reqd? _

2          Type: P64_ MCU-Ext: 6001_ Meet-Me Number: xxxxxxxxxxxxxx
         Redial? _   App Comp? _          Basic/Enh: Basic
   Dial-Out # 1: _____________________ Dial-Out # 2: ____________________
  Password Reqd? _

3          Type: P64_ MCU-Ext: 6002_ Meet-Me Number: xxxxxxxxxxxxxx
         Redial? _   App Comp? _          Basic/Enh: Basic
   Dial-Out # 1: _____________________ Dial-Out # 2: ____________________
  Password Reqd? _

4          Type: P64_ MCU-Ext: 6003_ Meet-Me Number: xxxxxxxxxxxxxx
         Redial? _   App Comp? _          Basic/Enh: Basic
   Dial-Out # 1: _____________________ Dial-Out # 2: ____________________
  Password Reqd? _
```

Figure 2-24. 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 and the other as the secondary MCU. This is done to make the MCU feature interactions as transparent as possible to the user. Generally, this administration can be performed to accommodate the requirements at the customer site.

To establish a primary MCU and a secondary MCU, do the following:

1. Access the Conference Record form for the MCU that is to serve as the primary MCU, 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

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, the secondary MCU must be configured by accessing the Conference Record form for this MCU, typing the MCU-extension of the cascade link in the *Control Ext* field on page 1 of the form, and then submitting the form.

Administering Dedicated Access Endpoints

The Dedicated Access feature allows endpoints to participate in multipoint conferences via non-signaled T1 or E1 facilities. Dedicated Access endpoints can be administered by populating and submitting a couple of forms. Each endpoint that is administered must have either the same bandwidth as the conference being joined or a higher bandwidth. For each endpoint that you wish to administer as a Dedicated Access endpoint, do the following:

1. From the console, bring up the "Access Endpoint" form by entering either **add access-endpoint *xxxx*** or **change access-endpoint *xxxx***, where ***xxxx*** is the endpoint number. The form appears as follows:

```
add access endpoint 4561                                  Page 1 of 1

                            ACCESS ENDPOINT

         Extension: _____              (Starting) Port: _______
Communication Type: wideband                      Name: __________
             Width: 6
               COR: 1_                             COS: __
               ITC: restricted
```

Figure 2-25. Access Endpoint Form

2. Once the form appears, complete (where necessary) the following fields:

   - **Extension**

     Display-only field that shows the extension number specified in the command line. This extension number is assigned to the nonsignaling trunk, and it is used to access the trunk endpoint.

   - **(Starting) Port**

     Enter the appropriate seven values according to the breakdown in the following table.

Character Position	Meaning	Value
1-2	Cabinet number	01
3	Carrier	A through E
4-5	Slot number	01 through 20
6-7	Circuit Number	01 through 31 (DS1 Interface ports)

The DS1 circuit number corresponds to the channel that carries the data traffic. Channels 1 through 30 (for the UDS1 circuit pack only) or channels 1 through 24 (for the DS1 or UDS1 circuit pack) may be used when the DS1 signaling type is "robbed-bit." For Common Channel Signaling (CAS) or ISDN-PRI signaling, Channels 1 through 30 (for the UDS1 circuit pack only) or channels 1 through 23 (for the DS1 or UDS1 circuit pack) may be used. A channel can be administered as an access endpoint regardless of the DS1 signaling type.

NOTE:
The UDS1 circuit pack is required to administer Wideband Dedicated Access endpoints.

- **Communication Type**

  For a DS1 access endpoint, enter either **56k-data**, **64k-data**, or **voice-grade-data**, as appropriate (64k-data is not permitted for robbed-bit trunks). Default is **voice-grade-data**. For a Wideband access endpoint, enter **wideband**.

- **Name**

  Enter up to ten alphanumeric characters to identify the endpoint. Default is blank.

- **Width**

  This field is displayed only if *Communication Type* contains "wideband." Beginning with the specified Starting Port, enter the number of adjacent DS0 ports (2 to 31) that comprise the Wideband Access Endpoint (WAE). A value of **6** (default) defines a 384k WAE. This field cannot be blank.

- **COR**

  Enter the appropriate class of restriction (COR) number (**0-95**). The COR is administered to enable only an administered connection endpoint to be connected to another administered connection endpoint. Default is **1**.

- **COS**

  Enter the appropriate class of service (COS) number (**0-15**). The COS is administered to prevent the use of the Call Forwarding All Calls feature for access endpoints.

- **ITC** (Information Transfer Capability)

  This field is displayed whenever *Communication Type* contains "56k-data," 64k-data," or "wideband." The field is used to determine the type of transmission facilities to be used for ISDN calls originated from the corresponding endpoint. Valid entries are **restricted** (default for "56k-data" or "64k-data") and **unrestricted** (default for "wideband").

The value **unrestricted** indicates that only unrestricted facilities are to be used to complete the call. An unrestricted facility is a transmission facility that does not enforce “1’s” density digital transmission (that is digital information is sent exactly as is). Always enter **unrestricted** for Wideband Access Endpoints.

The value **restricted** indicates that either restricted or unrestricted transmission facilities will be used to complete the call. A restricted facility is a transmission facility that enforces “1’s” density digital transmission (that is, a sequence of eight digital “0’s” will be converted to a sequence of seven “0’s” and a digital “1”).

Whenever an access endpoint is added with the ITC administered as “unrestricted,” the endpoint’s associated port must be a channel of a DS1 circuit pack with the *Zero Code Suppression* field administered as “B8Zs.” Otherwise, form submission is rejected with the appropriate error message. On the other hand, whenever an access endpoint is added with the ITC administered as “restricted,” the endpoint’s associated port can be a channel from a DS1 circuit pack with the *Zero Code Suppression* field administered as either “ZCS” or “B8ZS.”

For an existing access endpoint, the field can be changed from “restricted” to “unrestricted” only if its associated port is a channel of a DS1 circuit pack with its *Zero Code Suppression* field administered as “B8ZS.” Otherwise, form submission is rejected with the appropriate error message.

3. Submit the Access Endpoint form.
4. Access the Conference Record form by entering from the console **add conference *xxx***, where ***xxx*** is the conference number.
5. Enter **DA** (Direct Access) into the *Type* field on page 2 of the form.
6. Enter the appropriate MCU-extension into the *MCU-ext* field.
7. For a Wideband endpoint, enter the appropriate endpoint extension into the *Dial-out #1* field. For a 2B endpoint, enter an appropriate 2B extension into the *Dial-out #1* field and another one into the *Dial-out #2* field. An example of the form completed to administer 2B Dedicated Access endpoints appears as follows:

```
add conference 123                                          page  2 of 8
                              CONFERENCE RECORD
CONFEREES
1          Type: DA__ MCU-Ext: 5000_ Meet-Me Number: xxxxxxxxxxxxxx
         Redial? _   App Comp? _          Basic/Enh: Basic
   Dial-Out # 1: 5020_________________ Dial-Out # 2: 5021_____________
  Password Reqd? _

2          Type: P64_ MCU-Ext: _____ Meet-Me Number: xxxxxxxxxxxxxx
         Redial? _   App Comp? _          Basic/Enh: Basic
   Dial-Out # 1: ____________________ Dial-Out # 2: ____________________
  Password Reqd? _

3          Type: P64_ MCU-Ext: _____ Meet-Me Number: xxxxxxxxxxxxxx
         Redial? _   App Comp? _          Basic/Enh: Basic
   Dial-Out # 1: ____________________ Dial-Out # 2: ____________________
  Password Reqd? _

4          Type: P64_ MCU-Ext: _____ Meet-Me Number: xxxxxxxxxxxxxx
         Redial? _   App Comp? _          Basic/Enh: Basic
   Dial-Out # 1: ____________________ Dial-Out # 2: ____________________
  Password Reqd? _
```

Figure 2-26. Conference Record Form for 2B Dedicated Access Endpoints.

8. Submit the Conference Record form.

Administering Direct Connect Endpoints

The BRI/DCP Direct Connect feature allows endpoints to connect directly to the MCU via ISDN-BRI and/or Digital Communications Protocol (DCP). A maximum of 12 Direct Connect endpoints can be supported simultaneously on the MCU. This total can include up to 12 BRI Direct Connect endpoints but no more than four DCP Direct Connect endpoints. Direct Connect endpoints can be administered by populating and submitting a couple of forms, as follows:

1. Access the System-Parameters Customer-Option form by entering **change system-parameters customer-options**. For BRI connectivity, enter the maximum number of BRI ports that can be allocated in the system at the same time into the *Max. BRI Direct Connect Port Capacity* field (for this release, enter **0** or **12**). For DCP connectivity, enter the maximum number of DCP ports that can be allocated in the system at the same time into the *Max. DCP Direct Connect Port Capacity* field (for this release, enter **0** or **4**). Then submit the form.

2. Access the Processor Interface Data Module form by entering **change data-module *xxxx***, where ***xxxx*** is the extension number. To provide ISDN-BRI connectivity via an ISDN-BRI data module, enter **7500.** For DCP connectivity, enter the Processor Data Module (PDM) type**.** Then submit 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.

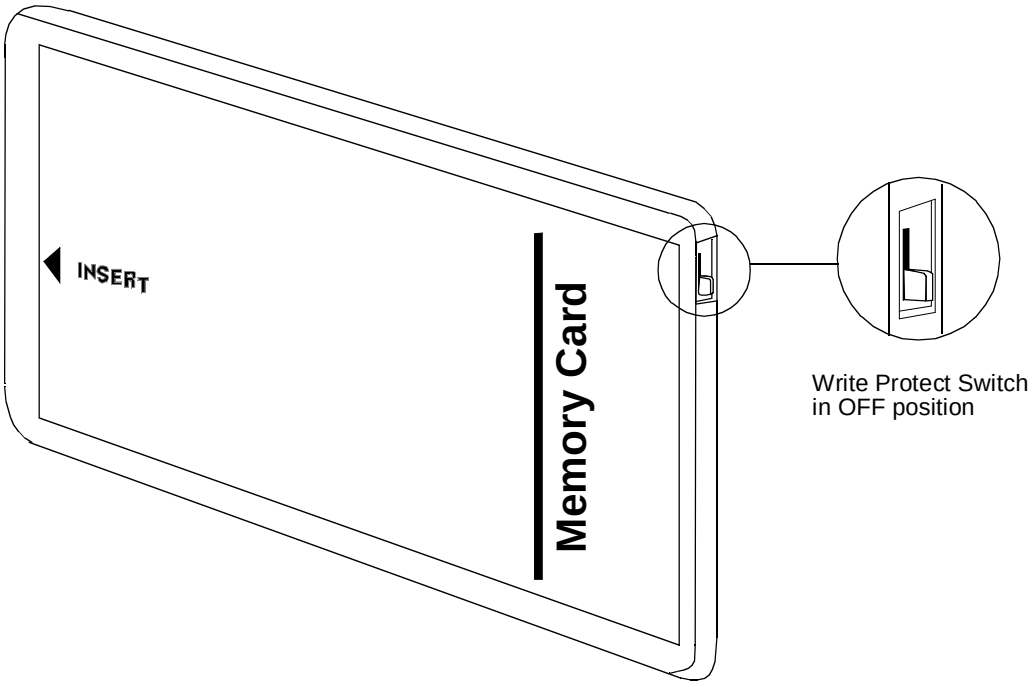

Figure 2-27. Memory Card

To save translations on the original system memory card and to make a backup memory card, follow these procedures:

1. Be sure the system memory card is inserted in the Network Control circuit pack with the write-protect switch in the OFF position.
2. Log in as **cust** on the MCU-MT.
3. At the *command:* prompt, enter **save translation** and press RETURN.
4. Check for a **0** in the *Error Code* column; a **0** indicates the save translation was successfully completed.

   NOTE:
   If a number other than **0** appears, the save translation process did not complete. Record the number in the *Error Code* field along with any *Error Message,* then call either the AT&T Video Technical Center (VTC) at **800 242-2121** or the Global Technical Access Center (GTAC) at **303 538-4666**. Select **Video** from the voice menu and a VTC or GTAC 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 and then swap the memory cards. Place the backup memory card in a safe place.

NOTE:
Translations can also be saved during scheduled maintenance by using the System Parameters Maintenance form.

Checking Status of the MMI Circuit Packs

The MMI Boards form provides information on all the MMI circuit packs within the MCU system.

The following table provides the command used to access the form.

Table 2-19. Processor Channels Administration Commands

Action	Object	Qualifier
list	mmi	-----

To access the form, follow this procedure:

1. At the *command:* prompt, enter **list mmi**.
2. Verify that the screen displays the *MMI Boards* form (see "MMI Boards Form" on page 2-81 for a list of the fields and their values).

MMI Boards Form

The following list identifies the fields on the MMI Boards form and the possible values.

- *MMI Address*

  Indicates the address of the MMI port circuit pack.

- *Code*

  Indicates the "TN" code for the MMI port circuit pack (for example, TN787D).

- *Cabled DS1*

  Indicates the address of the DS1 circuit pack that is cabled to the MMI circuit pack.

- *DS1 Interface*

  Indicates whether the MMI circuit pack is cabled to the network-DS1 circuit pack (**network**) or to the ESM-DS1 circuit pack (**ESM**).

- *Synch Status*

  Indicates the synchronization status of the MMI circuit pack. Values include **providing**, **listening**, and **out-of-synch**.

- *Ports In Service*

  Indicates the total number of MMI resources in service per circuit pack. Number ranges from **0** to **32** for the MSM and **0** to **24** for the ESM.

  NOTE:
  This field applies to MMI ports that are not in a maintenance-busy state and are therefore available for use.

- *Ports In Use*

  Indicates the total number of MMI resources being used per circuit pack. Number ranges from **0** to **32** for the MSM and **0** to **24** for the ESM.

  NOTE:
  This field applies to MMI ports that are either reserved or being used. Even if the "Appl Compliant" flag is set to "no," the ports are reserved. This is true even in the case where the ports are not in use due to the flag being changed from "no" to "yes" while the conference is active.

- *Cable Status*

  Indicates the status of the cable linking the DS1 and MMI circuit packs. Values include **in-service**, **out-of-service**, and **maint-busy**.

Checking MCU Performance

3

Overview

The MCU provides a variety of reports upon request. These reports provide details about the performance and status of trunks and MCU ports.

The optional system printer, not the local printers connected to the MCU-MT or MCU-ST, automatically prints all reports when the command includes the schedule qualifier. The system printer is remotely connected to the MCU via a 7400B data module.

If you do not have a system printer, individual reports can be printed on local printers by adding the print qualifier to the command line.

Scheduling Reports

The schedule option appears as a qualifier for many MCU list, display and test commands. Whenever a command containing the schedule qualifier is executed, a J*ob Id* is generated and a Report Scheduler screen appears so you can change the *Print Interval*. A maximum of 50 *Job Ids* can be scheduled for printing on the system printer.

The *Print Interval* field specifies when each report is printed (**immediate**, **deferred**, or **scheduled**). The **immediate** option prints the report on-demand and is the default.

The **deferred** option schedules the report for a single printing at some day or time in the future. Once the report is printed, the *Job Id* is automatically deleted.

Both the **deferred** and **scheduled** options provide two additional fields for *Print TIme* and *Days of Week* to specify when the report is to be printed. The

scheduled option differs from the **deferred** option because the specified report continues to print at the scheduled time until it is removed from the Report Scheduler.

Table 3-1 lists the commands that relate to the Report Scheduler feature.

Table 3-1. Report Scheduler Administration Commands

Action	Object	Qualifier
list	report-scheduler	
change	report-scheduler xx*	
remove	report-scheduler xx	

* xx = the Job ID

When a scheduled report needs to be changed with the **change report-scheduler** command, the *Job Status* field indicates whether the report is scheduled to be printed (**print-next**), currently printing (**printing**), finished printing (**printed**), or not scheduled to print (**waiting**) during the current 15-minute interval. To print a scheduled report immediately, change the *Print Interval* field to **immediate**.

Recent Change History

The Recent Change History Report allows a history of the most recent administration and maintenance changes to be viewed or printed. This report may be used for diagnostic or informational purposes.

The MCU maintains a log in the software buffer of up to 250 of the most recent administration and maintenance commands. This log is called the transaction log. The commands must be data-affecting and are called data commands (see Table 3-2).

The Recent Change History Report shows the command, date and time, MCU port (such as, MCU-MT, INADS, or CDR) and the login used to enter the command. The transaction log can be displayed on screen or printed using the following command:

Table 3-2. Recent Change History Administration Commands

Action	Object	Qualifier
list	history	print

The following list shows the data commands available on the MCU:

Table 3-3. MCU Data Commands

Command	Command	Command
add	enable	reset
busyout	get	rp
change	list	save
clear	mark	set
disable	monitor	status
display	recycle	test
download	release	upload
duplicate	remove	wp

Trunk Group Measurements Report

Three reports provide traffic measurements for trunk groups and Call-By-Call (CBC) reports. The Trunk Group CBC Report displays the last-hour traffic data for any specified trunk group. The Trunk Group Summary Report tracks the activity of the trunk group. The Trunk Group Measurements Hourly Report captures hourly trunk group usage.

Table 3-4 shows the commands that relate to obtaining trunk group measurements.

Table 3-4. Trunk Group Measurements Administration Commands

Action	Object	Qualifier
change	meas-selection trunk-group	print or schedule
display	meas-selection trunk-group	print or schedule
list	measurements trunk-group hourly ##	print or schedule
list	measurements trunk-group summary	yesterday-peak, today-peak, last-hour, print, or schedule
list	measurements cbc-trunk-group <number>	last-hour, print, or schedule

Trunk Group Measurements Call-By-Call Report

The Trunk Group Measurements CBC Report monitors the trunk group and determines if the usage selected by the Usage Allocation Plan (UAP) is optimal.

Each UAP specifies the network services and features assigned to the trunk group. By implementing the UAP, the CBC trunk group can be optimized within certain limits without involving any of the Inter-Exchange 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 this report can be analyzed, the intent of the UAP must be known.

The report appears as follows:

```
list measurements cbc-trunk-group 99 last-hour
Switch Name: Cust_Switch_Name
Switch Name:                                  Date: 2:15 pm WED DEC 13, 1995

                      CBC TRUNK GROUP MEASUREMENTS
                  Peak Hour For CBC Trunk Group 99 : 1300
Queue Size          : 40                            Usage Allocation Plan Used
Calls Queued        : 23                   Plan Number:  1  0   0   0   0   0
Queue Overflow      : 0                       Duration: 60  0   0   0   0   0
Queue Abandonments  : 4
Out of Service      : 0
                  Min  Max Meas Total Total Inc.  Ovf  Ovf  Ovf  %   %   %Out
Service/Feature   Chn  Chn Hour Usage Seize Seize  TG  S/F  Max TBM ATB  BLK
  outwats-bnd       5   12 1300   240   333   0     0   0   23   10   2    1
  sdn               4    8 1300    40    30  22     0   0    0   62   1    0
  other             0   20 1300    70    41  36     0   0    0    0   0    0
```

Figure 3-1. Trunk Group Measurements Call-By-Call Report

The following list identifies the data collected by the Trunk Group Measurements CBC Report:

- *Queue Size*

  This field is set to **0**; no queue is administered.

- *Calls Queued*

  This field is not used.

- *Queue Overflow*

  This field is not used.

- *Queue Abandonments*

  This field is not used.

- *Out of Service*

  The number of trunks in the trunk group that are out-of-service at the time the measurements are taken. An individual trunk may be taken out-of-service automatically by the MCU if too many errors occur or by maintenance personnel to run tests.

- *Usage Allocation Plan Used*

  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 Trunk Group Summary Report appears as follows:

```
list measurements trunk-group summary last-hour

Switch Name: Cust_Switch_Name                    Date: 1:58 pm MON DEC 11, 1995

                          TRUNK GROUP SUMMARY REPORT

                       Peak Hour For All Trunk Groups: 1000

Grp Grp  Grp  Grp Meas Total Total Inc.  Grp  Que  Calls Que Que Out   %  %Out
No. Size Type Dir Hour Usage Seize Seize Ovfl Size Qued  Ovf Abd Serv ATB Blk
20  1    did  inc 1200 0     0     0     0    0    0     0   0   0    0   *
30  1    tie  two 1200 36    0     0     0    0    0     0   0   0    100 0
37  22   tand two 1200 598   179   0     0    0    0     0   0   0    0   0
38  23   isdn two 1200 171   654   0     0    0    0     0   0   0    0   0
39  22   isdn two 1200 270   762   762   0    0    0     0   0   0    0   0
40  5    co   two 1200 61    32    0     6    0    0     0   0   0    6   15
```

Figure 3-2. Trunk Group Summary Report

The following list identifies the data collected by the Trunk Group Summary Report:

- *Peak Hour for All Trunk Groups*

  Peak hour and busy hour are one in the same. The peak hour is the highest usage for each day regardless of the average across days.

- *Grp No.*

  A number that identifies each trunk group associated with the displayed data.

- *Grp Size*

  The number of administered trunks in a trunk group.

- *Grp Type*

  The type of trunk (**isdn-pri** or **tie**) in the trunk group.

- *Grp Dir*

  The direction (**incoming**) of the trunk group.

- *Meas Hour*

  The hour (using the 24-hour clock) in which the measurement was taken.

- *Total Usage*

  The total time the trunks are busy during the one-hour measurement period.

- *Total Seize*

  The number of times a trunk in the trunk group was seized. The call may or may not have been completed.

- *Inc. Seize*

  The total number of times a trunk in the trunk group was seized to receive an incoming call. The call may or may not have been completed.

- *Grp Ovfl*

  This field is not used.

- *Que Size*

  This field is not used.

- *Calls Qued*

  This field is not used.

- *Que Ovf*

  This field is not used.

- *Que Abd*

  This field is not used.

- *Out Serv*

  The total number of trunks in the trunk group that are out-of-service (listed as **maintenance busy**) at the time data was collected. An individual trunk may be taken out-of-service by the MCU when an excessive number of errors occurs or by maintenance personnel to run diagnostic tests. If the trunks were taken out-of-service by the MCU, it may be appropriate to notify maintenance personnel.

- *% ATB*

  The percentage of time that all trunks in the trunk group were simultaneously in use during the measurement interval. In use means all trunks were busy either processing calls or they were busied-out by maintenance. A high number in this field probably indicates calls are being blocked.

- *% Out Blk*

  This field is not used.

Trunk Group Hourly Report

NOTE:
To generate this report, the number of trunks you want measured hourly must be entered in the *Trunk Group Number* field of the Trunk Group Measurement Selection. If no selections are made, no measurements are taken.

The following list identifies the data collected by the Trunk Group Hourly Report:

- *Grp No.*

  A number that identifies each trunk group associated with the displayed data.

- *Grp Size*

  The number of administered trunks in a trunk group.

- *Grp Type*

  The type of trunk (**isdn-pri** or **tie**) in the trunk group.

- *Grp Dir*

  The direction (**incoming**) of the trunk group.

- *Que Size*

  This field is not used.

- *Total Usage*

  The total time the trunks are unavailable due to the trunks being busy, busied-out by maintenance, or any other reason.

- *Maint Usage*

  The total usage of trunks in this trunk group for maintenance busy or any other non-call situation where trunks are unavailable.

- *Total Seize*

  The total number of seizures on the trunk group.

- *Inc. Seize*

  The total number of times a trunk in the trunk group was seized to receive an incoming call. The call may or may not have been completed.

- *Tandem Seize*

  This field is not used.

- *Grp Ovfl*

  This field is not used.

- *Calls Qued*

  This field is not used.

- *Que Ovf*

  This field is not used.

- *Que Abd*

  This field is not used.

- *Out Serv*

  The total number of trunks in the trunk group that are out-of-service (listed as **maintenance busy**) at the time data was collected. An individual trunk may be taken out-of-service by the MCU when an excessive number of errors occurs or by maintenance personnel to run diagnostic tests. If the trunks were taken out-of-service by the MCU, it may be appropriate to notify maintenance personnel.

- *% ATB*

  The percentage of time that all trunks in the trunk group were simultaneously in use during the measurement interval. In use means all trunks were busy either processing calls or they were busied-out by maintenance. A high number in this field probably indicates calls are being blocked.

- *% Out Blk*

  This field is not used.

Trunk Outage Report

This report lists up to a maximum of five trunks in each trunk group that were out-of-service when sampled. The number of times the trunks were out-of-service when sampled is also given. The trunk outage data is kept for the current day, the previous day and the last hour.

Table 3-5 shows the command that relates to obtaining trunk outage measurements.

Table 3-5. Trunk Outage Measurements Administration Commands

Action	Object	Qualifier
list	measurements outage-trunk	yesterday, today, print, or schedule

Trunk Outage Measurements Report

The following list shows the data collected for the Trunk Outage Measurements Report:

- *Grp No.*

  A number that identifies each trunk group associated with the displayed data.

- *Grp Type*

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

- *Grp Dir*

  Identifies the direction (**inc** for incoming only) of the trunk group.

- *Grp Size*

  The number of trunks in the trunk group.

- *Grp Mbr #*

  The number that identifies a specific trunk member in the group that is out-of-service.

- *#Sampled Outages*

  The number of times the group member is sampled as out-of-service over the period covered by the report (**yesterday**, **today**, or **last hour**). If there are no outages then no data is displayed. The sampling period is once per hour. If the report covers several hours but the column indicates a small number of outages, the trunk member may be providing intermittent service. Use the facility test calls feature to determine whether a specific trunk member is functioning. If you suspect the trunk is causing problems,

use the ACA feature to monitor the particular trunk group. If the trunk is out-of-service, you may choose to refer the problem to maintenance personnel.

Trunk Group Traffic Report

The Trunk Group Status Report gives a current indication of the load on various trunk groups. For comparative analysis, the trunk members in the group that are active on calls are also displayed. Table 3-6 shows the command that relates to obtaining trunk status reports:

Table 3-6. Trunk Status Administration Commands

Action	Object	Qualifier
monitor	traffic trunk-groups	

Trunk Status Report

The following list shows the data collected for the Trunk Status Report:

- *#*

  Group number that identifies each trunk group.

- *S*

  Group size that shows the number of trunks administered for the trunk group.

- *A*

  Shows the number of trunk members in the group that are active on a call. Busied-out trunks are not active.

- *Q*

  This field is not used.

- *W*

  This field is not used.

Automatic Circuit Assurance Reports

The Automatic Circuit Assurance (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 2-20 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** 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 are 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 ACA Measurements Report appears as follows:

```
list measurements aca

                                                Today: 9:09 am WED DEC 20, 1995
Automatic Circuit Assurance Measurements

Day & Time          Trunk            Trunk          Trunk         Type of
of Referral       Group No.      Access Code       Member        Referral
```

Figure 3-3. Automatic Circuit Assurance 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*

  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 (UDS1) Measurements

There are two performance measurements for DS1 (UDS1) links. The DS1 (UDS1) Link Performance Measurements Summary Report provides an indication of the quality of a DS1 (UDS1) link that connects to a DS1 (UDS1) interface circuit pack. The DS1 (UDS1) Link Performance Detailed Log Report lists errored events for the past 24 hours. The errored event records are listed for each 15-minute interval. Included are 96 records (for each 15-minute interval) dating from the current time back to 24 hours before the current interval, beginning with the oldest record.

Check these reports if you receive complaints from video endpoints about the stability of the far-end handshake, especially when the Status of Conference screen reports **Handshake** as the drop reason (see the *AT&T MultiPoint Control Unit Reservation Agent Manual* for troubleshooting details).

Table 3-8. DS1 (UDS1) Measurements Administration Commands

Action	Object	Qualifier
clear	measurements-ds1 (uds 1)	
list	measurements ds1 (uds 1)	location, print, or schedule
list	measurements ds (uds)1-log	location, print, or schedule

DS1 (UDS1) Link Performance Measurements Summary

The following list shows the data collected on the DS1 (UDS1) Link Performance Measurements Summary form:

- *Counted Since*

  The date and time the counters were last cleared and restarted. The counters are set to 0 and start accumulating data when the MCU is administered or reinitialized. The current system time appears in the field once the system clock is set. An error message appears if the system clock has not been set following a reinitialization.

- *Number of Seconds Elapsed*

  The total number of 15-minute intervals (**0** to **96**) in the past 24 hours that have valid values. An invalid value is any 15-minute interval when the time was changed, a system reinitialization occurred or the DS1 (UDS1) circuit pack was removed.

- *Worst 15-Min Interval*

  The date, ending time, and count for the 15-minute period that contained the maximum count in each error category. If there are no errors, the field displays a **0** with the oldest time.

- *Total of 24-Hour Count*

  The total count in each error category for the last 24 hours (**0** to **65535**).

- *Current 15-Min Interval Count*

  The count-so-far in each error category for the15-minute interval in progress when the report was requested. If no errors have occurred in any of the categories during the current 15-minute interval, the respective field will contain a **0**. If the system is busy performing call processing functions and cannot respond within eight seconds, the field will show **N/A**.

DS1 (UDS1) Link Performance Detailed Log Report

The following list shows the data collected on the DS1 (UDS1) Link Performance Detailed Log Report form:

- *Date*

  The date on which the event occurred (**yesterday** or **today**).

- *Time*

  The ending times for the 15-minute sampling intervals.

- *Errored Second*

  The number of errored seconds (maximum of **900**) for the specified interval. An errored second is any second in which one or more data transmission errors occurred. **N/A** indicates the count for that interval was not available because there was no data.

- *Bursty Err Sec*

  The number of bursty errored seconds (maximum of **900**) for the specified interval. A bursty errored second is any second in which two to 319 data transmission errors occurred. **N/A** indicates the count for that interval was not available. An error count of this severity causes a minor alarm. Depending on local arrangements, it may be appropriate to notify maintenance personnel if they have not already been alerted.

- *Severely Err Sec*

  The number of severely errored seconds for the specified interval (maximum of **900**). A severely errored second is any second in which 320 or more data transmission errors occurred. An error count of this severity results in a major alarm. Depending on local arrangements, it may be appropriate to notify maintenance personnel if they have not already been alerted.

- *Failed Second*

  The number of failed seconds states for the specified interval (maximum of 900). A failed second state exists any time that ten or more consecutive severely errored seconds have occurred. An error count of this severity results in a major alarm. **N/A** indicates the count for this interval was not available. Depending on local arrangements, it may be appropriate to notify maintenance personnel if they have not already been alerted.

- *Valid Interval*

  Specifies whether or not a valid count is provided by the DS1 (UDS1) interface circuit pack. A value of **y** indicates all counts are valid for the interval. An invalid interval is any 15-minute interval during which the system clock was changed, a system reinitialization occurred, or the specified circuit pack was removed.

Hunt Group Measurements

There are three measurement reports for hunt groups. The Hunt Group Measurements Report monitors the hunt group. This report shows hunt group measurements for yesterday's peak hour, today's peak hour, and the last hour. A peak hour is the hour within a 24-hour period when there was the greatest usage for the specified day.

The Hunt Group Performance Report gives the slowest hourly average speed of answer for each hunt group. The Hunt Group Status Report provides an

instantaneous indication of the load that is pending for various hunt groups. Table 3-9 lists the commands related to hunt group measurements.

Table 3-9. 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 Hunt Group Measurements Report appears as follows:

```
list measurements hunt-group today-peak

                                          Date: 9:14 am WED DEC 20, 1995
                          HUNT GROUP MEASUREMENTS

Grp Grp             Grp  Grp  Meas Total  Calls Calls Que  Calls Time  Speed
No. Name            Size Type Hour Usage  Ans.  Aban. Size Que.  Avail Ans(sec)

1   Netcon Channels 4    ucd  800  0      0     0     0    0     144   0
```

Figure 3-4. Hunt Group Measurements Report

The following list shows the data collected on the Hunt Group Measurements Report form:

- *Grp No.*

  A number that identifies each hunt group.

- *Grp. Name*

  Name assigned, during administration, to the hunt group.

- *Grp Size*

  The number of extensions assigned to the hunt group.

- *Grp Type*

  The type of hunt group.

- *Meas Hour*

  The peak hours for the specified hunt group. Depending on the time of day the report is generated, today's peak hour data may vary.

- *Total Usage*

  The sum of all times that the members of a hunt group are busy on hunt group calls.

- *Calls Ans*

  The total number of all hunt group calls answered by the hunt group.

- *Calls Aban*

  The total number of calls which attempt to reach the hunt group but abandon the attempt before being answered.

- *Queue Size*

  This field is not used.

- *Calls Queued*

  This field does not apply.

- *Time Avail*

  The total time that the hunt group extensions are not in use but are available to receive hunt group calls during the measurement hour.

- *Speed of Answer*

  The average time interval (in seconds) from when the call first enters the hunt group until the call is answered by a hunt group member.

Hunt Group Performance Report

The following list shows the data collected on the Hunt Group Performance Report:

- *Grp No.*

  A number that identifies each hunt group.

- *Grp Size*

  The number of extensions assigned to the hunt group.

- *Grp Type*

  The type of hunt group.

- *Slowest Speed of Answer (sec)*

  A bar graph representation of the slowest hourly average speed of answer for the report interval (**yesterday** or **today**).

- *Speed Ans (sec)*

  The number of seconds corresponding to the slowest hourly average speed of answer (longest amount of time to answer) for the report interval (**yesterday** or **today**).

- *Meas Hour*

  The starting time (using the 24-hour clock) of the hour during which the data was recorded.

- *Daily Avg.*

  The number corresponding to the 24-hour daily average speed of answer for each hunt group.

Hunt Group Status Report

The Hunt Group Status report appears as follows:

```
monitor traffic hunt-groups

                      HUNT GROUP STATUS         9:18 am WED DEC 20, 1995

    #    S    A    Q    W    LCIQ        #    S    A    Q    W    LCIQ
    1    4    0    0    0    0

(#: Group; S: Grp Size; A: Active Members; Q: Q Length; W: Calls Waiting)
(LCIQ: Longest Call in Queue in seconds                                 )
```

Figure 3-5. 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 Security Violations Status report appears as follows:

```
monitor security-violations

                        SECURITY VIOLATIONS STATUS

                                              Date: 9:21 am WED DEC 20, 1995

     SYSTEM MANAGEMENT VIOLATIONS               REMOTE ACCESS VIOLATIONS

 Date  Time  Login  Port        Ext        Date  Time    TG No. Mbr   Ext
 05/04 15:22 testid SYS-PORT    71111      05/03 21:14     3     50   81111
 05/04 15:23 testit SYS-PORT    73456      05/04 22:10    21     43   83333
 05/04 15:25 testx  SYS-PORT    72198      05/04 22:13    21     43   83333
```

Figure 3-6. 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 Monitor System View1 report appears as follows:

```
monitor system view1

                                                MAINTENANCE STATUS

                                              # of alarms for trunks: 0
                                              # of alarms for stations: 0
                                              # of alarms for other res: 2
                                              INADS has been informed? n


                              TRAFFIC STATUS
                            Measurement Hour: 8
         Trunk Group Measurement           Hunt groups Measurement
  (4 grps with highest %time ATB)     (4 grps with highest # of qued calls)
      Grp no:                                 Grp no:
     Grp dir:                               Calls qued:
  Calls qued:                               Calls aban:
   %Out blkg:
   %Time ATB:
                                                    9:26 WED DEC 20 1995
                - press CANCEL to quit -
```

Figure 3-7. 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 3-28 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 Performance Summary Report appears as follows:

```
list performance summary yesterday                                Page 1

Switch Name:  Customer_Switch_Name
SUMMARY PERFORMANCE REPORT                    Today: 4:38 pm  SAT DEC 16, 1995
PEAK HOUR TRUNK BLOCKING                          DAILY ARS CALLS CARRIED
Grp - %Out Blocking or % ATB - Grp       Grp -  % Calls Per Group Type -   %
No. 1 2 3 4 5 6 7 8 9 10 20 50 Blk
54  ////////////////////////   42        Type 1 10 20 30 40 50 60 80 100 Calls
59  ////////////////////////    39
58  ////////////////////////    36
63  ////////////////////////    34
61  ////////////////////        10
      TRUNKS OUT OF SERVICE                    TRUNKS NOT SERVICE
Grp Trunks Out Of Service All Day        Grp Trunks With No Calls All Day
No. ----------------------------         No. ----------------------------
41    9 19                               55    1
73    7                                  60    9
211   1 2 3 4 5 6 7 8 9 10               223   19 20 21 22 23
      more out of service                      more out of service
```

Figure 3-8. Performance Summary Report

The following list shows the data collected for the Performance Summary Report:

- *Peak-Hour Trunk Blocking*

  Lists up to the maximum of five trunk groups that have the highest percentage of blocking in a measurement hour. For incoming trunk groups, the percentage of blocking is shown in the *% ATB* field. The number in this field represents the percentage of time all trunks in the trunk group were simultaneously in use during the measurement interval.

- *Daily ARS Calls Carried*

  This field is not used.

- *Trunks Out of Service*

  Lists four trunk groups with out-of-service trunks over the report interval. A list of the first 10 trunks that are out-of-service is also given. The message `more out of service` is displayed when more than four trunk groups are out of service or more than ten members of any trunk group are out of service. This measurement is a summary of the Outage Trunk Measurement Report.

- *Trunks Not Used*

  Lists the four trunk groups with trunks that have not been used over the report interval. A list of the first five trunks, in each of the identified groups, that have not been used are also listed. The message `more out of service` displays when more than four trunk group have not been used or whenever more than five members of any group have not been used. You should determine the exact reason why trunks are not being used. If there is only one trunk member identified for a trunk group and that member has zero calls, the trunk may be defective.

Trunk Group Performance Report

The Trunk Group Performance Report appears as follows:

```
list performance trunk-group yesterday
Switch Name:  Cust_Switch_Name
Trunk Group Performance                        Today: 4:28 pm  SAT DEC 16, 1995
             HIGHEST HOURLY TRUNK GROUP BLOCKING
Grp Grp      Grp Grp   --% Outgoing Blocking or % ATB-- %Out  %Time Meas  Total
No. Type     Dir Size  1 2 3 4 5 6 7 8 9 10 20 30 40 50 Blkg  ATB   Hour  Calls
1   isdn-pri in  6     ////////////////                       9     1200  876
2   isdn-pri in  5     ////////////////////////               30    1400  94
3   tie      in  14    ////////////                           36    1300  312
5   tie      in  10    ////////////////////                   99    1300  542
12  tie      in  18    ////////////////                       96    1400  614
23  tie      in  7     ////////////////////                   81    1400  359
41  tie      in  8     ////////////////////                   91    1300  411
221 tie      in  5     ////////////////////                   77    1300  109
```

Figure 3-9. Trunk Group Performance Report

The following list shows the data collected for the Trunk Group Performance Report:

- *Group No.*

  A number that identifies the trunk group associated with the displayed data.

- *Grp Type*

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

- *Grp Dir*

  The direction (**incoming**) of the trunk group.

- *Grp Size*

  The number of trunks in the trunk group.

- *% Outgoing Blocking or % ATB*

  A graphical representation equivalent to the numerical value of all trunks busy. For incoming trunks, peak hour is the largest *% ATB*. Since outgoing blocking is meaningless for incoming trunks, it is displayed as ***** in that column.

- *%Out Blkg*

  This field is not used.

- *%Time ATB*

The percentage of time that all trunks in the trunk group were simultaneously in use during the measurement interval. In use means the trunks were busy processing calls or are busied-out by maintenance.

- *Meas Hour*

  The starting time (using the 24-hour clock) of the hour during which the data was recorded.

- *Total Calls*

  Because the value collected is not relevant for incoming trunks, it is displayed as ***** in that column.

Occupancy Reports

These four reports track the percentage of time the MCU is busy performing call processing tasks, maintenance tasks, administration tasks, and the operating system support for each of these tasks. The percentage of time when the MCU is not used is referred to as Idle Occupancy (*Idle Occ*). These reports determine usage, available capacity and help to diagnose certain customer-reported problems.

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

Table 3-13. 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 Occupancy Summary Report appears as follows:

```
Switch Name:  Customer_Switch_name                          Page 1

                                                Date: 3:12 pm MON DEC 11, 1995
                        OCCUPANCY SUMMARY MEASUREMENTS
                         Peak Hour For Occupancy: 1000
Meas    Stat   CP    Sm    Idle   Total   Total   Intcom   Inc     Out     Pnet
Hour    Occ    Occ   Occ   Occ    Calls   Atmpt   Atmpt    Atmpt   Atmpt   Atmpt
1400    4      13    22     61    28463   43908   3970       0       0     39938
1300    5      14    23     58    35626   51094   5784       0       0     45310
1200    5      14    23     58    35632   51120   5783       0       0     45337
1100    5      14    23     58    35645   51130   5788       0       0     45342
1000    5      16    24     55    24342   31756   3590       0       0     28166

            press CANCEL to quit  --  Press NEXT PAGE to continue
```

Figure 3-10. 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.

The Occupancy Last-Hour Measurements Report appears as follows:

```
Switch Name:  Customer_Switch_name                          Page 1

                                          Date: 3:13 pm MON DEC 11, 1995
                        OCCUPANCY LAST-HOUR MEASUREMENTS

Meas    Stat   CP    Sm    Idle   Total   Total   Intcom   Inc     Out     Pnet
Min     Occ    Occ   Occ   Occ    Calls   Atmpt   Atmpt    Atmpt   Atmpt   Atmpt
15:11    2      5     26    67      646    1421       0       0       0      1421
15:08    4      5     22    69      641    1412       0       0       0      1412
15:05    5      6     25    64      639    1410       0       0       0      1410
15:02    4      6     24    66      645    1420       0       0       0      1420
14:59    3      4     22    71      639    1411       2       0       0      1409
14:56    3      5     24    68      639    1412       2       0       0      1410
14:53    6      6     24    64      645    1418       3       0       0      1415
14:50    4      5     27    64      641    1418       3       0       0      1415
14:47    4      6     19    71      648    1429       3       0       0      1426
14:44    2      5     26    67      639    1405       3       0       0      1402
14:41    4     14     19    63     1624    2399     243       0       0      2156
14:38    6     15     28    51     1786    2556     290       0       0      2266

            press CANCEL to quit  --  Press NEXT PAGE to continue
```

Figure 3-11. Occupancy Last-Hour Measurements Report

Occupancy Busiest-Interval Measurements Report

The data collected for the Occupancy Busiest-Interval Measurements Report is the same as the Occupancy Summary Report except that the measurements are a collection of the 20 busiest 3-minute intervals (instead of one-hour intervals) within the past two months. This report is useful to maintenance personnel who need to investigate habitual performance problems or problems that were not reported when they occurred. Be sure to check the hardware error log for an excessive amount of maintenance activity when investigating a problem.

An Occupancy Busiest-Interval Measurements Report for 3-minute interval measurements appears as follows:

```
Switch Name:  Customer_Switch_name                              Page 1
                                                Date: 3:13 pm MON DEC 11, 1995
              OCCUPANCY BUSIEST 3-MINUTE INTERVALS MEASUREMENTS

Date of      Stat  CP   Sm   Idle  Total  Total  Intcom   Inc   Out    Pnet
Occur        Occ   Occ  Occ  Occ   Calls  Atmpt  Atmpt   Atmpt Atmpt  Atmpt
11/11/10:20  16     9   26   49     686   1225    245     0     0     980
11/11/11:14   8    16   27   49    1788   2558    286     0     0    2272
11/11/12:38   7    15   21   57    1786   2554    286     0     0    2268
11/11/13:41   6    16   26   52    1786   2553    290     0     0    2263
11/11/14:11   7    15   25   53    1780   2557    285     0     0    2272
           press CANCEL to quit  --  Press NEXT PAGE to continue
```

Figure 3-12. Occupancy Busiest-Interval Mesaurements Report

Occupancy Communications Link Measurements Report

The Occupancy Communications Link Measurements Report appears as follows:

```
list measurements communications-links                          Page 1
                                                Date: 1:55 pm TUE DEC 12, 1995
                        COMMUNICATION LINK MEASUREMENTS

 Meas     Link     Link     Link     Link     Link     Link     Link     Link
 Hour     9        10       11       12       13       14       15       16
 1200     10471    576      24       4        0        40       2        0
 1100     13764    612      24       14       0        313      4        0
 1000     12217    550      24       4        0        36       9        0
 900      12365    601      26       4        0        32       2        0
 800      12630    559      28       4        0        36       4        0
 700      12714    412      24       4        0        36       4        0
 600      12531    299      24       4        0        40       4        0
 500      12407    352      24       4        0        42       2        0
 400      12173    311      34       4        0        32       2        0
 300      12121    301      24       4        0        36       4        0
 200      12561    412      24       4        0        36       4        0
 100      12501    478      24       4        0        36       2        0
           press CANCEL to quit  --  Press NEXT PAGE to continue
```

Figure 3-13. 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 Tone Receiver Measurements Report appears as follows:

```
list measurements tone-receiver last-hour

Switch Name:  Cust_Switch_Name                  Date: 4:16 pm TUE DEC 19, 1995

                          TONE RECEIVER MEASUREMENTS

                 Total   Total   Peak   Total    Peak     Total    Peak
      Hour  Type  Avail    Req    Req    Queued   Queued   Denied   Denied
      1400  DTMF    40      23     13      0        0        0        0
      1500  GPTD     6      94      7                        1        1

                      PN      PN         Peak      Total      Peak
       PN     Type     Req     Alloc      Alloc     Off-PN     Off-PN
       1      DTMF      8       8          200         0          5
       1      GPTD     12      12          3          0          0

             Press CANCEL to quit   -   press NEXT PAGE to continue
```

Figure 3-14. Tone Receiver Measurements Report

The following list shows the data collected for the Tone Receiver Measurements Report:

- *Hour*

  The starting time (using 24-hour clock) of the hour during which the greatest number of requests for tone receivers were made.

- *Type*

  The type of tone receiver being measured. Each TN748 circuit pack provides four DTMF ports (for touchtone reception) and two GPTD ports (for call progress tone reception).

- *Total Avail*

  The systemwide total number of DTMF and GPTD receivers that are available and not busied-out for maintenance.

- *Total Req.*

  The systemwide total number of requests by the tone receivers during the listed hour. The total number of requests is calculated by incrementing a counter for each request.

- *Peak Req*

  The systemwide peak number of simultaneous requests for tone receivers that occurred at any one time for the listed hour. The peak (or maximum) number is calculated by incrementing a counter for each request and decreasing the counter when the request fails or a tone receiver is released. Denied requests fail and are given the reorder tone.

- *Total Queued*

  The systemwide total number of tone receivers that were queued during the listed hour. There is no queuing for GPTD receivers.

- *Peak Queued*

  The systemwide maximum number of DTMF receivers that were queued at any one time during the listed hour. The system has a maximum queue size of four. GPTD receivers cannot be queued.

- *Total Denied*

  The systemwide total number of requests for tone receivers that were denied because none were available.

- *Peak Denied*

  The systemwide peak number of requests for tone receivers that were denied because no receivers were available during the listed hour.

- *PN*

  The port network where the circuit pack is located.

- *Type*

  The type of tone receiver being measured (**DTMF** or **GPTD**).

- *PN Req*

  The port number requests.

- *PN Alloc*

  The total number of tone receivers located in the port network that were allocated for use during the listed hour.

- *Peak Alloc*

  The peak number of tone receivers located in listed port network that were used simultaneously during the listed hour.

- *Total Off-PN*

  This field is not used.

- *Peak Off-PN*

  This field is not used.

Px64 Interoperability

A

Overview

This appendix contains the following:

- Interoperability results for the AT&T-MCU with different Px64 endpoint vendors
- Network connectivity results for the AT&T-MCU with different PBXs and Central Office switches

The Px64//H.320 standards have been implemented on the AT&T MultiPoint Control Unit (MCU) in conformance with the International Telecommunications Union-Telecommunications (ITU-T) standards on video teleconferencing. AT&T has tested and will continue the 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 certification is of Px64 (H.320) protocol interoperability; the products for different manufacturers will vary in video and audio quality.

The endpoints disclosed in the first two tables interoperate acceptably with the AT&T-MCU. Interoperability testing is done by using specific network configuration and terminal adapters. As such, correct operation may not occur by using other equipment. Nevertheless, the equipment indicated in this appendix is not necessarily endorsed or recommended by AT&T. Also, note that past interoperability with a particular endpoint does not ensure future interoperability. Only the versions identified in the first two tables were tested for interoperability with the AT&T-MCU.

The ITU-T standards are implemented on a “self-certification” basis. There is neither an official ITU-T conformance testing body nor 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 its ITU-T implementation with untested versions of ITU-T compliant endpoints.

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

On-Site Tested Endpoints

The table in this section shows the current results of AT&T’s Px64 interoperability testing for on-site tested endpoints. Tests are conducted on-site in a laboratory environment with purchased or loaned video endpoints, and the tests carry the distinction of AT&T certification of results.

Table A-1. Results for On-Site Tested Endpoints

Endpoint Version	Audio Modes		Network Connection								Video Coding Mode	Communication Modes		MCU Release(s) Tested	Terminal Adapter Release
	G.711	G.728	2x56k (112)	2x64k (128)	384k (H0)	6x56k (336)	6x64k (384)	768k	1472k	1536k (H11)		Meet-Me	Dial Out		
AT&T Vistium PVS 1300 Version 1.01	Y	Y	Y	Y	NA *	NA *	NA *	NA *	NA *	NA *	CIF & QCIF	Y	Y	1.1 & 2.0	Direct DEFINITY BRI
AT&T Vistium PVS 1300 Version 1.03.10	Y	Y	Y	Y	NA *	NA *	NA *	NA *	NA *	NA *	CIF & QCIF	Y	Y	1.1 & 2.0	Direct DEFINITY BRI
AT&T Vistium PVS 1200 Version 1.03.10	Y	Y	Y	Y	NA *	NA *	NA *	NA *	NA *	NA *	CIF & QCIF	Y	Y	2.0	Direct DEFINITY BRI
AT&T Vistium PVS 1300 Version 1.04.01	Y	Y	Y	Y	NA *	NA *	NA *	NA *	NA *	NA *	CIF & QCIF	Y	Y	2.0 & 3.0	Direct DEFINITY BRI
AT&T Vistium PVS 1200 Version 1.04.01	Y	Y	Y	Y	NA *	NA *	NA *	NA *	NA *	NA *	CIF & QCIF	Y	Y	2.0 & 3.0	Direct DEFINITY BRI
AT&T GVS 1000 Version 1.0	Y	Y	Y	Y	NA *	NA *	NA *	NA *	NA *	NA *	CIF & QCIF	Y	Y	1.1 & 2.0	Direct DEFINITY BRI
AT&T GVS 1000 Version 1.1	Y	Y	Y	NA *	NA *	NC	NC	NA *	NA *	NA *	CIF & QCIF	Y	Y	1.1 & 2.0	Direct DEFINITY BRI
AT&T GVS 1000 Version 1.1C	Y	Y	Y	Y	Y	NC	NC	NA *	NA *	NA *	CIF & QCIF	Y	Y	1.1	Ascend Multiband Plus Release 3.4F+
AT&T GVS 4000 Version 3.0P.03	Y	NA *	Y	Y	Y	NA †	NA †	NA *	NA *	NA *	CIF & QCIF	Y	Y	1.1 & 2.0	Ascend Multiband Plus Release 2.3
AT&T GVS 4000 Version 3.0P.03	Y	NA *	Y	Y	Y	Y	Y	NA *	NA *	NA *	CIF & QCIF	Y	Y	2.0	Ascend Multiband Plus Release 3.3A+ Release 3.4F+
AT&T GVS 4000 Version 4.1V	Y	NA *	Y	Y	Y	NA †	NA †	NA *	NA *	NA *	CIF & QCIF	Y	Y	1.1 & 2.0	Ascend Multiband Plus Release 2.3
AT&T GVS 4000 Version 4.1V	Y	NA *	Y	Y	Y	Y	Y	NA *	NA *	NA *	CIF & QCIF	Y	Y	2.0	Ascend Multiband Plus Release 3.3A+ Release 3.4F+
AT&T GVS 4000 Version 4.1V	Y	NA *	Y	Y	Y	Y	Y	NA *	NA *	NA *	CIF & QCIF	Y	Y	2.0	Promptus OASIS 3.51V6

Table A-1. Results for On-Site Tested Endpoints — *Continued*

Endpoint Version	Audio Modes		Network Connection								Video Coding Mode	Communication Modes		MCU Release(s) Tested	Terminal Adapter Release
	G.711	G.728	2x56k (112)	2x64k (128)	384k (H0)	6x56k (336)	6x64k (384)	768k	1472k	1536k (H11)		Meet-Me	Dial Out		
AT&T GVS 4000 Version 4.2V	Y	NA *	Y	Y	Y	Y	Y	NA *	NA *	NA *	CIF & QCIF	Y	Y	2.0	Ascend Multiband Plus Release 3.3A+ Release 3.4F+
AT&T GVS 4000 Version 4.2V	Y	NA *	Y	Y	Y	Y	Y	NA *	NA *	NA *	CIF & QCIF	Y	Y	2.0	Promptus OASIS 3.51V6
CLI Rembrandt II/VP Version 7.94	Y	NA *	Y	Y	Y	NA *	NA *	NC	NC	NC	CIF & QCIF	Y	Y	1.1 & 2.0	Ascend Multiband Plus Release 2.3+
CLI Rembrandt II/VP Version 7.94	Y	NA *	Y	Y	Y	Y	Y	Y	NA ‡	Y	CIF & QCIF	Y	Y	2.0	Ascend Multiband Plus Release 3.3A+^ Release 3.4F+^
CLI Rembrandt II/VP Version 7.94	Y	NA *	Y	Y	Y	Y	Y	Y	NA ‡	Y	CIF & QCIF	Y	Y	2.0	Ascend Multiband MAX Release 2.2N^
CLI Rembrandt II/VP Version 7.94	Y	NA *	Y	Y	Y	Y	Y	NA ‡	NA ‡	Y	CIF & QCIF	Y	NC	2.0	Teleos 11-1v1.1
CLI Rembrandt II/VP Version 7.94	Y	NA *	Y	Y	Y	Y	Y	NA ‡	NA ‡	Y	CIF & QCIF	Y	Y	2.0	Promptus OASIS 3.51V6
CLI Radiance Version 9.18	Y	NA *	Y	Y	Y	NA †	NA †	NC	NC	NC	CIF & QCIF	Y	Y	1.1	Ascend Multiband Plus Release 2.3
CLI Eclipse Version 1.1	Y	Y	Y	Y	NA *	NA *	NA *	NA *	NA *	NA *	QCIF only	Y	Y	1.1 & 2.0	Direct DEFINITY BRI
PictureTel S1000 Version 1.0	Y	Y	Y	Y	NA *	NA *	NA *	NA *	NA *	NA *	CIF & QCIF	Y	Y	1.1 & 2.0	Direct DEFINITY BRI
PictureTel S1000 Version 1.1	Y	Y	Y	NA *	NA *	NC	NC	NA *	NA *	NA *	CIF & QCIF	Y	Y	1.1 & 2.0	Direct DEFINITY BRI
PictureTel S1000 Version 1.1C	Y	Y	Y	Y	Y	NC	NC	NA *	NA *	NA *	CIF & QCIF	Y	Y	1.1	Ascend Multiband Plus Release 3.4F+

Table A-1. Results for On-Site Tested Endpoints — *Continued*

Endpoint Version	Audio Modes		Network Connection								Video Coding Mode	Communication Modes		MCU Release(s) Tested	Terminal Adapter Release
	G.711	G.728	2x56k (112)	2x64k (128)	384k (H0)	6x56k (336)	6x64k (384)	768k	1472k	1536k (H11)		Meet-Me	Dial Out		
PictureTel LIVE PCS-100 Version 1.0	Y	Y	Y	Y	NA *	NA *	NA *	NA *	NA *	NA *	CIF & QCIF	Y	Y	1.1	Direct DEFINITY BRI
PictureTel S4000 Version 3.0P.03	Y	NA *	Y	Y	Y	NA †	NA †	NA *	NA *	NA *	CIF & QCIF	Y	Y	1.1 & 2.0	Ascend Multiband Plus Release 2.3
PictureTel S4000 Version 3.0P.03	Y	NA *	Y	Y	Y	Y	Y	NA *	NA *	NA *	CIF & QCIF	Y	Y	2.0	Ascend Multiband Plus Release 3.3A+ Release 3.4F+
PictureTel S4000 Version 4.1V	Y	NA *	Y	Y	Y	NA †	NA †	NA *	NA *	NA *	CIF & QCIF	Y	Y	1.1 & 2.0	Ascend Multiband Plus Release 2.3
PictureTel S4000 Version 4.1V	Y	NA *	Y	Y	Y	Y	Y	NA *	NA *	NA *	CIF & QCIF	Y	Y	2.0	Ascend Multi band Plus Release 3.3A+ Release 3.4F+
PictureTel S4000 Version 4.1V	Y	NA *	Y	Y	Y	Y	Y	NA *	NA *	NA *	CIF & QCIF	Y	Y	2.0	Promptus OASIS 3.51V6
PictureTel S4000 Version 4.2V	Y	NA *	Y	Y	Y	Y	Y	NA *	NA *	NA *	CIF & QCIF	Y	Y	2.0	Ascend Multiband Plus Release 3.3A+ Release 3.4F+
PictureTel S4000 Version 4.2V	Y	NA *	Y	Y	Y	Y	Y	NA *	NA *	NA *	CIF & QCIF	Y	Y	2.0	Promptus OASIS 3.51V6
GPT Version 2.70	Y	NA *	Y	Y	Y	NC	NC	NC	NC	NC	CIF & QCIF	Y	NC	1.1	HSL to DEFINITY
SONY PCS 2000A/2000AP Version 1.04	Y	NC	NC	NC	Y	NC	NC	NA *	NA *	NA *	CIF & QCIF	Y	NC	1.1	Ascend Multi band Plus Release 2.3+
VTEL MediaMax Version 2.4	Y	NA *	Y	Y	Y	NC	NC	NC	NA *	NA *	CIF & QCIF	Y	NC	2.0	Teleos 11-1v1.1

Table A-1. Results for On-Site Tested Endpoints — *Continued*

Endpoint Version	Audio Modes		Network Connection								Video Coding Mode	Communication Modes		MCU Release(s) Tested	Terminal Adapter Release
	G.711	G.728	2x56k (112)	2x64k (128)	384k (H0)	6x56k (336)	6x64k (384)	768k	1472k	1536k (H11)		Meet-Me	Dial Out		
BT 2300 Version E2.00	Y	NC	Y	Y	Y	Y	Y	Y	NA ‡	NA ‡	CIF & QCIF	Y	Y	2.0	Ascend Multiband Plus Release 3.3A+^ Release 3.4F+^
BT 2300 Version E2.00	Y	NC	Y	Y	Y	Y	Y	Y	NA ‡	NA ‡	CIF & QCIF	Y	Y	2.0	Ascend Multiband Plus Release 2.2N^
BT 2300 Version E2.00	Y	NC	Y	Y	Y	Y	Y	NA ‡	NA ‡	Y	CIF & QCIF	Y	NC	2.0	Teleos 11-1v1.1
BT 2300 Version E2.00	Y	NC	Y	Y	Y	Y	Y	NA ‡	NA ‡	Y	CIF & QCIF	Y	Y	2.0	Promptus OASIS 3.51V6

Y — Endpoint passed a Voice-Activated Switching test with the AT&T MCU and participated in a conference.

NA* — Endpoint does not support the feature.

NA† — Terminal adapter does not support standards-based BONDing.

NA‡ — Terminal adapter does not support standards-based 768k and/or 1472k connectivity.

NC — Test was not conducted.

^ — Must be optioned for multirate ISDN to achieve 768k and 1472k connectivity.

Remotely Tested Endpoints

The following table shows the results for endpoints tested remotely from the vendor's site.

Table A-2. Results for Remotely Tested Endpoints

Endpoint Version	Audio Modes		Network Connection								Video Coding Mode	Communication Modes		MCU Release(s) Tested	Terminal Adapter Release
	G.711	G.728	2x56k (112)	2x64k (128)	384k (H0)	6x56k (336)	6x64k (384)	768k	1472k	1536k (H11)		Meet-Me	Dial Out		
BT 2300 Version E1.00 with ISDA 1001	Y	NA *	Y	Y	Y	NA †	NA †	NC	NC	NC	CIF & QCIF	Y	Y	1.1	Ascend Multi band Plus Release 3.4K
BT 7000 Version J4	Y	Y	Y	NC	NA *	NA *	NA *	NA *	NA *	NA *	CIF & QCIF	Y	Y	1.1	Direct BRI
Datapoint Version 1.4B	Y	Y	Y	Y	NA *	NA *	NA *	NA *	NA *	NA *	CIF & QCIF	Y	Y	2.0	
GPT Version 3.4	Y	NA	Y	NC	NC	NA	NA	NA *	NA *	NA *	CIF & QCIF	Y	Y	1.1	Ascend Mult band Plus Release 3.4K
Hitachi DP-200 FE1 Version M-PUI15AC (02.02.0002)	Y	NA *	Y	Y	NA *	NA *	NA *	NA *	NA *	NA *	CIF & QCIF	Y	NC	1.1	ADTRAN NT1
INTEL ProShare Room Video Conf. H.320 Version 1.8Ha	Y	NA *	Y	Y	NA *	NA *	NA *	NA *	NA *	NA *	QCIF only	Y	Y	2.0 & 3.0	Direct BRI
Mosaic GV200R Version 1.1	Y	Y	NC	Y	NA *	NA *	NA *	NA *	NA *	NA *	CIF & QCIF	Y	NC	3.0	Direct BRI from NYNEX
Panasonic KXC-M6500 Version 1.7	Y	NA *	Y	Y	NA *	NA *	NA *	NA *	NA *	NA *	CIF & QCIF	Y	Y	3.0	Hitachi Release HN 5101-1B
Panasonic KXC-M6500 Version 1.7	Y	NA *	Y	Y	NA *	NA *	NA *	NA *	NA *	NA *	CIF & QCIF	Y	Y	3.0	Adtran Release ISU 2x64
Scientific Atlanta CONEXT Version 1.0	Y	Y	Y	NA *	NA *	NA *	NA *	NA *	NA *	NA *	CIF & QCIF	Y	Y	1.1	Promptus Release 1.0
Scientific Atlanta CONEXT Version 1.1	Y	Y	Y	Y	Y	Y	Y	NA *	NA *	NA *	CIF & QCIF	Y	Y	1.1 2.0	Ascend Multi band Plus Release 3.4R+

Table A-2. Results for Remotely Tested Endpoints — *Continued*

Endpoint Version	Audio Modes		Network Connection								Video Coding Mode	Communication Modes		MCU Release(s) Tested	Terminal Adapter Release
	G.711	G.728	2x56k (112)	2x64k (128)	384k (H0)	6x56k (336)	6x64k (384)	768k	1472k	1536k (H11)		Meet-Me	Dial Out		
SONY PCS 2000A/2000AP Version 2.1	Y	Y	Y	Y	Y	NA *	NA *	NA *	NA *	NA *	CIF & QCIF	Y	NC	1.1 2.0	Ascend Multi band Plus Release 3.4R+
VIVO 320 Version 1.0	Y	NA *	Y	Y	NA *	NA *	NA *	NA *	NA *	NA *	CIF & QCIF	Y	Y	2.0	
VTEL 115/117/127 Version 1.2	Y	Y	Y	Y	NA *	NA *	NA *	NA *	NA *	NA *	CIF & QCIF	Y	Y	1.1	Ascend Mult band Plus Release 3.4
Zydacron Z200 1.0 Beta	Y	Y	Y	Y	Y	Y	Y	NA *	NA *	NA *	CIF & QCIF	Y	Y	1.1 2.0	Ascend Multi band Plus Release 3.4

Y — Endpoint passed a Voice-Activated Switching test with the AT&T MCU and participated in a conference.

NA* — Endpoint does not support the feature.

NA† — Terminal adapter does not support standards-based BONDing.

NA‡ — Terminal adapter does not support standards-based 768k and/or 1472k connectivity.

NC — Test was not conducted.

^ — Must be optioned for multirate ISDN to achieve 768k and 1472k connectivity.

Network Connectivity Testing

The AT&T-MCU can be connected to the switched network directly, or it can be connected to an AT&T PBX or an access switch that is in turn connected to the switched network. Either ISDN-PRI facilities or robbed-bit DS1 facilities can be used to connect the AT&T MCU and the network.

The following tables contain the tested networking configurations and networking services for the AT&T-MCU.

Table A-3. Network Connectivity Results for the AT&T MCU and AT&T PBXs

	AT&T-MCU Termination Arrangement Tested		
AT&T PBX	**Robbed-Bit DS1 or T1 (Switched 56)**	**ISDN-PRI***	**Comments**
DEFINITY G3iV2 PBX	Yes	Yes	† ‡
DEFINITY G3sV2 PBX	Yes	Yes	† ‡
DEFINITY G3rV2	Yes	Yes	† ‡
DEFINITY G3iV1.1 PBX	Yes	Yes	†
DEFINITY G3sV1.1 PBX	Yes	Yes	†
DEFINITY G3rV1.1 PBX	Yes	Yes	†
DEFINITY G1 PBX	Yes	Yes	†
DEFINITY G2.1 PBX	Yes	Yes	†
System 85 R2V4 PBX	Yes	Yes	†

* Tested with multiple DS1 facilities as well as with NFAS and D-channel backup.

† Network connections to the PBX can be established by first using AT&T SDDN and ACCUNET SDS network services and then private ISDN-PRI or robbed-bit DS1 facilities from the PBX to the AT&T MCU.

‡ For an H0 or switched 384 conference, the PBX should be enabled for Wideband (this PBX has Wideband capability).

Table A-4. Network Connectivity Results for the AT&T MCU and Access Switches

Access Switch	AT&T-MCU Termination Arrangement Tested	
	Robbed-Bit DS1 or T1 (Switched 56)	ISDN-PRI
Teleos Video Hub	Yes	Yes

Table A-5. Network Connectivity Results for the AT&T MCU and Central Office

Central Office	AT&T-MCU Termination Arrangement Tested		Comments
	Robbed-Bit DS1 or T1 (Switched 56)	ISDN-PRI	
AT&T 5ESS	Yes	Yes	*
AT&T 4ESS	Yes	Yes	*
Northern Telcom DMS-250	†	Yes	‡

* Network connections to the AT&T-MCU can be established by using either AT&T SDDN or ACCUNET SDS network services.
† Tests were not conducted.
‡ The AT&T MCU was directly connected to MCU ISDN-PRI.

Paper-Based Scheduling Forms

B

The blank forms on the following pages can be copied for use with a paper-based scheduling system. Make as many copies as you need for your scheduling notebook.

Included in this appendix are the following forms:

- Site Profile*
- Conference Appointment Form
- MCU-Extensions*
- Port and Extension Usage Chart

* Ask your telecommunications manager to complete this form.

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

Table B-1. Sample Conference Appointment Form

Conference Appointment Form

Conference ID _____

Conference Date: ___/___/___	*Circle choices below*
Contact Name: _____	If present or broadcast, control ext.: _____
Contact Phone: _____	If broadcast, scan interval: _____seconds (default is **15** seconds)
Password Scope: _____	Cascade Mode: **primary**, secondary, blank, disabled
Password: _____	Audio mode: **auto**, G.728 (ld_celp), G.722 (7khz), G.711 (pcm)
Conference Name: _____	Bandwidth **56k**, 64k, 112k, 128k, 168k, 192k, 224k, 256k, 280k, 320k, 384k, 768k, 1472k, 1536k, 1920k
Billing ID*:* _____	Entry/Exit Tones: **yes**, no
Class: **reserved,** dedicated, file	Warning Tone: **yes**, no
Start Time: _____:_____	Rate Adaptation: **yes**, no
Stop Time: _____:_____	Data Mode: **blank**, mcs/mlp, none
Dial-Out Number: _____	Basic/Enh _____
Password Required? y,n	Interworking: yes, no

Location	Network Type	Meet-Me Extension	Number to Dial
_____	_____	_____	_____
_____	_____	_____	_____
_____	_____	_____	_____
_____	_____	_____	_____
_____	_____	_____	_____

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

Date:

Port	Bandwidth	6:00	6:15	6:30	6:45	7:00	7:15	7:30	7:45	8:00	8:15	8:30	8:45	9:00	9:15	9:30	9:45	10:00	10:15	10:30	10:45	11:00	11:15	11:30	11:45	12:00	12:15	12:30	12:45	13:00	13:15	13:30	13:45	14:00	14:15	14:30	14:45	15:00	15:15	15:30	15:45	16:00	16:15	16:30	16:45	17:00	17:15	17:30	17:45	18:00
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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, any vintage may be used. If a vintage or suffix is listed for the circuit pack, a circuit pack with a greater vintage or suffix may be used.

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	Tone Det	Tone Gen Clock	MMI	VC
US and Canada	μ	TN748D	TN768	TN787D	TN788B
Australia	A	TN420C	TN768	TN787D	TN788B
UK	A	TN420C	TN768	TN787D	TN788B

Table C-2 and Table 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
US and Canada	TN767E	TN767E	μ
Australia	TN2207	TN2207	A
UK	TN2207	TN2207	A

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

Country	Dig Line	Encoding A/μ
US and Canada	TN754B	μ
Australia	TN754B	A
UK	TN754B	A

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

Circuit Pack Name	Description	# Ports	Supports Administrable Timers?	Encoding A/m
TN754B	**Digital Line**	**8**	**N/A**	**Cm***
TN767E	**DS1**	**24**	**Yes**	**m**
TN2207	**UDS1**	**24/32**	**Yes**	**A**

*** Software controlled — m-Law default**

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

Country	Processor Circuit Pack	AC Power Supply	DC Power Supply	Network Control
US and Canada	TN765 w/CPP1*	†	676B	TN777B
Australia	TN765 w/CPP1*		676B	TN777B
UK	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 predefined 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	US and Canada
2	Australia
3	Japan
4	Italy
5	Netherlands
6	Singapore
7	Mexico
8	Belgium
9	Saudi Arabia
10	UK
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
US and Canada	1	1	1	1
Australia	2	2	2	2
UK OTR001	10	10	10B	3
UK 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	
TN767E	DS1 Interface	
TN768	Tone Clock	Stratum 3
TN787D	Multimedia Interface	
TN788B	Voice Conditioner	

Administration

All timers and option selections default to the values appropriate for operation in the United States. 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	
TN787D	Multimedia Interface	
TN788B	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 US operation; therefore, certain values are likely to be inappropriate for international use. In particular, do 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	
TN787D	Multimedia Interface	
TN788B	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 US operation; therefore, certain values are likely to be inappropriate for international use. In particular, do 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 only the following call types be used 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 UK 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.

Application Compliant flag
A per-user flag in the Conference Record that indicates which endpoints use non-compliant applications.

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 with questions about the AT&T MCU.

Audio Add-On
Feature that allows the MCU to support up to six non-Px64 audio-only endpoints per non-cascaded conference and up to12 such endpoints per cascaded conference.

Audio Add-On Echo Control
An echo canceler function implemented in the MCU to prevent echoes caused by Audio Add-On conferees who dial into a conference through an analog circuit. Echoes occur at the two- to four-wire conversion (hybrid) in the network or in PBXs and are caused by impedance mismatches at those points.

audio port
The port used for an Audio Add-On party to join a conference. See also *port*.

Automatic 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), 1-channel multirate ISDN bandwidths of 128k, 192k, 256k, 320k, 384k, 768k, 1472k, 1536k, and 1920k, or BONDing bandwidths of 112/128k, 168/192k, 224/256k, 280/320k, and 336/384k.

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.

Basic/Enhanced Service Flag
A flag in the Conference Record that disables commands/caps that are known to cause problems (such as crashing) to specific endpoints.

BRI/DCP Direct Connect Interface
Feature that allows the user to connect BRI or DCP endpoints directly to the MCU without involving a public or private network, PBX, or MUX. The feature allows up to 12 BRI stations or up to four DCP group systems to connect directly to the MCU.

BONDing
See “bandwidth on demand.”

BONDing ports
The ports in an MCU conference that use one of the following BONDed bandwidths: 112/128k, 168/192k, 224/256k, 280/320k, or 336/384k.

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

Broadcast/scan
A conference mode where one video endpoint’s video, audio and data are broadcast to all other conferees. The broadcaster views the other locations one after another for a fixed duration of time. The broadcaster is constantly viewed by all other locations.

C

cascading
Connecting two or more AT&T MCUs 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.

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.

Conference Redial flag
A flag in the Conference Record that allows an endpoint to be "redialed" during a conference (therefore, this flag is in effect for Dial-Out calls).

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 any time.

Dedicated Access
A feature that enables endpoints to participate in multipoint audio and video conferences via non-signaled T1 facilities. For this feature, the MCU connects to a Multiplexer (MUX), a Digital Access Crosspoint System (DACS), or to H.320 endpoints via DS1 facilities. A maximum of 20 DS1 connections to the MCU are supported.

Dial-out
A feature that allows 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
A feature that allows the user to add and remove conference participants before the conference starts as well as during the conference.

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.

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.

G

G.711 Audio
See PCM.

G.722 Audio
An audio mode that enables the system to bridge G.722 (7kHz) audio. 7kHz is the audio bandwidth provided by G.722. This bandwidth allows for a more natural sounding voice conference. The MCU supports G.722-conferenced audio at 48k or 56k, as defined in ITUT-T Recommendation G.722.

G.728 Audio
See LD-CELP.

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 established in 1993 to develop standards to allow video endpoints to communicate with each other.

L

LD-CELP (G.728)
A means of encoding audio so a smaller portion of bandwidth is allocated to audio. This results in greater bandwidth for video, thus improving picture quality.

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.

Low-/High-Speed Interworking
A feature that allows 56k or 64k endpoints to join higher-speed conferences as Audio Add-On endpoints via conventional dial-in Px64 ports. Multiple Px64 ports in a conference can implement this feature simultaneously.

M

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

Manufacturer and Product Information Exchange
The MCU exchanges ns-caps with each endpoint that joins the conference. Besides information regarding capabilities, ns-cap messages contain manufacturer and product information for the MCU or endpoint. Such information received by the MCU from the endpoint is displayed on the Status Conference form.

MCS/MLP
A multipoint data conferencing feature based on the H.221 MLP feature that enables data collaboration capabilities within a conference.

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.

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.

Per-conference password
Password for a conference that is provided by the customer at conference reservation time. Conferees are prompted for the password when they join the conference.

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 (G.711)
Pulse Code Modulation. A means of encoding audio by assigning less bandwidth to video.

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 with 2B-channels that are on both 56k and 64k networks.

Real-Time Conference Status
Feature that allows the real-time status of a conference and its associated conferees to be displayed on the MCU-MT. The **monitor conference** command is used for this purpose. Once the command is issued, the display is automatically updated approximately once every 30 seconds. Accordingly, this feature is intended for troubleshooting purposes.

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

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

reservations 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

Terminal Names
A feature that enables the MCU to poll and ascertain “naming/identification” parameters from endpoints and to pass the information on to other endpoints. Therefore, conference participants whose endpoints display this information can identify other conference participants.

tones
See Notification Tones.

U

V

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

Video-Switching Mode and Broadcaster Notification
A feature used by the MCU to announce to all the conference endpoints the current video-switching mode and broadcaster. The feature is also used whenever the endpoint initiates a See-Me (User-Activated Presentation) request that cannot be honored.

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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