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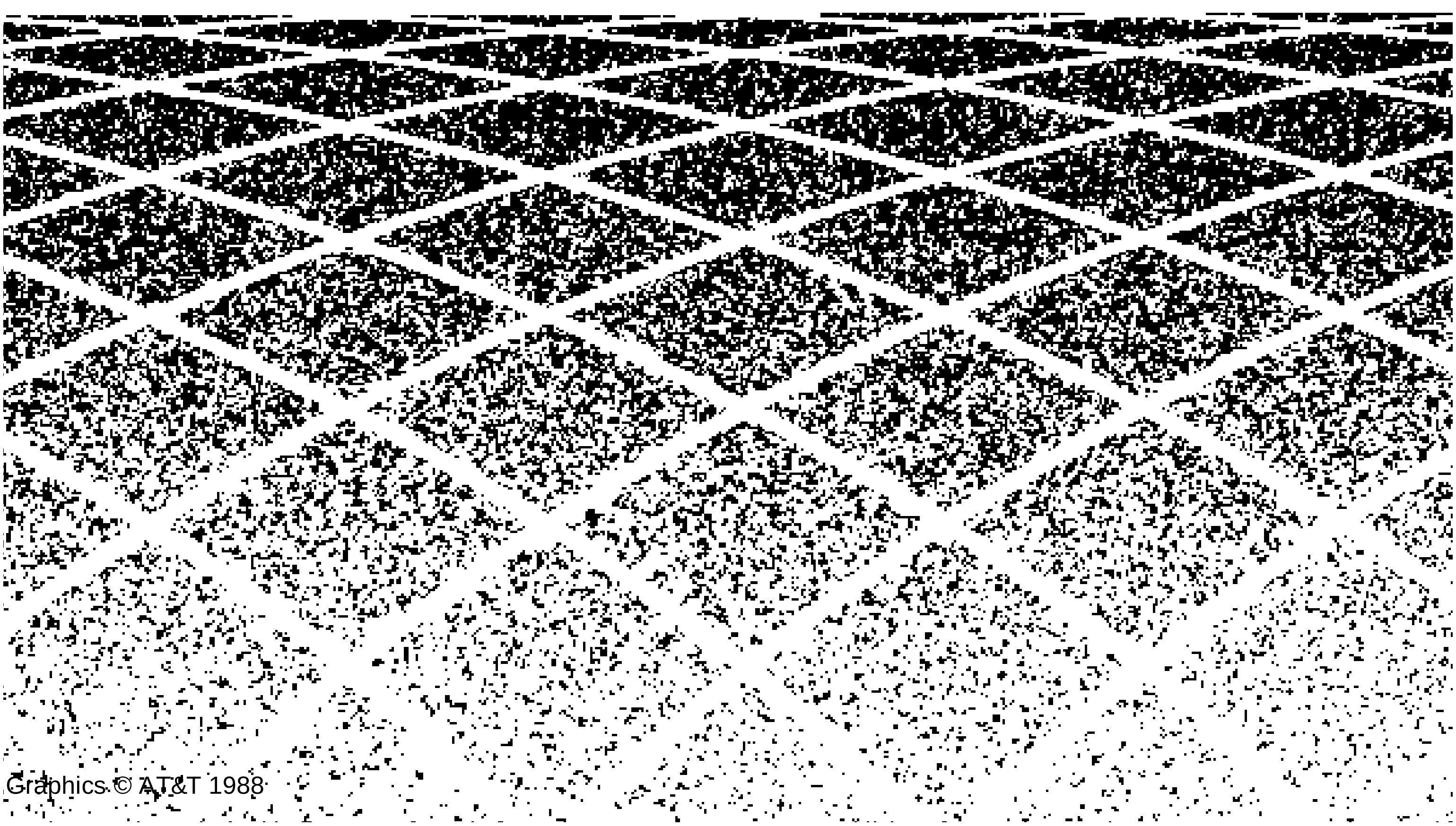
Multi-Point Control Unit R3.0 System Description



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Introducing the AT&T MCU

1

The AT&T MultiPoint Control Unit (MCU) Release 3.0 supports the WorldWorx Personal Conferencing Service (PCS) through a number of interfaces and offers feature packages in the following models:

— Customer-premises models:

- SX (Preconfigured Single-Conference MCU)
- DX
- FX
- EX

— Service provider models:

- Model WW WorldWorx model

The factors that distinguish one customer model from another are the number of ports and available options. Each AT&T MCU model provides high-quality multimedia conferencing with endpoints that communicate via the International Telecommunications Union-Telecommunications (ITU-T)* PX64 (H.320) video standard. Additionally, a single endpoint *not* using the PX64 standard can join a conference as an *audio-only* endpoint. The SX, DX, and FX models offer the ability to upgrade to the next higher models. Unless otherwise stated, the descriptions in this document are pertinent to all AT&T MCU models. In the following sections, various aspects of the AT&T MCU are discussed as an introduction to the system.

* ITU-T was formerly known as the Consultative Committee for International Telephone and Telegraph (CCITT).

Model Comparisons

Table 1-1 provides information about the WorldWorx model. The maximum number of conferences is equal to half the port capacity. In other words, one port is required for each conferee allowing 32 simultaneous 2-party conferences in a 64-port system. The AT&T MCU can support as many as 24 conferees per conference. Refer to Chapter 4, "Feature Descriptions", for more details on the AT&T MCU features.

Table 1-1. AT&T WorldWorx Model

Port Capacities	Standard Features	Options
All FX model port capacities	All standard MS and VS features except: UCC and H.261 Annex D Audio Add-On Additional Video Ports MCS/MLP Both Per-Conference and Per-User Passwords Business Card Status Conference VMS Service Indicator Notification Package: Tones Terminal Name Play Conference Tone Command Endpoint Notification of Time Left Disconnect Notification User Identification Number (UIN) WorldWorx Service Indicator	Dial-Out Accelerator Board An Additional Processor Interface Board 1 to 12 Direct Connect BRI Endpoint Connections 1 to 4 Direct Connect DCP Endpoint Connections

Physical Description

All AT&T MCU models include the following physical components:

- One to three cabinets with port, service, control, and interface circuit packs.
- One AT&T MCU management terminal (MCU-MT). (The term MCU-MT is used to refer to the WorldWorx Local MT throughout this document.)
- One maintenance alarm terminal.
- One AT&T MCU scheduling terminal (MCU-ST). (The term MCU-ST is used to refer to the terminal for the WorldWorx PCS Meeting Reservation and Control System throughout this document.)

The following components can be added to the AT&T MCU system:

- One or more customer-supplied PC remote scheduling terminals or CRS client servers with CRS software (optional).
- One or more customer-supplied system printers (optional).
- Expansion Services Module (ESM).

The AT&T MCU circuit packs are housed in one or more of the following cabinet configurations:

1. Enhanced single-carrier cabinet (ESCC) (containing a control carrier)
2. Single-carrier cabinet (SCC) (containing a port carrier)
3. Multicarrier cabinet (MCC) (containing both control and port cabinets)

The cabinets with enclosed circuit packs make up the bulk of the electronic components and contain the software responsible for system operation. The following figure illustrates an ESCC cabinet. The ESCC and SCC cabinets are identical in external appearance.

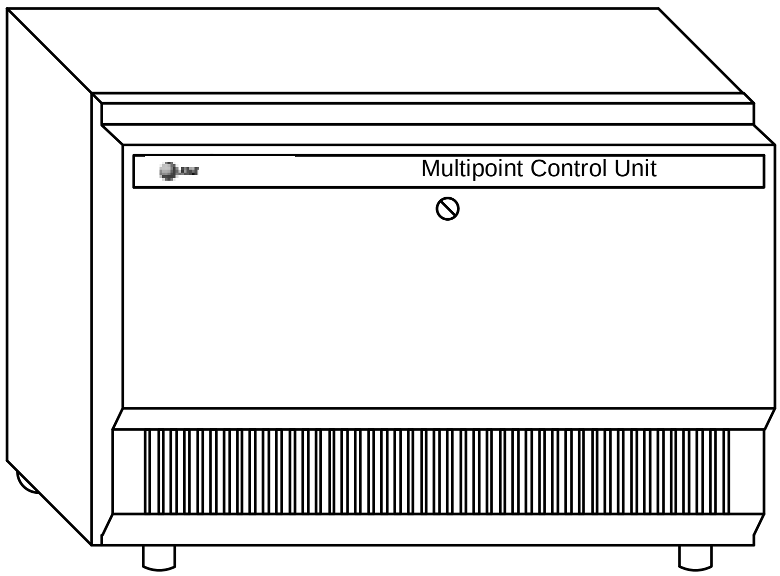


Figure 1-1. AT&T MCU ESCC Cabinet

The following figure illustrates an MCC cabinet.

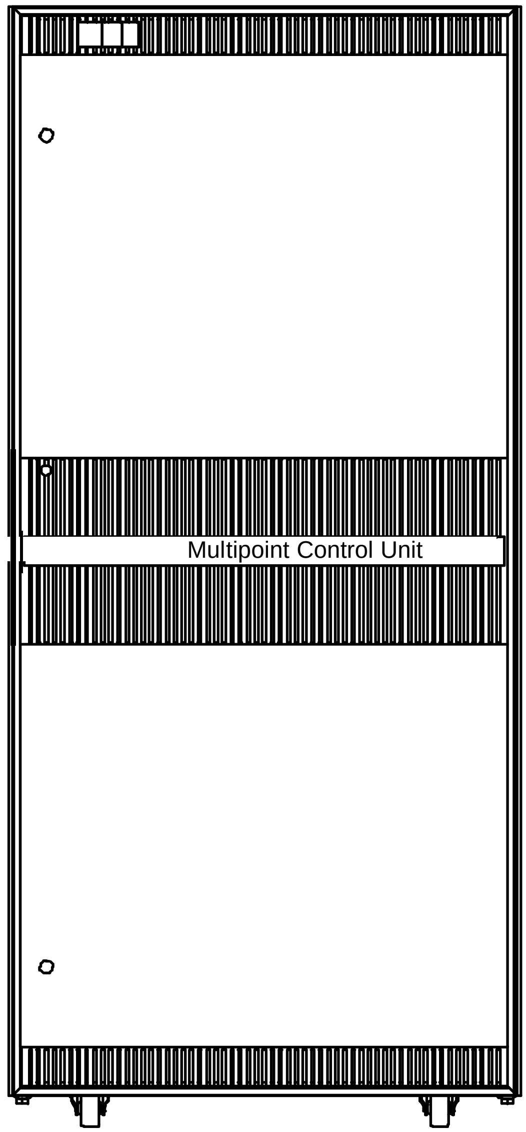


Figure 1-2. AT&T MCU MCC Cabinet

Functional Description

The following figure illustrates the AT&T MCU connections.

NOTE:
"OA&M" in the following figure stands for "operation, administration, and maintenance."

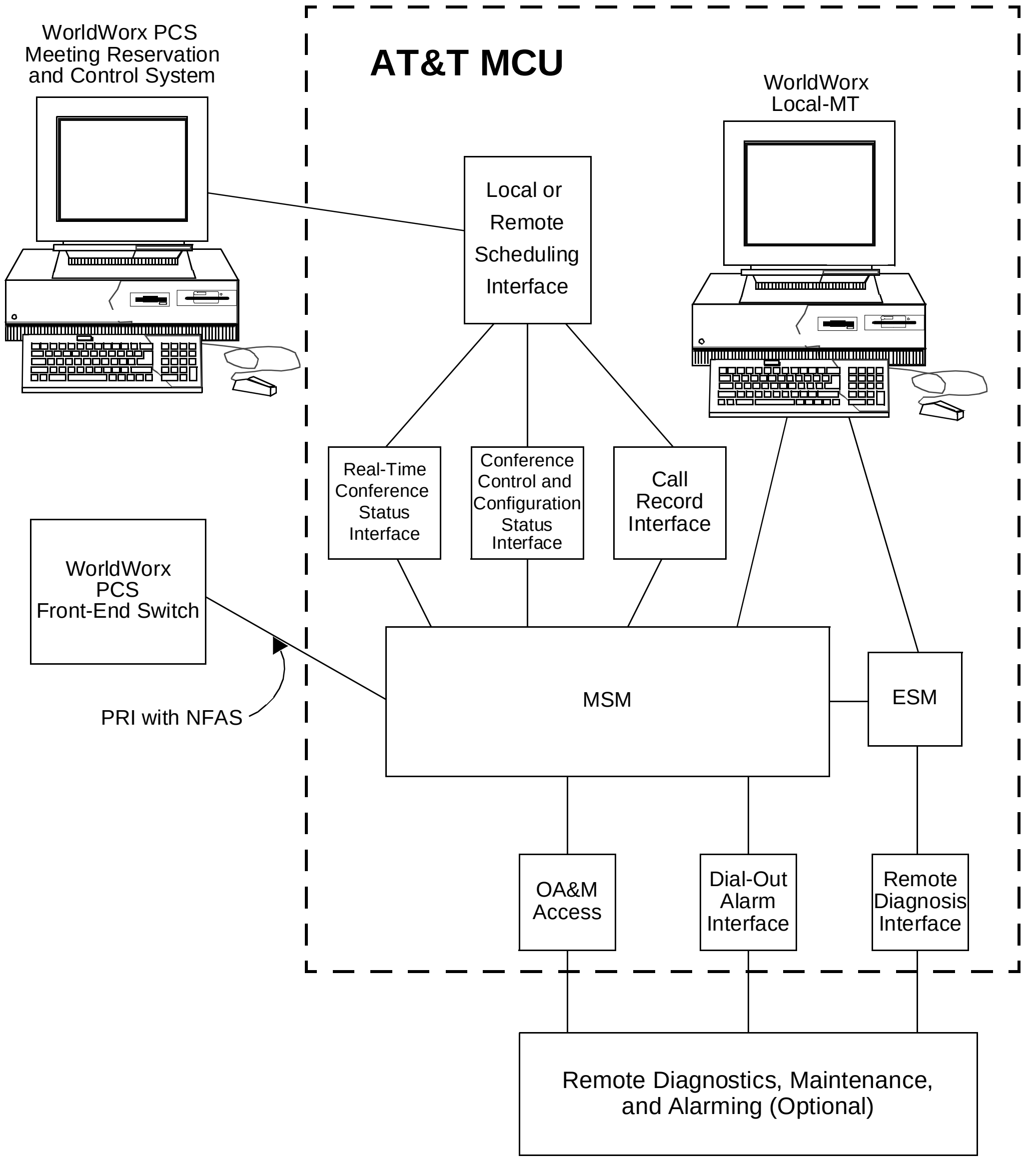


Figure 1-3. AT&T MCU System Block Diagram (WW Model)

The AT&T MCU-MT typically connects directly to the AT&T MCU via an RS-232 interface. The AT&T MCU-ST typically connects to the AT&T MCU via a dial-up connection through a data module. Either terminal can be used for administration, conference scheduling, and maintenance. The WorldWorx PCS Meeting Reservation and Control System automates conference scheduling and provides system control. The WorldWorx PCS local MT provides local maintenance and control functions and typically connects directly to the AT&T MCU via an RS232 interface. The Multimedia Server Module (MSM) provides the MCS/MLP capabilities for models with that capability. One Expansion Services Module (ESM) attaches to the cabinet containing the MSM via an E1 ISDN PRI interface. The ESM provides remote maintenance capability.

The AT&T MCU is part of a system in which two or more PX64-compliant endpoints can make a common connection to share video, audio, or data information. Endpoints can connect to the AT&T MCU either via a public communications network, such as the AT&T Switched Digital Network (SDN), or via a private branch exchange (PBX) network capable of supporting switch digital data services. Several network connection types are available, including an ISDN-PRI, ISDN-BRI, or BRI/DCP interface, depending on the model chosen. The AT&T MCU can handle a variety of video endpoints, including AT&T video conferencing systems such as the group and personal video systems. Two AT&T MCUs can be connected in a configuration known as *cascading* to join two conferences together into one joint conference. Details on endpoint interoperability are later in this chapter.

The following figure illustrates a high-level view of some AT&T MCU network connections.

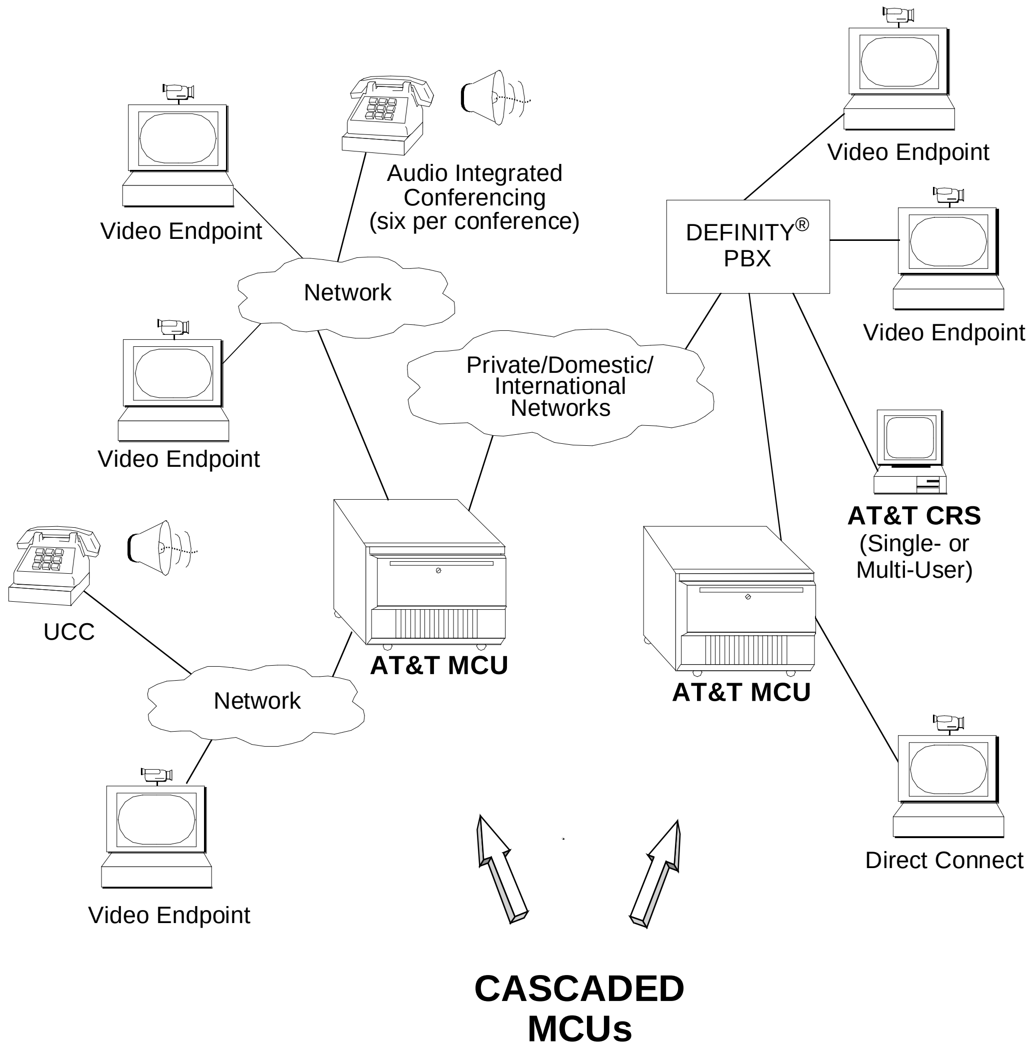


Figure 1-4. Typical AT&T MCU Network Connections

As shown, the AT&T MCU can be connected to a network via a PBX, directly to the network switch, or cascaded with another AT&T MCU.

Before a multimedia conference begins on the AT&T MCU, conference parameters are entered via the CRS remote scheduling terminal, AT&T MCU-MT, or AT&T MCU-ST.

Feature Summary

The following table provides an alphabetical listing of AT&T MCU multimedia conference features. See Chapter 4, "Feature Descriptions", for details on these and other AT&T MCU features.

NOTE:
Some of these features are optional and not included with all AT&T MCU models. See Table 1-1 for feature availability.

Table 1-2. Multimedia Conference Feature Summary

Feature	Description
Application Compliance Flag	The application compliance flag identifies endpoints (or other multipoint control units) that operate properly with MCS/MLP applications.
Integrated Audio Conferencing with Dynamic Echo Cancellation (Audio Add-On)	Allows a finite number of endpoints that do not support PX64 framing to join a conference as audio-only endpoints.
Audio Modes	The AT&T MCU supports the following audio modes: ■ G.722 without data sharing ■ G.728 (LD-CELP) with data sharing ■ G.711 (PCM) without data sharing
Basic/Enhanced Service Flag	The AT&T MCU uses this flag to identify endpoints that can operate properly with the full set of MCS/MLP commands.
BONDing	Bandwidth On Demand Interoperability Group (BONDing) allows the AT&T MCU to establish conferences (between supporting endpoints) without the use of ISDN-PRI wideband or H.221 channel aggregation. BONDING bandwidths currently supported by the AT&T MCU are: 112/128K, 168/192K, 224/256K, 280/320K, and 336/384K.
Business Cards	The business cards feature supports endpoints that supply information (such as business card data) for display at other endpoints in the conference.
Cascading	The AT&T MCU can “cascade” with another AT&T MCU. Typically, this feature is used to combine two separate conferences into one conference.
Dedicated Access	Dedicated Access allows an endpoint to participate in a conference without using signaled T1/E1 facilities. To enable this, the MCU identifies a group of DS0s that are always associated with (or dedicated to) that endpoint.
Direct Connect	This feature allows BRI or DCP endpoints to connect *directly* to the AT&T MCU instead of via a public or private network.

Table 1-2. Multimedia Conference Feature Summary — *Continued*

Feature	Description
Dynamic Resizing	With Dynamic Resizing, parties can be added or removed from the conference even while the conference is in progress.
High-Speed/Low-Speed Interworking	This feature allows lower-speed endpoints (56K/64k) to participate in higher-speed (H0, 768k, 1472k, 1536k, or 1920k) conferences as audio-only endpoints.
MCS/MLP	This feature is not accessed directly by endpoints but allows endpoints to use applications that require this feature for data conferencing. With this feature, conferees can set up sub-conferences and share data with other data-capable endpoints even if there are other endpoints in the conference that do not support data conferencing.
Notification Package	The notification package is an AT&T MCU feature that provides each of the following capabilities: ■ Terminal names feature ■ Conference tones ■ Video-switching mode notification ■ Broadcaster notification ■ Time-left notification tone ■ Disconnect notification tone ■ Play tone command
Passwords	Passwords, for an added level of conference security, are supported on a per-conference or per-user basis.
Rate Adaptation (56k/64k) for two channels	This feature allows the AT&T MCU to interwork endpoints operating with 2B-channels that are on both 56k networks and 64k networks.
Selected Communications Mode (SCM) Upgrades	Provides automatic conference configuration for audio mode, video resolution, and minimum picture interval as endpoints join or leave the conference.
Status WorldWorx Conference	Command that allows the WW model AT&T MCU to gain a status report of all the current conferences in the system.

Table 1-2. Multimedia Conference Feature Summary — *Continued*

Feature	Description
Terminal Names	Feature that allows the AT&T MCU to query and pass endpoint terminal names as defined in the ANSI243 (ANSI version of H.243) recommendation. With this feature, endpoints can identify themselves and other endpoints involved in a conference.
User Identification Number (UIN)	UIN allows endpoints to be assigned a unique identification number when they register for service.
Video Conference Control	The AT&T MCU video conference control functions (or modes) are as follows: ■ **Voice-Activated Switching** allows the current speaker to be the broadcasting endpoint. The *contributor* video function of voice-activated switching is used for tasks such as sending high-resolution video still images to other endpoints. ■ **Chair Control** allows a conference chairperson to select which endpoint video is being broadcast to all the other endpoints. ■ **Presentation** allows the MCU to broadcast a single endpoint (*presenter*) to all other endpoints. All the other endpoints are set to *Voice-Activated Switching* mode. ■ **Broadcast with Autoscan** allows the MCU to broadcast audio and video from a single endpoint to all other endpoints. The broadcasting endpoint does not receive audio but receives video from each endpoint in turn.
WorldWorx Data Compliance	Provides endpoint self-identification of WorldWorx data compliance.
WorldWorx Service Indicator	Identifies endpoints that are WorldWorx endpoints, which can have the WorldWorx “look and feel.”

System Administration

System administration is performed using the AT&T MCU-MT or AT&T MCU-ST. By accessing and editing a series of screens, conferences can be scheduled and conference options administered. The following figure is a typical system administration screen.

```
add conference 123                                          page  1 of 8
                                CONFERENCE RECORD

  Conference ID: 123                          Status: inactive
Conference Name: _____________   Conference Mode: broadcast
     Billing ID: ________            Control Ext: _____ Scan Int: 15_
 Password Scope: per-user            Cascade Mode: _________
                                       Audio Mode: auto
          Class: reserved_              Bandwidth: 56k__
     Start Time: __:__           No. of Channels: 2
      Stop Time: __:__          Entry/Exit Tones? y
                                    Warning Tone? y
                                 Rate Adaptation? n
                               MCS/MLP Data Mode: ww-pcs
                              Lo/Hi Interworking? n__
```

Figure 1-5. Typical System Administration Form

Endpoint Interoperability

AT&T provides a total end-to-end solution for multimedia conferencing with a varied selection of endpoints that can interface with the AT&T MCU. In addition, other vendor endpoints are being tested as they become available. The AT&T MCU is designed to operate with any endpoint that complies strictly to the Px64 standard.

Specifically, the endpoints in the following two tables have successfully completed interoperability testing with the AT&T MCU at the time of this printing.

Table 1-3. AT&T Endpoint Interoperability

Endpoint	Release or Model Number
AT&T Group Video S4000 Systems	Models 150E, 200E, 400E, 600E, and 800E; Versions 3.0P.03, 4.1V, and 4.2V
AT&T Group Video S1000 Systems	Models 30 and 50, Versions 1.0, 1.1, and 1.1C
AT&T Vistium™ Video	Versions 1.01, 1.00.12, 1.03.10, 1.04.01, and 2.0

Table 1-4. Non-AT&T Endpoint Interoperability

Endpoint	Release or Model Number
BT 2300	Versions E2.00 and E1.00 with ISDA 1001
BT 7000	Version J4
Compression Labs, Incorporated (CLI®) Rembrandt® II/VP	Version 7.94
CLI Eclipse	Version 1.1
CLI Radiance	Version 9.18
Datapoint	Version 1.4B
GPT Video Systems	Versions 2.70 and 3.4 Hitachi™ DP-200 FE1 Version M-PU15AC (02.02.0002)
Intel ProShare Room Video	Version 1.8Ha
Mosaic GV200R	Version 1.1
Panasonic KXC-M6500	Versions 1.7 and 1.9
PictureTel™	System 1000™ Versions 1.0, 1.1, and 1.1C
PictureTel	System 2000™ Version C0.70
PictureTel	System 4000™ Versions 3.0P.03, 4.1V, and 4.2V
PictureTel	LIVE PCS-100™ Version 1.0
Scientific Atlanta CONTEXT	Versions 1.0 and 1.1
SONY PCS-2000A/2000AP	Versions 1.04 and 2.1
VIVO 320	Version 1.0
VTEL MediaMax™	Version 2.4
VTEL 115/117/127	Version 1.2
Zydacron Z200	Version 1.0

AT&T testing determined only Px64 compatibility and not a level of video and audio quality. Video and audio quality varies from endpoint to endpoint. Past interoperability with a particular endpoint is not assurance of future interoperability with that endpoint.

Ordering MCU Numbers

MCU numbers must be ordered from network services when the AT&T MCU is ordered. Order at least one network access number for each AT&T MCU port. For example, if the AT&T MCU has 16 2-channel ports and eight 384k (H0) ports, order at least 16 access numbers.

If your system is to be used at or near port capacity for many conferences, order twice as many ports as the maximum number of AT&T MCU ports. For example, if the AT&T MCU has 16 2-channel ports and eight 384K (H0) ports and you are going to schedule many conferences, order 32 access numbers.

Upgrades

The AT&T MCU models offer the following upgrade options:

- DX to FX
- DX to EX
- FX to EX
- Release 1.1 to 2.0
- Release 2.0 to 3.0

Additions

The following additional options are available:

- 4-port increments for 2-channel, 384k (H0), BONDing, and higher-speed ports
- Integrated Audio Conferencing (Audio Add-On) option (in 2-port increments)
- BONDing capability option with BONDing administration
- For MCC configurations, processor interface board

For more information about upgrading, call your account executive.

Architecture

2

The AT&T MultiPoint Control Unit (MCU) provides the functionality required for multimedia conferencing. Figure 2-1 illustrates the AT&T MCU system functional architecture.

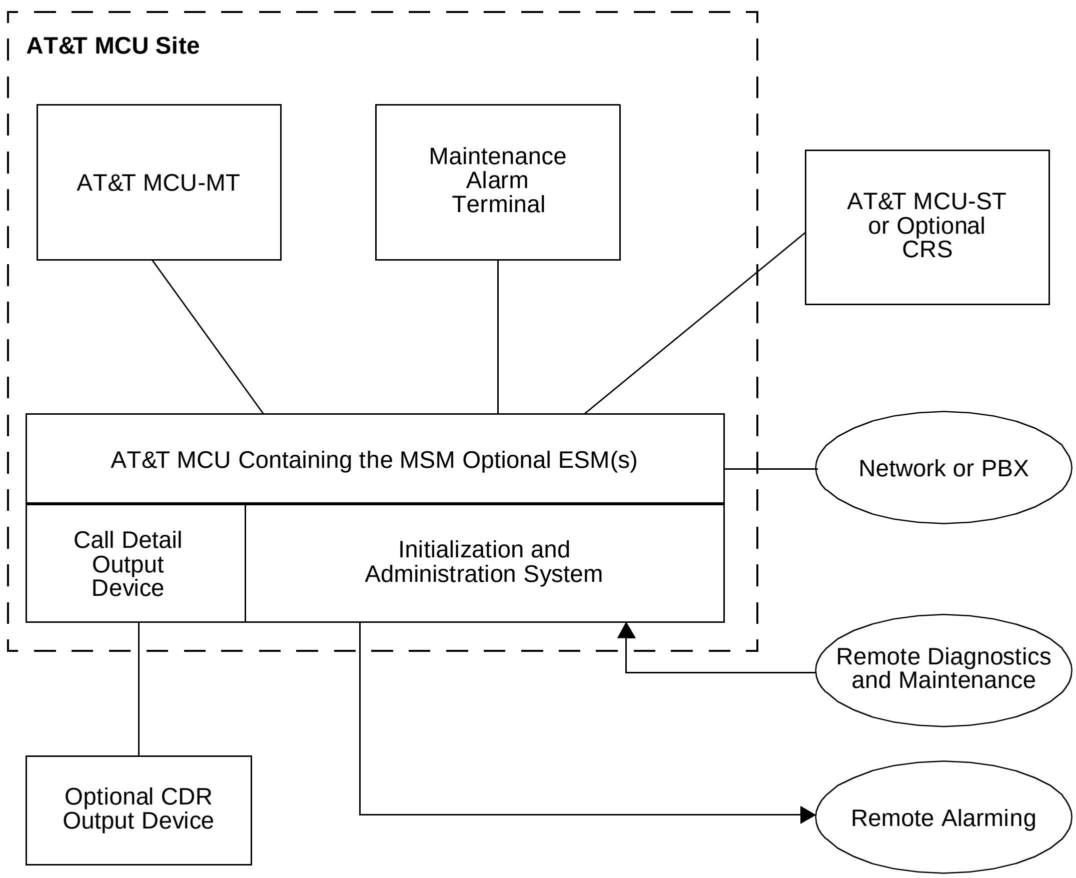


Figure 2-1. System Architecture

> **NOTE:**
> The AT&T MCU-ST can be located on or off the AT&T MCU site.

Hardware Architecture

An AT&T MCU system includes the following components:

- One or more cabinets with port, service, control, and interface circuit packs contained in a Multimedia Server Module (MSM) and an optional Expansion Service Module (ESM).
- One management terminal (MCU-MT).
- One maintenance alarm terminal.
- One scheduling terminal (MCU-ST).

The following components can be added to your AT&T MCU system:

- One or more customer-supplied system printers (optional).

The basic component of the system is a cabinet containing control and port circuits with supporting power supplies. The port circuits are connected to internal common buses that allow them to communicate with one another. The AT&T MCU consists of:

- **MSM:**
  - Time-division multiplex (TDM) bus, which runs internally throughout each cabinet in the system, and is terminated by resistors on each end. The TDM bus consists of two eight-bit parallel buses: TDM bus A and TDM bus B. TDM buses A and B carry switched digitized voice and data signals, and control signals continuously between all port circuits and between port circuits and the processing element. The port circuits place digitized voice and data signals on a TDM bus. TDM bus A and TDM bus B are normally active simultaneously. If one TDM bus fails, the other TDM bus takes over.

    NOTE:
    If one of the TDM buses fails and the other takes over, there is noticeable service degradation in the form of reduced capacity.

  - Flash-ROM, which contains system software and translations.
  - Port circuits, which provide the links between external trunks, lines and communications equipment, and the TDM bus.
  - Service circuits, which provide the following:
    - PX64 framing and termination.
    - Tone detection and generation.
    - Voice conditioning.
    - Connections to external terminal(s) for monitoring, maintaining, and troubleshooting the system.

— **ESM:** (Optional)

- The ISA Bus, which runs through the entire ESM cabinet and is terminated at each end. It is a 16-bit bus that provides the interface and control link between ESM circuit packs.
- Independent processor that controls ESM operation within the MCU.
- MSM interface circuits that connect the ESM with the MSM via an E1 high-speed data link interface.
- Remote maintenance circuits that provide access to the MCU from remote maintenance facilities for the following tasks:
  - — Remote cold boot of the ESM.
  - — Remote shutdown of the operating system.
  - — Initiation of ROM diagnostics that test the system board, memory, and video circuits.
  - — Auto monitoring of the operating system and auto reset of the ESM if the operating system fails to handshake with the remote maintenance circuit board without technician intervention.
  - — Remote log to the ESM if the ESM is down (provided the ESM power supply is still operating properly).

Software Architecture

The software architecture can be divided into the following categories for both the MSM and ESM:

- Operating system layer
- Application layer
- Conference reservations
- Internal connectivity

Operating System Layer

Oryx/Pecos is the AT&T proprietary real-time multiprocessing operating system used in the AT&T MCU MSM. Oryx/Pecos allows for multiprocessing via interprocess message passing. Drivers provide for peripheral interfaces to the system.

The ESM operating system is UNIXWARE™, which provides a familiar environment for technicians and users.

Application Layer

The application layer is made up of the following major subsystems:

- Call processing
- Management (administration)
- Maintenance

Each subsystem is described in the following sections.

Call and Data Processing

Call processing is the sequence of events required to connect, disconnect, and manage multimedia conference calls. A conference can be administered for one or more of the following video control modes:

- Voice-activated switching (also referred to as automatic control)
- Chair control (part of user control)
- Presentation (part of advanced control)
- Broadcast with autoscan (part of advanced control)

Conference modes are discussed in Chapter 4, "Feature Descriptions".

Management (System Administration)

Management software controls the internal processes to install, administer, and maintain the system. A layered software architecture presents capabilities to the user in as simple and straightforward a manner as possible while the internal complexity of the system remains transparent.

Through the use of an on-line video display terminal, management software permits a customer or technician to install, test, rearrange, and change equipment and services, and select user and system options.

System management software provides four functions, all of which are available through a terminal:

- *Measurement collection and reporting* — formatted reports of hourly traffic data on engineered resources such as trunk groups.
- *Maintenance testing and reporting* — demand testing of circuit packs, terminal equipment, and the display of system error and alarm logs.
- *Translation data backup* — backup of translation data on flash ROM.
- *Translation database management* — which provides four functions:
  - *Data view mapping* allows a system administrator to display and change all translation data related to a station, trunk, or feature as a single task.
  - *Database validation* ensures that data entered into the system is individually correct and consistent with other data; for example, that MCU extensions assigned to conferences are consistent with the dialing plan.
  - *Form transactions* ensure that all the translation data entered on a system form is either accepted as valid or rejected as inconsistent.
  - *Concurrency control* allows for multiple terminal users and ensures that AT&T MCU system software does not use critical data that is being changed.

Maintenance

Maintenance software offers a high level of service availability with minimum disruption to the system. Its interface with other software and hardware provides a quick and highly reliable fault-detection system and recovery action if possible. If a problem occurs that cannot be solved by recovery action, LEDs on the circuit packs and/or alarm and error logs quickly indicate isolatable component faults to remote trouble isolation or the on-site system technician.

Each of the following areas contributes to overall system reliability:

- Initialization — each software or hardware component maintained must be "initialized" (processes started, ports cleared, etc.). Maintenance software initializes the system at boot time, including creating and starting processes, and inserting the circuit packs and ports.
- Processing element recovery — recovery restart levels aid in maintaining processing element stability over transient processing element hardware or software errors. The processing element is the control complex that runs call processing, maintenance, and administration software.
- Hardware background testing — extensive background testing is done by firmware and hardware on the circuit packs; when problems are found, in-line error messages are sent to maintenance software on the processing element.
- Maintenance software periodic and scheduled testing — periodic tests (nondestructive tests) are typically run once an hour. Scheduled tests (including destructive tests) are run once a day. Whenever appropriate, maintenance software runs either periodic or scheduled tests to ensure that all errors are found and recovery or alarming can take place.
- Error analysis — maintenance software increments software counters, performs tests, and/or recovery actions when the following situations occur:
  - In-line errors are reported (typically by firmware)
  - Other errors are reported from software processes
  - When periodic, scheduled, or demand testing for maintenance objects is performed

  When software error counters go over threshold, additional testing and/or recovery is performed as appropriate.
- Demand testing — various demand tests can be run to check on the sanity of the system and individual maintenance objects. A "test short" command that includes destructive tests and a "test short" command that has only nondestructive tests can be used for most hardware maintenance objects.
- Busyout and release — this allows system technicians to remove components from normal service for testing and troubleshooting and to bring them back into normal service after testing.
- Other miscellaneous activity — these include control and manipulation of emergency transfer, power and environmental sensing, and treatment of the whole system as a maintained component.

Conference Reservations

The conference reservation software provides the following:

- Response to user requests for MCU numbers, and reservations based on the required video bandwidth
- Entry and scheduling of video conferences and conference information
- Management and tracking of conference resources
- Response to trouble reports

Video conferences are managed using *conference records*. Typical conferences can be stored as templates in *conference files*. The AT&T MCU accepts video scheduling up to 24 hours prior to the desired conference end time. AT&T MCU schedules can be entered with either a PC equipped with Conference Reservation System (CRS) software or the MCU-MT. One advantage of using a PC with CRS software is that conferences can be scheduled more than 24 hours prior to the conference end time and the AT&T MCU can be notified within its 24 hour requirement.

Processor Complex Connectivity

The AT&T MCU has two kinds of links into the processing element: system links and application links. System links are the ISDN signaling links for internal system control. Application links are used for peripherals such as Call Detail Recording (CDR) devices.

The ISDN signaling system link provides for ISDN connectivity through the DS1 circuit pack to PRI trunks. The application links provide for connectivity through the network control data channel (RS-232 interface).

Components

3

An AT&T MCU system is made up of one or more cabinets containing the system electronics (circuit packs), an AT&T MCU scheduling terminal (MCU-ST), and an AT&T MCU management terminal (MCU-MT). All AT&T MCUs contain a Multimedia Server Module (MSM) in the form of control/port carriers. Systems that support the Multipoint Communication Service (MCS)/Multilayer Protocol (MLP) feature require an Expansion Services Module (ESM). An ESM provides the resources necessary to support 24 MCS/MLP ports. Initialization and administration system circuits in the AT&T MCU cabinets provide for remote alarming, diagnostics, and maintenance.

Cabinets

The AT&T MCU system supports one multicarrier cabinet (MCC), one enhanced single-carrier cabinet (ESCC), or one ESCC with as many as three single-carrier cabinets (SCC). Every AT&T MCU system must contain at least one control carrier in the MCC or in the form of an ESCC.

The following sections describe the MCC, ESCC, and SCC cabinets.

Multicarrier Cabinet

Figure 3-1 illustrates the MCC configuration as seen from the front of the open cabinet.

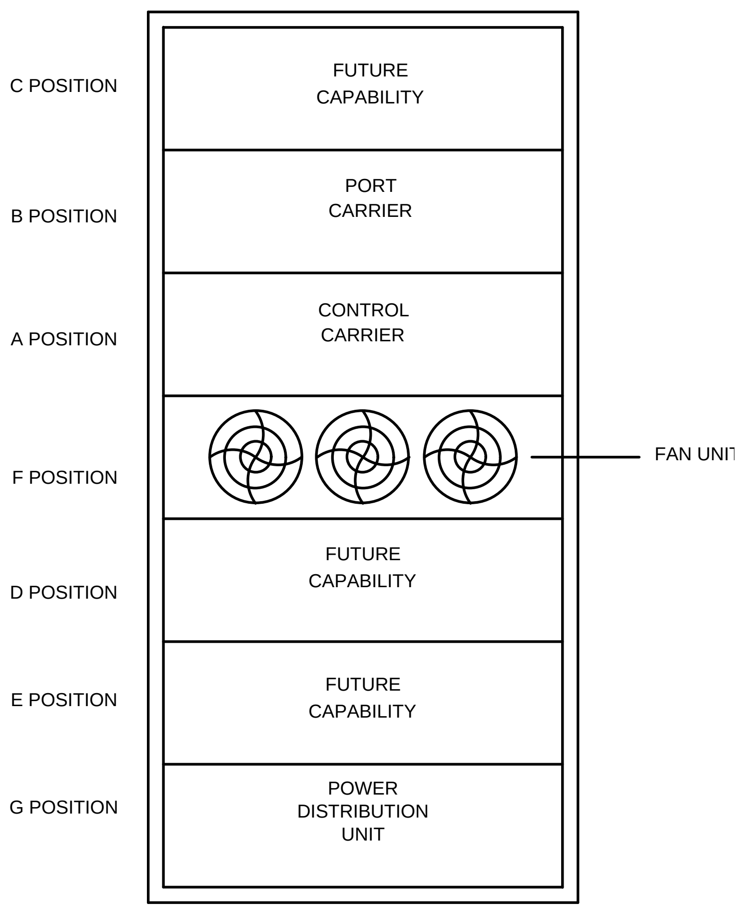


Figure 3-1. Multicarrier Cabinet Configuration

Positions and equipment in the cabinet are identified with letter designations as follows:

- Control carrier, position ‘‘A’’
- Port carrier, positions ‘‘B’’ through ‘‘D’’
- Position ‘‘E’’ provides for future system growth (blank faceplates are provided)
- Fans, position ‘‘F’’
- Power distribution, position ‘‘G’’

Doors on the front and rear of the cabinet protect the internal equipment and allow easy access. The front cabinet door is secured by screw-type latches located on the left side of the front door. The two doors at the rear of the cabinet are secured by screw-type latches located in the middle of the cabinet. Turning the screws clockwise loosens the latches so the front and rear doors can be opened.

Slotted areas at the top and bottom of the cabinet’s front and rear are used for air circulation.

Each cabinet is equipped with casters. When a cabinet is in place, leveling screws keep it from rolling. Each corner of a cabinet can be bolted to the floor when required.

MCC Dimensions and Weight

Table 3-1 lists the MCC dimensions and average weights.

Table 3-1. MCC Dimensions and Average Weights

Height	Width	Depth	Weight
70 inches (178 cm)	32 inches (81 cm)	28 inches (71 cm)	800 lb (360 kg)

Heat Dissipation

The average heat dissipation of an MCC is 2420 British Thermal Units (BTUs) per hour for a two-carrier system, and 2600 BTUs per hour for a three-carrier system.

MCC Control Carrier

The following figure provides a rear view of the MCC control carrier. The following table describes the functions for each MCC control carrier connector.

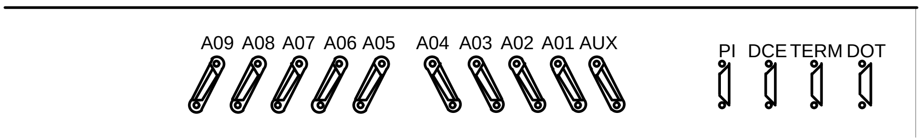


Figure 3-2. MCC Control Carrier Rear View

Table 3-2. MCC Control Carrier Connectors

Connector	Function
1 to 9 25-pair	Interfaces between port circuit packs and the cross-connect field.
AUX (auxiliary)	Connects a tip/ring pair to the cross-connect field, customer alarms, and emergency power transfer.
PI (processor interface)	Connects directly to the processor interface circuit pack. This connector is disabled in DC-powered systems.
DCE (digital communications equipment)	Connects the processor to the CDR equipment. It can be used for AC or DC configurations.
TERM (terminal)	Connects a terminal to the processor.
DOT (duplication option terminal)	Not used.

MCC Port Carrier

The following figure provides a rear view of the MCC port carrier. The following table describes the functions for each MCC port carrier connector.

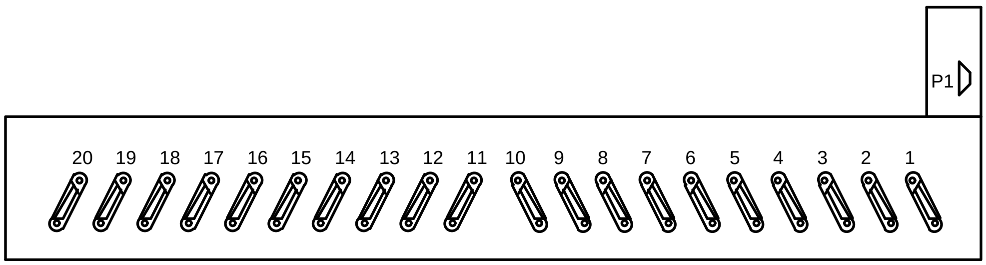


Figure 3-3. MCC Port Carrier Rear View

Table 3-3. MCC Port Carrier Connectors

Connector	Function
1 through 20	Ports that are interfaces between the circuit pack slots and the cross-connect field
P1	Provides the following: ■ Position indicator of port carrier ■ Ringing voltage input to carrier ■ Access to alarm and control circuits

Single and Enhanced Carrier Cabinets

The ESCC is designed to contain a control carrier. The SCC is designed to contain a port carrier. An SCC can be stacked on top of an ESCC to provide one port and one control carrier. When stacked, the ESCC is said to be the ‘‘A’’ carrier and the SCC is said to be the ‘‘B’’ carrier.

A screw-type latch, located below the identification stripe, secures the front door to the cabinet. Turning the screw counterclockwise loosens the latch and releases the door. Two holes in the rear bottom of a cabinet can secure it to the floor to provide earthquake protection.

Cabinet clips in the front of the cabinets connect the cabinets. At the rear of the cabinets, a ground plate connected between cabinets provides ground integrity.

SCC/ESCC Dimensions and Weight

The following table lists the SCC and ESCC dimensions and average weights.

Table 3-4. SCC/ESCC Dimensions and Average Weights

Height	Width	Depth	Weight
20 inches (51 cm)	27 inches (69 cm)	22 inches (56 cm)	125 lb (56 kg)

Heat Dissipation

The average heat dissipation of an ESCC/SCC system is 1200 BTUs per hour for a single carrier, and 2420 BTUs per hour for a two-carrier system.

ESCC Control Carrier

The following figure provides a rear view of the ESCC control carrier. The following table describes the functions for each ESCC control carrier connector.

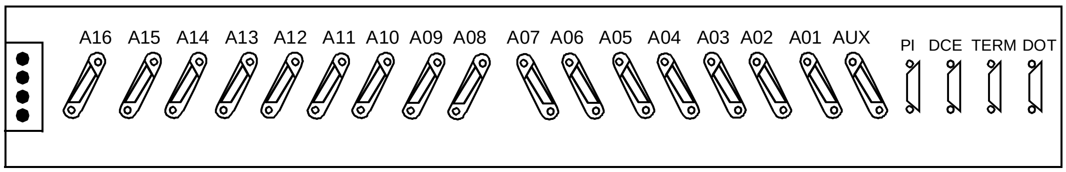


Figure 3-4. ESCC Control Carrier Rear View

Table 3-5. ESCC Control Carrier Connectors

Connector	Function
1 through 16 25-pair	Ports that are interfaces between the circuit pack slots and the cross-connect field. **NOTE:** Within the ESCC cabinet, only 14 of the 16 circuit pack locations can be used.
PI (processor interface)	Connects directly to the processor interface circuit pack. This connector is disabled in DC-powered systems.
DCE (digital communications equipment)	Connects the processor to the CDR equipment. It can be used for AC or DC configurations.
TERM (terminal)	Connects a terminal to the processor.
DOT (duplication option terminal)	Not used.

SCC Port Carrier

The following figure provides a rear view of the SCC port carrier. The following table describes the functions for each SCC port carrier connector.

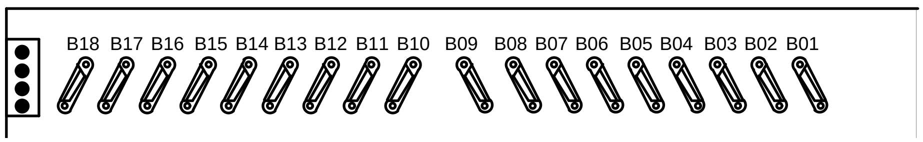


Figure 3-5. SCC Port Carrier Rear View

Table 3-6. SCC Port Carrier Connectors

Connector	Function
1 to 18	Ports that provide interfaces between circuit packs and the cross-connect field.

Circuit Packs

Circuit packs provide power distribution, control, and processing within carriers.

The circuit packs contain solid-state circuits mounted on printed wiring boards. All circuit packs are approximately 8 in. (20 cm) by 13 in. (33 cm). A 200-pin connector is attached to each TN-labeled pack. Faceplates on the circuit packs are sized to fill the width of a slot, which is typically .75 in. (1.9 cm). Each faceplate has a standard pattern of three colored light-emitting diodes (LEDs) that indicate different circuit pack conditions:

- Red—which indicates a fault condition
- Green—which indicates a test in progress
- Yellow—which indicates an in-use condition

A special grounding latch on each pack protects it from electrostatic discharge during installation. An electrostatic discharge wrist strap must be used when installing a circuit pack in a cabinet. Four types of circuit packs can be installed into carriers:

- Port circuit packs, which provide links between digital lines, trunks, networks, and external communications equipment, and the system's TDM bus. These circuit packs can be installed in any port slot.
- Control circuit packs, which include processor, network control, protocol interfaces, administration, and maintenance. These circuit packs are installed in dedicated slots in the control carrier and do not work in any other slots, including port slots.
- Service circuit packs, which terminate the PX64 protocol, condition voice calls, and detect and supply tones.
- Power unit circuit packs, which supply DC voltages to the port, and control circuit packs in the carriers and single-carrier cabinets. These required circuit packs are installed in indicated white slots in all carriers and single-carrier cabinets.

A color code on each faceplate identifies the circuit type and can be matched for installation by the same color below the carrier slot.

AT&T MCU circuit packs are listed in order of their respective code numbers in the following table.

Table 3-7. Circuit Packs

Code	Name	Type
631DA1	AC Power Unit	Power
631DB1	AC Power Unit	Power
644A1	DC Power Unit	Power
645B1	DC Power Unit	Power
676B	DC Power Unit	Power
982LS	Current Limiter	Power
CPP1	Memory Expansion	Control
ESM-HDLC	MSM Interface Card	Service
ESM Processor	ESM Processor	Control
RMB	ESM Remote Maintenance	Service
TN556	ISDN-BRI	Control
TN744B	Call Classifier	Service
TN748D or TN420C	Tone Detector	Service
TN754B	Digital Line	Port
TN765	Processor Interface	Control
TN767D or TN2207	DS1 Interface and MSM-ESM Data Link	Port
TN768 or TN780	Tone-clock	Control
TN771	Maintenance/Test	Service
TN777B	Network Control	Control
TN778	Packet Control	Control
TN786B	Processor	Control
TN787D	Multimedia Interface	Service
TN788B	Voice Conditioner	Service
WP-91153	AC Power Unit	Power
ZAHFA	TDM/LAN Bus Terminator	Control

The following sections describe each circuit pack in the order in which they are listed in the previous table.

631DA1 - Power Unit for MCC

The 631DA1 accepts 120VAC 60Hz and produces +5VDC at 60A. The +5VDC is distributed on the carrier backplanes to circuit pack slots in the carriers.

During normal operation, the 631DA1 converts the 120VAC input to +5VDC. If the unit's AC input power fails, the unit converts 144VDC supplied by optional batteries in the AC power distribution unit to +5VDC. A circuit in the battery charger unit detects the highest equivalent AC or DC input voltage and switches in the correct input voltage.

631DB1 - AC Power Unit for MCC

The 631DB1 accepts 120VAC 60Hz and produces -48VDC at 8A and -5VDC at 6A. The -48VDC and -5VDC are distributed on the carrier backplanes to circuit pack slots in the carriers. The -48VDC also supplies power to the cabinet fans.

During normal operation, the 631DB1 converts the 120VAC input to -48VDC and -5VDC. If the unit's AC input power fails, the unit converts 144VDC supplied by optional batteries in the AC power distribution unit to -48VDC and -5VDC. A circuit in the optional battery charger unit detects the highest equivalent AC or DC input voltage and switches in the correct input voltage.

644A1 - DC Power Unit for MCC

The 644A1 converts a -48VDC input to a +5VDC output at 60A. The +5VDC is distributed on the carrier backplanes to circuit pack slots in the carriers.

645B1 - DC Power Unit for MCC

The 645B1 converts a -48VDC input to outputs of -48VDC at 8A and -5VDC at 6A. The -48VDC and -5VDC are distributed on the carrier backplanes to circuit pack slots in the carriers. The -48VDC also powers the cabinet fans.

676B - DC Power Unit for ESCC/SCC

The 676B DC power supply is used in SCC or ESCC configured for DC power. A single multi-output DC power supply is located in the power supply slots. The following figure shows the DC power supply.

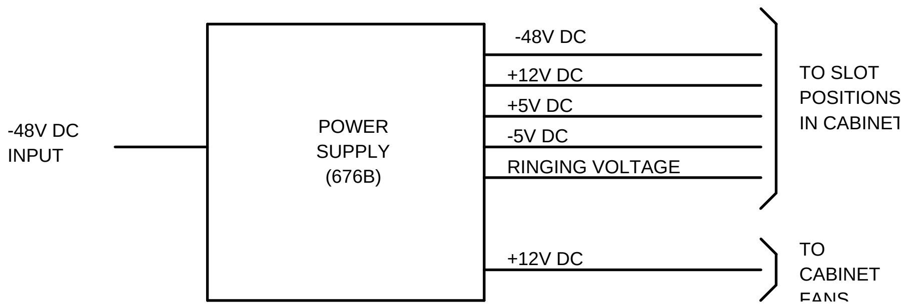


Figure 3-6. DC Power Supply (676B) in Single-Carrier Cabinet

A -48VDC source supplies power to the DC power supply.

The DC power supply produces the following DC outputs: +5V, -5V, -48V, +12V, and a ringing voltage. The DC outputs are distributed on the cabinet backplane to the slots for the circuit packs. The power supply has circuit breakers and Electro-Magnetic Interference (EMI) filtering.

982LS - Current Limiter

The 982LS connects to the back of the processor circuit pack slot and provides the following:

- Current-limited accessory 48VDC
- Emergency transfer logic
- Current-limited 5VDC to trip main circuit breaker when there is high temperature
- Duplicated 48VDC to fan units in the cabinet

CPP1 - Memory Expansion

The CPP1 circuit pack provides additional memory to enhance system software performance.

MSM Interface Card

The MSM Interface Card is a DS1/E1 interface pack in the ESM that provides the ESM side of the ESM-MSM high-speed data link. It is a standard ISA bus board for all AT&T MCU systems that use an ESM.

The ESM-HDLC fits in a 16-bit slot in the ESM and terminates one or two E1s with an on-board DSU.

ESM Processor

The ESM processor circuit pack has two primary tasks:

- Responds to the following messages from the SPE:
  - Start conference. Data includes the number of ports and DS1 channels to use and the MCS domain parameters.
  - End conference
  - Drop a party
  - Add a party
- Implements parts of the ISDN protocol stack that do not fit onto the HDLC protocol resource.
- Implements the MCS protocol specified in T.125.

RMB - Remote Maintenance Board

The RMB remote maintenance circuit pack contains hardware and software to allow remote and local ESM testing. It continuously monitors temperature and power conditions and responds with appropriate actions if those parameters exceed established boundaries. The RMB circuit pack provides for:

- Remote ESM cold boot
- Remote ESM Unix operating system shutdown
- Built-in ROM diagnostics for the ESM system board, memory, and video circuits
- Automatic operating system monitoring and ESM reset if the operating system fails to handshake with the RMB
- Remote login to the ESM even if the ESM is down (provided ESM power is not lost)

The RMB circuit pack is an ISA bus board that is required in all AT&T MCU systems that use an ESM.

TN556 - ISDN-BRI

This circuit pack provides connections for ISDN-BRI station sets and data modules. The Packet Bus is used to carry signaling information for sets and data modules.

TN744B - Call Classifier Board

The TN744B has eight detectors that detect tones in the Dial-Out feature. The TN744B detects special intercept tones used in network intercept tone detection in the dial-out application. The TN744B does not classify data calls. Data calls are classified by the TN748C Tone Detector circuit pack. In addition, if the TN744B has not classified the call by the end of 60 seconds, it is removed from the call and Timed Far-End Supervision is used to classify the call.

TN748D or TN420C - Tone Detector

The TN748D or TN420C has four touch-tone receivers and two general-purpose tone receivers that detect the following: call progress tones, modem answer-back tones, transmission test tones, and noise.

The TN748D is designed for the United States, while the TN420C is for Australia, Singapore, and the United Kingdom.

TN754B - Digital Line

The TN754B has eight asynchronous Digital Communications Protocol (DCP) ports that can be connected to an AT&T 7444D maintenance alarm terminal or 7400B data module.

TN765 - Processor Interface

The TN765 has four data links to the TDM bus and a link through the memory bus to the processor. This circuit pack provides the Integrated Digital Services Network (ISDN) interface. The TN765 allows direct access to one data link from an EIA port on the circuit pack in AC powered systems. The other data links connect to a digital line circuit and ISDN. Data links can connect to DS-1 tie trunks to access ISDN applications.

The TN765 terminates ISDN-LAPD protocols. The multicarrier cabinet supports two TN765 circuit packs, providing a total of eight data links. The ESCC supports one TN765 circuit pack to provide four data links each.

TN767D or TN2207 - DS1 Interface

The TN767D allows DS1 and ISDN-PRI signaling to be carried over DS1 facilities. Each also allows the ISDN-PRI signaling to be carried on any of the 24 trunk ports between the TDM bus and the DS1 facility. The DS1 interface circuit pack also performs robbed-bit signaling using TIE signaling protocol in any remaining ports.

The TN2207 provides the same functions but for 30 trunk ports. A TN2207 must be used in the MSM for the ESM data link.

The TN767D is designed for the United States, while the TN2207 is designed for Australia, Singapore, and the United Kingdom.

TN768 and TN780 - Tone-Clock

The TN768 or TN780 supplies timing, including Stratum 4 timing, to carrier circuit packs. Each produces the following tones: call progress, touch tones, answer-back, and trunk transmission tests. They have 2 MHz, 160 kHz, and 8 kHz clocks and can transmit the system clock and tones on TDM bus A, TDM bus B, or both buses.

The TN768 is designed for the United States, while the TN780 is designed for Australia, Singapore, and the United Kingdom.

TN771 - Maintenance/Test

This circuit pack is the workhorse of Packet Bus maintenance. It can detect all Packet Bus failures for the Port Network in which it resides. The circuit pack provides a stand-alone mode (that is, one that does not involve communication with the SPE) for inspecting the Packet Bus for faults. This is a critical tool for the Packet Bus Fault Correction procedures.

TN777B - Network Control

The TN777B does the following:

- Communicates control channel messages between the processor circuit pack and the distributed network of port circuit packs on the TDM bus.
- Controls the four data channels that process and route information directly from the processor circuit pack to customer-connected equipment. The TN777B can be used to connect a system printer or CDR device to the system via the TN754B circuit pack.

- Has the time-of-day clock with a slow discharge capacitor for power failure or low-voltage conditions. The capacitor provides backup for approximately 24 hours and is automatically recharged whenever power is applied to the system. This circuit pack also has a 24-hour clock used with record keeping and system maintenance.
- Monitors the status of the system clocks and alerts the processor circuit pack in the event of a failure of any clock.
- Handles all the control channel messages over the TDM.
- Contains the AT&T MCU Memory Cartridge.

TN778 - Packet Control

This circuit pack provides the SPE interface to the Packet Bus, just as the TN777 Network Control circuit pack does to the TDM Bus. All traffic on the Packet Bus passes through the Packet Control.

The Packet Control can detect failures of certain control leads on the bus. Such failures are indicated by an inability to transmit data. The Packet Control can also detect data errors on the Packet Bus.

TN786B - Processor

The TN786B provides the means for storing and executing the software which operates the system features. The processor circuit pack consists of:

- A 80386SX Intel® processor
- 6 megabytes of Flash ROM
- 4 megabytes of DRAM

In addition, the processor performs maintenance functions such as:

- Monitoring the sanity of the system processor
- Reporting system processor failures
- Releasing or resetting the system processor on duplicated systems
- Monitoring and controlling cabinet level power supplies
- Managing the alarm panel LED indicators

TN787D - Multimedia Interface

The Multimedia Interface (MMI) circuit pack provides the H.221 protocol and BONDing terminations for data received from the trunk circuit packs. It demultiplexes incoming bit streams in the H.221 or BONDed signal (including audio, video, data, and control and indication signals) and routes the demultiplexed data to the TDM bus for processing in the appropriate circuit packs.

In addition, the MMI receives data bit streams from the various system circuit packs (such as conferenced audio, broadcast video, broadcast data, and control and indication), generates the H.221 framing signals, and multiplexes the data into a single H.221 bit stream for transmission to the original trunk circuit pack.

The MMI circuit pack supports all the relevant International Telecommunications Union-Telecommunications (ITU-T) H-series specifications: H.231, H.243, H.221, H.230, and H.242. It also supports all BONDed and non-BONDed calls. Each MMI circuit pack terminates as many as 32 B-channels in any combination as long as the aggregate bandwidth does not exceed 32 B-channels as in any of the following examples:

- 16 2-channel ports
- Five 384k (H0) ports
- Four 384k (H0) ports with four 2-channel ports
- 2-, 3-, or 4-channel BONDed ports
- 2-, 3-, or 4-channel multirate ports

The MMI circuit pack communicates with the processor complex using the standard control channel via the Network Control circuit pack.

TN788B - Voice Conditioner

Each Voice Conditioner circuit pack provides the circuits necessary to perform voice processing tasks for as many as four endpoints in a conference. Voice processing includes encoding, decoding, and summing audio signals. In addition, the voice conditioner circuit pack can provide the following:

- Mixed conference processing for any of the following:
  - G.711 PCM (A-law and 5-law) at 48K, 56K, or 64K
  - G.722 at 48K and 56K
  - G.711 in G.722 interworking mode at 48K and 56K
  - G.728 LD-CELP
- Voice energy detection information for the processor to use for voice energy activated switching

WP-91153 - AC Power Unit for ESCC/SCC

In a cabinet powered from an AC source, a single, plug-in, multi-output AC power supply is located in the power supply slot. A power cord with a three-prong plug on one end and a single connector on the other end connects the supply to a dedicated AC power source. The following figure shows the supply.

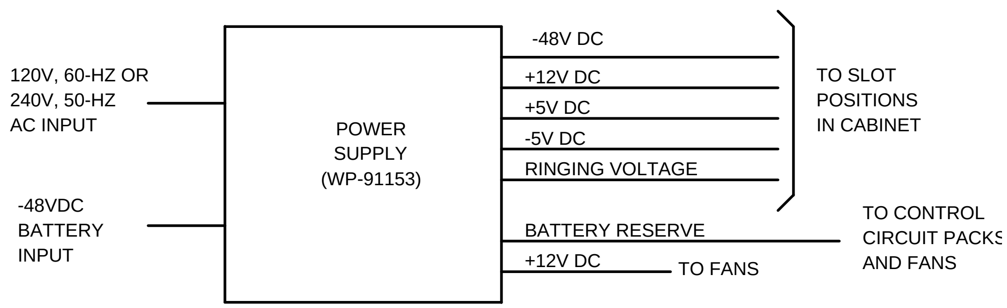


Figure 3-7. AC Power Supply (WP-91153) in Single-Carrier Cabinet

The inputs to the power supply can be (depending on list version):

- 120VAC, 60Hz, 15A to 20A; three wires in the power cord: one hot wire, one neutral wire, and one ground wire.

The AC power supply produces the following DC outputs: +5V, -5V, -48V, +12V, a ringing voltage, and a battery charging voltage. The DC outputs are distributed on the cabinet backplane to the slots for the circuit packs. The power supply has a circuit breaker and EMI filtering.

A 250-ms holdover circuit in the power supply allows a system to operate normally during AC power interruptions. When AC input power fails, reserve batteries supply power to the memory and processor circuit packs and fans for two minutes. All port circuit packs are inactive during this time.

The power supply contains a battery charger that charges the holdover batteries located in the bottom of the control cabinet.

ZAHFA - TDM/LAN Bus Terminator

The ZAHFA circuit pack provides system TDM/LAN bus termination.

AT&T MCU-MT

The AT&T MCU-MT is the primary system management vehicle for the AT&T MCU. It is an asynchronous terminal that provides access to the administrative and maintenance functions of the system software. The AT&T MCU-MT provides a means to administer system and conference options.

AT&T MCU-ST

The AT&T MCU-ST provides a remote interface that allows the same functionality as the AT&T MCU-MT.

Fans

System fans provide necessary cooling to prevent equipment damage. See Chapter 5, "Power and Fans" for details on system fans.

Remote Maintenance and Dial-Out Alarms

The AT&T MCU allows a dial-in interface that provides remote access to perform diagnostics or monitor conditions. In addition, the AT&T MCU can be administered to dial-out automatically and perform remote notification. These functions are performed via the initialization and administration system (INADS).

Cables

The following table lists the cables that can be used for the AT&T MCU and their respective COMCODEs.

Table 3-8. Cables and COMCODEs

Cable Description	COMCODE
M25A (9 ft) for data module	846823656
M25B (50 ft) for AT&T MCU-MT	846823730
Wall field connector cables:	
25 ft	H600307 G1
50 ft	H600307 G2
75 ft	H600307 G3
100 ft	H600307 G4
125 ft	H600307 G5
150 ft	H600307 G6
B25A Cable INADS + 754B:	
10 ft*	846300994
15 ft*	846301000
20 ft	846301018
25 ft	846301026
30 ft	846301034
35 ft	846301042
40 ft	846301059
45 ft	846301067
50 ft	846301075
55 ft	846301083
60 ft	846301091
65 ft	846301109
70 ft	846301117
75 ft	846301125
80 ft	846301133
85 ft	846301141
90 ft	846301158

Table 3-8. Cables and COMCODEs — *Continued*

Cable Description	COMCODE
95 ft	846301166
100 ft	846301174
125 ft	846301182
150 ft	846301190
175 ft	846301208
200 ft	846301216
DS1/MSM Cable	ED1E434-11 G506
DS1/MSM Y-cable D8W-87	103866109
356A Connector	104158829
AC system power cord	J58890H-1 L9
DC system power cord	J58890H-1 L10

* Typically, an ESCC/SCC system uses the 10 ft. B25A cable and an MCC system uses the 15 ft. B25A cable.

The required number of cables depends on the number of ports for which the AT&T MCU is configured. All systems require either one DC or AC power cable and two B25A cables. The number of required wall field connector cables is equal to the number of DS1 interface circuit packs.

Feature Descriptions

4

This chapter describes the primary AT&T MultiPoint Control Unit (MCU) features.

The features are arranged in alphabetical order, regardless of the functional area to which they apply. The information for each feature is presented under six headings: Description, Considerations, Interactions, Administration, and Hardware and Software Requirements.

- Description

  Defines the feature, tells what it does for the user or how it serves the system, and briefly describes how it is used.

- Considerations

  Discusses the applications and benefits of the feature, followed by the feature parameters and any other factors to be considered when the feature is used.

- Interactions

  Lists and briefly discusses other features that may significantly affect the feature being described. Interacting features are those that:

  — Depend on each other — one of the features must be provided if another particular feature is being used.

  — Cannot coexist — one of the features cannot be provided if the other feature is provided.

  — Affect each other — the normal operation of one feature modifies, or is modified by, the normal operation of the other feature.

  — Enhance each other — the features, in combination, provide improved service to the user.

- Administration

  States whether or not administration is required, how the feature is administered, who administers the feature, and lists items requiring administration.

- Hardware and Software Requirements

  Lists any additional hardware and/or software requirements needed for the feature.

NOTE:
Many of the AT&T MCU features and options are closely tied to administration and reports generation. This chapter is intended to provide an overview of the AT&T MCU features and options. For information concerning administration and reports, see the *AT&T MultiPoint Control Unit (MCU) System Administration and Reports*, 555-027-727.

Also see Table 4-1 of this document for feature availability by model.

Feature Summary

Table 4-1 provides a brief summary of the system features.

Table 4-1. System Feature Summary

Feature	Category
Audio Add-On (Integrated Audio Conferencing with Dynamic Echo Cancellation)	Multimedia Conferencing
Audio Modes	Multimedia Conferencing
Automatic Alternate Routing	Routing
Automatic Circuit Assurance	Identifying trunk problems
Automatic Route Selection	Routing
Bandwidth On Demand Interoperability Group (BONDing)	Routing
Basic/Enhanced Service Flag	Multimedia Conferencing
BRI/DCP Direct Connect Interface	BRI or DCP connection directly to the MCU
Business Card	Multimedia Conferencing
Call Detail Recording	Detailed call information
Call-by-Call (CBC) Service Selection	Trunk service selection on a call-by-call basis (ISDN service)
Cascading	Multimedia Conferencing
Class of Restriction (COR)	Assigning restrictions to facilities
Conference Redial Flag	Automatic retry on conference dial failure
Conference Reservation System	Multimedia Conferencing
DS1 Trunk Service	DS1 trunk interface
Dedicated Access	Allows non-signalling T1/E1 endpoints to join a conference
Dial-Out	Automatically calls all conference participants
Dial Plan	Enables trunk, station, and video port administration
Dynamic Resizing	Multimedia Conferencing
Facility Test Calls (with Security Measures)	Provides for test calls from the maintenance alarm terminal

Table 4-1. System Feature Summary — *Continued*

Feature	Category
Facility and Non-Facility Associated Signaling	ISDN-PRI signaling
High-/Low-Speed Interworking	Allows interworking of varying low-speed transfer rates
Integrated Services Digital Network (ISDN) - Primary Rate Interface (PRI)	Used for connecting the AT&T MCU to an ISDN-PRI trunk
Multipoint Communication Service (MCS)/Multilayer Protocol (MLP)	Multipoint data conferencing
Multirate Bandwidths	Larger bandwidths with ISDN interface
Networking/Call Handling	Connection of network calls to the appropriate video port
Notification Package	Includes the following: ■ Terminal names feature ■ Conference tones ■ Video-switching mode notification ■ Broadcaster notification ■ Time-left notification tone ■ Disconnect notification tone ■ Play tone command
Passwords	Provides additional conference security with per-conference or per-user passwords.
Rate Adaptation	Adapts for mixed 56k/64k conferences
Recent Change History	History report generation
Report Scheduler and System Printer	Reports scheduling
Security Violation Notification	Notification of unauthorized attempt at system management access
Selected Communications Mode (SCM) Upgrades	Configures conferences as endpoints join or leave the conference

Table 4-1. System Feature Summary — *Continued*

Feature	**Category**
Service Flags	Provides for identification of interoperability concerns
Status of WorldWorx Conference	Multimedia Conferencing
System Measurements	Traffic and efficiency report generation
System Status Report	Report generation
Terminal Names	Provides endpoint identification during a conference
Uniform Dial Plan (UDP)	Shared dial plan among a group of AT&T MCUs or PBXs
Video Conference Control Functions	Multimedia Conferencing
WorldWorx Data Compliance	Endpoint self-identification of WorldWorx data compliance
WorldWorx Service Indicator	Flags endpoints that are WorldWorx service endpoints

Audio Add-On (Integrated Audio Conferencing with Dynamic Echo Cancellation)

Description

Audio Add-On (Integrated Audio Conferencing with Dynamic Echo Cancellation) enables a reservations agent to add up to six audio-only endpoints to a conference. In this way, an endpoint that is not Px64 compliant can join the audio portion of a conference.

Considerations

The number of Audio Add-On endpoints is limited to six per conference.

Interactions

The following features interact with the Audio Add-On feature.

- Dynamic Resizing

  Dynamic resizing allows Audio Add-On only has an available audio port.

- Dial-Out

  In the interest of security, dial-out to an Audio Add-On endpoint is not allowed. The Audio Add-On endpoint must dial in to a MCU number.

Administration

Audio Add-On is administered per system by the system administrator. For the Audio Add-On capability, the number of Audio Add-On ports must be administered. Audio Add-On ports are administered in increments of two.

Hardware and Software Requirements

See Table 4-1 for feature availability by model.

Audio Modes

Description

For each conference, a preferred audio mode must be selected. A conference can be set up to require each endpoint to call the AT&T MCU (known as *dial-in* or *meet-me* setup), for the AT&T MCU to call each endpoint (known as *dial-out* setup) in turn, or for a combination of dial in and dial out. As each endpoint connects with the AT&T MCU for the conference, it automatically provides a capabilities list. The AT&T MCU, in turn, is administered on a per-conference basis for one of three preferred audio modes or for auto selection. In most cases, conferences are set for automatic selection of audio mode.

The possible audio modes are:

- G.722 without data sharing
- G.728 (LD-CELP) with data sharing
- G.711 (PCM) without data sharing

In auto selection, the AT&T MCU refers to the capabilities list of each endpoint in a highest common denominator (HCD) algorithm to determine the audio mode to use. First, the AT&T MCU attempts to use the audio mode requiring the smallest bandwidth G.722 or G.728 (LD-CELP if the conference is also administered for data sharing) to provide greater video quality. If a connecting endpoint is not capable of G.722 or G.728 LD-CELP mode, G.711 pulse-code modulation (PCM) is selected and data sharing is disabled.

NOTE:
G.728 is the required audio mode for conferences using the MCS/MLP features. G.728 is the preferred mode in all 2-channel conferences. G.722 is the audio mode that allows the best video quality and is preferred in all other conference bandwidths.

If the endpoint capabilities list does not contain an audio mode common to the endpoints already connected and it is a single-channel endpoint attempting to connect to a 2-channel conference, the connecting endpoint is allowed an audio-only PCM connection to the conference. Except in the case of High-Speed/Low-Speed Interworking, any other mismatch (such as, a 112k endpoint attempting to connect to a 384K (H0) conference) the endpoint is not allowed to join.

In conferences involving endpoints at higher rates than 64k, endpoints that cannot join with the administered bandwidth are connected as audio-only PCM endpoints.

Considerations

Only the G.728 LD-CELP audio mode allows data sharing.

Interactions

The following features interact with the audio mode selection:

- High-Speed/Low-Speed Interworking.

High-Speed/Low-Speed Interworking allows mismatched bandwidths to join conferences as audio-only endpoints.

- Multipoint Communication Service/Multilayer Protocol.

  See the MCS/MLP feature description for details.

Administration

Audio modes can be administered via the Conference Record form.

Hardware and Software Requirements

See Table 4-1 for feature availability by model.

Automatic Alternate Routing

Description

Automatic Alternate Routing (AAR) provides alternative routing choices for private on-network calls. With AAR, the system automatically selects the most desirable (normally the least expensive) route over various trunking facilities for private network calls. AAR also provides digit modification to allow on-network calls to route through the public network when an on-network route is not available, and to convert incoming digits to MCU extensions in the AT&T MCU.

The private network of PBXs that utilizes the AAR feature is called an Electronic Tandem Network (ETN). An ETN is a hierarchical network of privately owned trunk and switching facilities that can provide a cost-effective alternative to toll calling between locations. An ETN consists of tandem switches, the inter-tandem tie trunks that interconnect them, the access or bypass tie trunks from a tandem switch to a main switch, and the capability to control call routing over these facilities.

Within an ETN, each switching facility is identified by a unique private network office code. Private network office codes may be one to eight digits in length. Throughout the rest of this description, the private network office code will simply be referred to as the “office code.”

ETN addresses for Uniform Dial Plan (UDP) destinations are limited to a seven-digit format. This means that the location code part for UDP is a three-digit code of the form RNX and the extension number is a four-digit number in the XXXX format (along with limitations that the UDP number cannot start with a 0). Note that five-digit UDP extensions are supported. For other destinations, ETN addresses are not limited to the seven-digit RNX format.

The principal use of AAR is to provide routing of private network calls. Private network calls originate and terminate at a customer location without accessing the public network. The normal scenario is as follows: the dial-out feature is used to dial the AAR access code followed by an on-network number. AAR then selects the route for the call and performs any necessary digit manipulation. AAR selects the most desirable route for the call. If the first choice route is not available, another route is chosen automatically. AAR provides up to six routes for each office code.

Feature operation is completely transparent to the user. The AAR access code is normally the digit 8. Normally, the called number is a private network number.

Private network (on-network) numbers are handled by the AAR feature. An on-network number can be changed into a public network direct-distance dialing number, a CDOS number, or an IDDD number by administering the “ars” call-type for such numbers.

The private network location codes may match public network central office codes. Therefore, the only way to determine the intended network for seven-digit calls is by the AAR or ARS access codes administered in the dial-out number(s) or by specific administration of the “ars” call-type on the AAR Analysis form.

AAR and Subnet Trunking provide a convenient means to place IDDD calls to a frequently called foreign city. Such calls route as far as possible over the private network before exiting the network. The office code is, of course, reserved to represent a particular country and city. At the final on-network switch, the office code is deleted. The international prefix code (011 in the US, 00 in most of Europe, and so on), the country code, and the city code are inserted. The inserted digits plus the last four digits of the originally dialed number constitute the IDDD number.

Similar to the IDDD case, certain domestic calls may reach a point on the network where they can route no further because tie trunks to the next switch are busy or none are provided. In this case, the office code can be deleted and the appropriate public network code inserted. Calls of this type route off-network via a central office. The central office may be connected to either an ETN tandem or main switch. Toll charges, if any, are from the final ETN switch to the destination.

Each office code can point to any one of several Routing Patterns, numbered 1 through a maximum limit of 254 for your system. More than one office code can point to the same pattern. A blank pattern provides intercept treatment and pattern 254 is the default for all office codes. Routing Patterns are shared with ARS. Access to a route within the pattern is controlled by Facilities Restriction Level (FRL) assignments. FRLs are fully described elsewhere in this chapter. For outgoing ISDN calls, route selection is dependent on Bearer Capability Class (BCC), FRL, and type of facility.

Digit Conversion

Once the AAR access code and the called number are dialed, the dialed number is compared to entries in the Matching Pattern fields of the AAR Digit Conversion Table screen. An example of this is shown in Figure 4-1. If all or part of the dialed number matches one of the Matching Patterns on the screen, the dialed number is replaced by a new number from the Replacement String field on the screen. This new number is then used to route the call, the call becomes an ARS call, and is routed using the ARS Analysis Table. This function may be used to route specific dialed number strings to a different number, intercept, and so on. The Digit Conversion Table is only used once per call.

```
                        AAR Digit Conversion Table
                                                        Percent Full:

 Matching Pattern Replacement String  Matching Pattern Replacement String
---------------- ----------------   ---------------- ----------------
---------------- ----------------   ---------------- ----------------
---------------- ----------------   ---------------- ----------------
---------------- ----------------   ---------------- ----------------
---------------- ----------------   ---------------- ----------------
---------------- ----------------   ---------------- ----------------
---------------- ----------------   ---------------- ----------------
---------------- ----------------   ---------------- ----------------
---------------- ----------------   ---------------- ----------------
---------------- ----------------   ---------------- ----------------
---------------- ----------------   ---------------- ----------------
---------------- ----------------   ---------------- ----------------
---------------- ----------------   ---------------- ----------------
---------------- ----------------   ---------------- ----------------
---------------- ----------------   ---------------- ----------------
---------------- ----------------   ---------------- ----------------
```

Figure 4-1. AAR Digit Conversion Table

The purpose of AAR digit conversion is simply to convert private network numbers to public network numbers. This allows the system to change specific dialed digits to a public network number and route some calls via ARS. Also, unauthorized private network calls can be routed to receive intercept treatment.

AAR Analysis

After an AAR call passes through AAR Digit Conversion, AAR Analysis is performed.

The system uses AAR Analysis to compare the dialed number with entries in an AAR Analysis Table. When the system finds a dialed string entry in the table that matches the dialed number, the AAR Analysis Table maps the dialed number to a specific Routing Pattern (discussed later) and Call Type. The selected Routing Pattern and Call Type are then used to route the call. If the Call Type field on the AAR Analysis Table for a digit string is “ars,” call processing crosses over to ARS and the call is processed as an ARS call. The AAR Analysis Table screen form also shows the minimum and maximum number of digits required for digit analysis of each dialed number. An example of the AAR Analysis Table screen form is shown in following screen.

Dialed string entries may contain the letter ‘‘x’’ or ‘‘X,’’ which is used as a ‘‘wildcard’’ character. This wildcard character matches any of the digits 0 through 9. For example, a dialed string entry of ''3x'' applies to all calls beginning with 30 through 39. This ''wildcard'' makes it possible for traditional three-digit RNXs to be represented in several ways in the AAR Analysis Table. For example, RNXs 200 through 299 can be assigned to the AAR Analysis Table in either of the following ways:

Dialed String	**Minium Number of Digits**	**Maximum Number of Digits**
2	7	7
	or	
20	7	7
21	7	7
22	7	7
...	...	...
29	7	7
	or	
2xx	7	7
	or	
20x	7	7
21x	7	7
22x	7	7
...	...	...
29x	7	7

It is possible that some numbers may overlap other numbers. For example, the AAR Analysis Table may have dialed string entries of ''645'' and ''6452.'' In this case, for example, the number 645-2045 will be routed according to the 6452 entry (the longest dialed string).

When the UDP is used, the three-digit RNX dial string representation must be used on the AAR Analysis Table to match the administration in the UDP table.

Possible Call Types in the AAR Analysis Table are as follows:

- **aar** — Regular AAR call
- **ars** — Crossover to ARS call
- **haar** — Home ETN address (indicates that the call should be terminated locally on the home switch instead of routing to another ETN node. If the UDP is administered, then that dial plan is used to convert the number to a local extension number. Otherwise, the location code is deleted from the dialed number, and the remaining digits are used to route the call to a local extension).

Figure 4-2 shows the AAR Analysis Table.

```
                          AAR Analysis Table
                     Partitioned Group Number:  1       Percent Full:___

     Dialed                 Route Call        Dialed                 Route Call
     String        Min Max  Pat   Type        String        Min Max  Pat   Type
_________________  __  __   ___   ____  _________________  __  __   ___   ____
_________________  __  __   ___   ____  _________________  __  __   ___   ____
_________________  __  __   ___   ____  _________________  __  __   ___   ____
_________________  __  __   ___   ____  _________________  __  __   ___   ____
_________________  __  __   ___   ____  _________________  __  __   ___   ____
_________________  __  __   ___   ____  _________________  __  __   ___   ____
_________________  __  __   ___   ____  _________________  __  __   ___   ____
_________________  __  __   ___   ____  _________________  __  __   ___   ____
_________________  __  __   ___   ____  _________________  __  __   ___   ____
_________________  __  __   ___   ____  _________________  __  __   ___   ____
_________________  __  __   ___   ____  _________________  __  __   ___   ____
_________________  __  __   ___   ____  _________________  __  __   ___   ____
_________________  __  __   ___   ____  _________________  __  __   ___   ____
_________________  __  __   ___   ____  _________________  __  __   ___   ____
_________________  __  __   ___   ____  _________________  __  __   ___   ____
```

Figure 4-2. AAR Analysis Table

Some special dialing patterns are automatically mapped to a specific Routing Pattern and Call Type. These special dialing patterns, shown in Table 4-2, are preassigned and require no administration. However, they can be changed by the system administrator.

Table 4-2. AAR Analysis Default Translations

Dialed String	Number of Digits		Route Pattern	Call Type
	Minimum	Maximum		
0	1	23	—	ars
1	4	23	—	ars
2	7	7	254	aar
3	7	7	254	aar
4	7	7	254	aar
5	7	7	254	aar
6	7	7	254	aar
7	7	7	254	aar
8	7	7	254	aar
9	7	7	254	aar
X11	3	3	—	ars

Normally, the **Route Pat** (routing pattern) field on the Analysis Table screen contains a routing pattern number (1 through 254). However, this field may instead contain a Remote Home Numbering Plan Area (RHNPA) table number (r1 through r32). When an AAR Analysis table points to an RHNPA table, the next three dialed digits are compared with codes in the selected RHNPA table. Each code on the table is then mapped to a specific routing pattern number (1 through 254).

Routing Patterns

The digit translations performed on an AAR call by the AAR Analysis and RHNPA tables cause a specific Routing Pattern to be selected for the call. The Routing Patterns are numbered 1 through 254. More than one combination of dialed digits can point to the same pattern. A blank entry instead of a Routing Pattern number provides intercept treatment. However, with AAR, digit translation should always point to a Routing Pattern. If calls to some numbers are to be denied, this should be handled by FRL assignment, not by intercept on the codes. FRLs are discussed elsewhere in this chapter.

The Routing Pattern applicable for a given call contains a list of up to six trunk groups that can be used for the call. Trunk group access is controlled by FRLs. The digit manipulation necessary to route the call is controlled by the Subnet Trunking feature. Otherwise, the digit string to be outpulsed is as dialed by the user or as converted by AAR Digit Conversion.

Considerations

AAR provides efficient use of private network facilities.

ARS and AAR Analysis tables together can have up to 2,000 entries.

If a customer changes AAR routing assignments, it is the customer's responsibility to notify the RSC network designer and the SCO technician of the changes to receive their continued support.

Digit deletion/insertion can be used to route calls to the AT&T MCU.

Internal memory resources used for AAR Digit Analysis are shared by ARS, AAR, Digit Conversion, and Toll Analysis features. A Percent Full field on the ARS and AAR Digit Analysis screens indicates how many of these resources have been used.

Interactions

The following features interact with the AAR Feature.

- ARS

  ARS and AAR can access the same trunk groups and share the same Routing Patterns and RHNPAs. Also, AAR calls can be administered to cross over to ARS via digit analysis and digit conversion.

- CDR

  An AAR call using a trunk group marked for CDR is indicated by the dialed access code and by a Condition Code. The dialed number is recorded as the called number.

  Subnet Trunking does not affect CDR. The dialed digits are recorded, not the outpulsed digits.

  The originating FRL associated with the call is recorded. However, if 15-digit CDR account codes are used, the FRL value is overwritten.

  If CDR generation is administered for a trunk group assigned to a Routing Pattern, data will be collected for all calls routed through the trunk group. If an CDR account code is to be dialed with an AAR call, it must be dialed before the AAR access code is dialed.

- UDP

  When a UDP number is dialed (four or five digits), the routing software initially converts the dialed number to a seven-digit format. The location code in the UDP table equates to a three-digit RNX plus four digits. In a four-digit UDP, the number created is the RNX plus the four extension digits originally dialed. In a five-digit UDP, the number created is the RNX plus the last four extension digits dialed.

UDP destinations are limited to a seven-digit format. This means that the location code part for UDP is a three-digit code of the form RNX and the extension number is a four-digit number in the XXXX format (along with limitations that the UDP number cannot start with a zero).

When the UDP is used, the three-digit RNX dial string representation must be used on the AAR Analysis Table to match the administration in the UDP table.

Administration

AAR is initially assigned on a per-system basis by an AT&T service technician. After the feature is activated, the following items are administered by either the system administrator or the service technician:

- AAR Access Code (one to three digits)
- AAR Analysis Table
- AAR digit conversion table Up to 32 RHNPA Tables
- Up to 254 Routing Patterns
- FRLs — Assigned via Class of Restriction to each originating facility, authorization codes, and barrier codes
- Trunk Groups to be used with AAR

Hardware and Software Requirements

AAR may require additional tie trunks. These additions are, however, cost effective when compared to the other alternatives for call routing.

Automatic Circuit Assurance

Description

Automatic Circuit Assurance (ACA) assists users in identifying possible trunk malfunctions. The system maintains a record of the performance of individual trunks relative to short and long holding time calls. The system automatically initiates a referral call to a maintenance alarm terminal when a possible failure is detected.

Holding time is the elapsed time from the time a trunk is accessed to the time a trunk is released. When the ACA feature is enabled by the system administrator, the system measures the holding time of each call.

A short holding time limit and a long holding time limit are preset by the system administrator for each trunk group. The short holding time limit can be from 0 to 160 seconds. The long holding time limit can be from 0 to 24 hours. The measured holding time for each call is compared to the preset limits for the trunk group being used.

A short holding time counter and a long holding time counter associated with each trunk group member are kept by the system. When the measured holding time of a call is compared to the preset limits, these counters are incremented or decremented as follows:

- Measured holding time less than short holding time limit — short holding time counter is incremented.
- Measured holding time greater than short holding time limit and less than long holding time limit — short holding time counter is decremented.
- Measured holding time greater than long holding time limit — long holding time counter is incremented.

The short holding time counter is constantly compared to a preset threshold. This threshold can be from 0 to 30 seconds and is set by the system administrator. The threshold for the long holding time counter defaults to 1 second. Each time a counter reaches a preset threshold, two things occur as soon as the system clock reaches the next hour or the call is dropped:

1. If ACA referral has been activated, the system sends a referral call to the maintenance alarm terminal.
2. An entry is made in an audit trail, which stores information on the occurrence.

When ACA is enabled by the system administrator, the ACA measurements are made and the audit trail is updated each time a preset counter threshold is reached. However, for a referral call to be sent, the ACA referral feature must be activated by the maintenance alarm terminal user pressing the ACA button. When this is done, the system can send referral calls to the maintenance alarm terminal.

The information appearing on the display identifies the call as an ACA call, identifies the trunk group access code and the trunk group member number, and shows the reason for referral (short or long holding time). When the call is answered, this information is displayed and remains displayed until the call is released.

Each time a counter threshold is reached, a record of the information is stored in the audit trail. The audit trail records are available to the system administrator. Each record contains the following information:

- Time and Date of occurrence
- Trunk group number, trunk access code, and trunk group member
- Type of referral (short or long holding time)

If the referral call destination does not answer the call within three minutes, the call times out and this information is entered in the audit trail. The audit trail is examined once each hour. If any entries indicate a referral call was not completed, the system tries the call again.

ACA can be enabled or disabled for the entire system by the system administrator. The system administrator can also enable or disable ACA for each individual trunk group. When ACA is disabled, ACA measurements are not made.

Two extensions must be assigned for the purpose of letting the referral call destination identify the type of ACA call (short or long holding time). The two extensions are assigned as a short holding time origination extension and a long holding time origination extension. These extension numbers do not require hardware circuit packs.

As an illustration of how ACA functions, assume the following:

- The ACA is enabled for the entire system.
- The ACA referral destination is extension 389.
- The ACA long holding time origination extension is 423.
- The long holding time limit for trunk group 3 (trunk access code is 9) is one hour.
- The threshold should be 1.
- The ACA referral is activated.

With the above information, assume that a call is made on a trunk in trunk group 3 and the call lasts more than one hour. Then, the threshold for the long holding time counter is reached, a referral call is made to extension 389, the display reflects a long holding time call, and the information is entered in the audit trail.

Considerations

The ACA feature provides better service through early detection of faulty trunks and consequently reduces out-of-service time. Some types of trunk failures cause people to shorten their calls. For example, an excessive number of short calls may indicate a noisy trunk. Similarly, a trunk that remains busy for an abnormally long time may be permanently busy due to a trunk fault. The ACA feature takes advantage of these characteristics to identify possibly defective trunks.

The audit trail contains a maximum of 64 records at any one time. The oldest information is overwritten by the newest information.

Measurements are not made on out-of-service trunks or trunks undergoing maintenance testing.

Interactions

None.

Administration

ACA is administered by the system administrator. The following items require administration:

- Whether ACA is enabled or disabled (per system)
- Short holding time origination extension (per system). Assigned name must reflect short holding time nomenclature.
- Long holding time origination extension (per system). Assigned name must reflect long holding time nomenclature.
- Referral destination (per system)
- Whether ACA is assigned (per trunk group)
- Short holding time limit (per trunk group)
- Long holding time limit (per trunk group)
- Threshold for short holding time counter (per trunk group)
- ACA activate/deactivate button on a maintenance alarm terminal

NOTE:
It is recommended that the short holding time be administered as a minimum of 45 seconds and the long holding time at or above the average conference time.

Administer these items on the ‘‘System Parameter Form,’’ ‘‘Trunk Group Form.’’

Hardware and Software Requirements

No additional hardware or software is required.

Automatic Route Selection

Description

The Automatic Route Selection (ARS) feature routes calls over the public network based on the preferred (normally the least expensive) route available at the time the call is placed.

ARS provides a choice of up to six routes for any given public network call. The following types of trunk groups can be accessed by ARS:

- **Tie trunks** — Used to provide access to an ETN or public network. In some cases, it is preferable to allow a private network to handle the routing of calls destined for the public network.
- **ISDN-PRI** — Used for calls over an ISDN and provides users access to a variety of switched nodal services such as the ACCUNET. digital service and allows access to other inter-exchange carriers or private networks.

A variety of numbers can be called using ARS, including seven-digit numbers, 10-digit numbers, International Direct Distance Dialing (IDDD) numbers, service codes, Customer-Dialed Operator-Serviced (CDOS) numbers (for example, in the United States, 0+ or 01+), and Inter-Exchange Carrier (IXC) numbers.

ARS Dialing

The dial-out feature uses ARS for outgoing calls. ARS is initiated if the ARS access code (normally the digit 9) is included in the first digit of the dial-out number(s) administered by the reservations agent.

Special Dialing Patterns

The system recognizes certain dialing patterns on outgoing calls and routes these calls accordingly. The descriptions of these dialing patterns reflect the system defaults as used in the United States. Other countries may require different administration of these values in the dial-out numbers. The following dialing patterns are recognized:

- **DDD Calls With Prefix Digit 1 Required**

  The dial-out number may or may not be required to contain a 1 before a seven- or 10-digit dial out string, depending on the system’s dial plan administration. There are two cases where the digit 1 must be included:

  — Some metropolitan areas are so densely populated that there simply are not enough traditional central office codes. Therefore, it is possible that some NPA codes, also called ‘‘area codes’’ may also serve as CO codes. In this case, the digit 1 must be included if a 10-digit call is intended. The first digit tells the system whether to route the call as a seven-digit call within the home NPA (1 not included) or as a 10-digit call to another NPA (1 included). In this case, the dial plan should be administered so that the 1 is included for 10-digit calls.

— Digit 1 may also be required in areas near an NPA boundary. In these areas, certain calls to the adjacent NPA may be local calls rather than toll calls. However, central office codes may be duplicated in the home and adjacent NPAs. Also a CO code in the home NPA may be a toll call. Therefore, if the digit 1 is not required on certain adjacent NPA local calls, then it must be included on the home NPA seven-digit toll calls so the system can differentiate between the intended destinations.

- **DDD Calls with Prefix Digit 1 Not Required**

  The first digit following the ARS access code may or may not be a 1. In systems where the 1 prefix is used, but not required (as administered on the Dial Plan form), using the 1 prefix before a 10-digit call is optional and the prefix is ignored.

- **IDDD Calls**

  IDDD numbers consist of a Country Code and a National Number. The National Number is simply the number used when calling within the country. The Country Code can be from one to three digits in length. In the NANP the National Number is 10 digits in length. The Country Code and National Number together cannot exceed 12 digits. In the NANP, international numbers are recognized by special prefix codes:

  — **011** — Indicates that dial out is making a station paid direct international call. The Country Code and National Number follow the 011 prefix.

- **Calls Dialed with Inter-Exchange Carrier (IXC) Access**

  The first digits following the ARS access code are an IXC Access Code. The access code may be followed by a DDD or an IDDD number. If included in the dial-out number(s), it selects the carrier or facilities used for routing the call.

Digit Conversion

Once dial out uses the ARS access code and the called number, the dialed number is compared to entries in the Matching Pattern fields of the ARS Digit Conversion Table screen. An example of this screen is shown in Figure 4-3. If all or part of the dialed number matches one of the Matching Patterns on the form, the matching part of the dialed number is replaced by a new number from the **Replace** field on the form. This new number routes the call via AAR over a private network. If no corresponding entry is found in the AAR Digit Analysis table, the ARS Digit Analysis table is searched for a match with the new modified number and routed accordingly. If the call fails and a match is not found, an intercept tone is sounded.

```
                        ARS Digit Conversion Table
                                                   Percent Full:

 Matching Pattern Replacement String  Matching Pattern Replacement String
---------------- ----------------   ---------------- ----------------
---------------- ----------------   ---------------- ----------------
---------------- ----------------   ---------------- ----------------
---------------- ----------------   ---------------- ----------------
---------------- ----------------   ---------------- ----------------
---------------- ----------------   ---------------- ----------------
---------------- ----------------   ---------------- ----------------
---------------- ----------------   ---------------- ----------------
---------------- ----------------   ---------------- ----------------
---------------- ----------------   ---------------- ----------------
---------------- ----------------   ---------------- ----------------
---------------- ----------------   ---------------- ----------------
---------------- ----------------   ---------------- ----------------
---------------- ----------------   ---------------- ----------------
---------------- ----------------   ---------------- ----------------
---------------- ----------------   ---------------- ----------------
```

Figure 4-3. ARS Digit Conversion Table

Digit analysis implies both route and conversion. The algorithm for deciding how to analyze a call is as follows:

- number of digits
- most digits matched
- explicit digits over wildcards (from left to right)
- conversion nodes over route nodes

The primary purpose of ARS digit conversion is to convert public network numbers (to be routed with ARS) to private network numbers (to be routed with AAR).

NOTE:
You can convert back to another AAR number or an extension, where the extension most likely is a UDP.

This can save toll charges and allow the reservations agent to route calls on the customer's private network facilities. Also, unauthorized public network calls can be blocked and routed to a receive intercept treatment via digit conversion.

Table 4-3 shows several ARS Digit Conversion examples.

The following conditions are assumed for the examples: ARS Access Code = 9, AAR Access Code = 8, Home RNX (Private Network Office Code) = 222, Prefix 1 is required on all long-distance DDD calls, Dashes ‘‘-’’ shown in the table are for readability only.

Table 4-3. ARS Digit Conversion Examples

Operation	Actual Digits Dialed	Matching Pattern	Replace-ment String	Modified Address	Notes
DDD call to ETN	9-1-303-538-1345	1-303-538	362	362-1345	The call will be routed via AAR on the route selected for RNX 362.
Unauthorized call to intercept treatment after 976 are ignored by ARS. The call receives intercept treatment.	9-1-212-976-1616	1-XXX-976	#	(blank)	The “#” signifies the end of dialing. Any digits dialed after 976 are ignored by ARS. The call receives intercept treatment.

NOTE:
The dial-out number digits are matched to the Matching Pattern that most closely matches them. For example, if the dial-out number is 957-1234 and matching patterns 957-1 and 957-123 are in the table, the match is on pattern 957-123. The call will be routed as dialed.

ARS Digit Analysis

ARS calls that pass through ARS Digit Conversion and Toll Analysis are analyzed based on the dial-out PGN.

The system uses ARS Digit Analysis to compare the dialed number with entries in an ARS Digit Analysis Table. When the system finds a Dialed String entry in the table that matches the dialed number, the ARS Digit Analysis Table maps the dialed number to a specific Routing Pattern, discussed later in this chapter, and Call Type. The selected Routing Pattern will then be used to route the call. The ARS Digit Analysis Table screen also shows the minimum and maximum number of trailing digits required for digit analysis of each dialed number. An example of the ARS Digit Analysis Table follows.

Possible Call Types in the ARS Digit Analysis Table are as follows:

- **fnpa** — 10-digit call within North America
- **hnpa** — 7-digit call within North America
- **int** — International call (in the United States, the international prefix is 011)
- **natl** — National numbers within a country (used outside North America)

```
                              ARS Analysis Table
                         Partitioned Group Number:  1       Percent Full:___

     Dialed                  Route Call         Dialed                  Route Call
     String       Min Max  Pat  Type         String       Min Max  Pat  Type
_________________ __  __  ___  ____   _________________ __  __  ___  ____
_________________ __  __  ___  ____   _________________ __  __  ___  ____
_________________ __  __  ___  ____   _________________ __  __  ___  ____
_________________ __  __  ___  ____   _________________ __  __  ___  ____
_________________ __  __  ___  ____   _________________ __  __  ___  ____
_________________ __  __  ___  ____   _________________ __  __  ___  ____
_________________ __  __  ___  ____   _________________ __  __  ___  ____
_________________ __  __  ___  ____   _________________ __  __  ___  ____
_________________ __  __  ___  ____   _________________ __  __  ___  ____
_________________ __  __  ___  ____   _________________ __  __  ___  ____
_________________ __  __  ___  ____   _________________ __  __  ___  ____
_________________ __  __  ___  ____   _________________ __  __  ___  ____
_________________ __  __  ___  ____   _________________ __  __  ___  ____
_________________ __  __  ___  ____   _________________ __  __  ___  ____
_________________ __  __  ___  ____   _________________ __  __  ___  ____
```

Figure 4-4. ARS Digit Analysis Table

Some special dialing patterns are automatically mapped to a specific Routing Pattern and Call Type. These special dialing patterns, shown in Table 4-4 are preassigned and require no administration. They can, however, be changed by the system administrator and probably will be changed for PBXs used outside North America.

Table 4-4. ARS Digit Analysis Default Translations

Dialed String	Trailing Digits Minimum	Trailing Digits Maximum	Pattern	Route Type
011	10	23		int
1	11	11		fnpa
10XXX011	15	23		int
2	7	7	1	hnpa
3	7	7	1	hnpa
4	7	7	1	hnpa
5	7	7	1	hnpa
6	7	7	1	hnpa
7	7	7	1	hnpa
8	7	7	1	hnpa
9	7	7	1	hnpa

Legend:
fnpa — foreign number plan area (10-digit call)
hnpa — home number plan area (7-digit call)
int — international
X — any digit (0-9)

Normally, the Route Pat (routing pattern) field on the ARS Digit Analysis Table screen contains a routing pattern number of 1 through 254. However, this field may instead contain a Remote Home Numbering Plan Area (RHNPA) table number (r1 through r32). An RHNPA is simply a concentrator for up to 1,000 calls. Calls are routed to these tables by ARS/AAR Digit Analysis when an ARS Digit Analysis table points to an RHNPA table. The next three dialed digits (the code) are compared with the codes in the selected RHNPA table. Each code on the table is then mapped to a specific routing pattern number (one through 254).

The RHNPA tables allow up to 1000 codes to be handled by one entry in the Digit Analysis table.

In summary, Digit Analysis is merely a method of selecting a routing pattern. The routing pattern may be selected in two ways:

- It may be selected directly from the Digit Analysis table.

- The Digit Analysis table may first have to select an RHNPA table that will, in turn, select the routing pattern.

Routing Patterns

The digit translations performed on an ARS call by the Digit Analysis and RHNPA tables cause a specific Routing Pattern to be selected for the call. The Routing Patterns are numbered 1 through 254. More than one combination of dialed digits can point to the same pattern. A blank entry instead of a Routing Pattern number provides intercept treatment. However, with ARS, digit translation should always point to a Routing Pattern. This way, calls to unassigned office codes will be intercepted by the central office, not by the system. By allowing the unassigned codes to be intercepted by the central office, the system administrator does not have to keep track of which office codes are in service. If calls to some codes are to be denied, this should be handled by FRL assignment, not by intercept on the codes. FRLs are discussed elsewhere in this chapter.

The Routing Pattern applicable for a given call contains a list of up to six trunk groups that can be used for the call. Trunk group access is controlled by FRLs. If access to the public network is through a main switch (an Access trunk group is selected for the call), then the call will route through the main switch to one of the public network offices serving the main switch. The digit manipulation necessary to route the call is controlled by the Subnet Trunking feature. Otherwise, the digit string to be outpulsed is controlled by ARS. ARS digit manipulation is called code conversion. Code conversion is used to determine whether or not to outpulse the digit 1 on toll calls and whether to insert, keep, or delete the NPA on toll calls.

The following paragraphs describe how the switch decides what digits to outpulse in specific situations.

Digit 1 Outpulsing

The digit 1 may or may not be required at the public network office to which the call will be routing. If 1 is dialed on 7-digit calls at a stand-alone system (non-ETN), the 1 is outpulsed by the system. In the other cases, the 1 outpulsing requirements are indicated in the system. Since any given call may have a choice of up to six routes, some of which may require a 1 and some of which may not, this indication is associated with each route. Five choices are available and are identified in translations by a Prefix Mark. Digit 1 outpulsing only applies to calls administered as ‘‘fnpa’’ or ‘‘hnpa’’ in the ARS Digit Analysis table. The values and meanings of the Prefix Marks are as follows:

- Prefix Mark 0 — Suppress a dialed Prefix digit 1 for 10-digit FNPA calls, but leave a Prefix digit 1 for the following types of calls:
  - 10-digit calls that are not administered as FNPA or HNPA types in the ARS Routing Table.
  - 7-digit HNPA calls
- Prefix Mark 1 — Send a 1 on 10-digit calls, but not on 7-digit calls.

- Prefix Mark 2 — Send a 1 on all toll calls (for example, all 10-digit calls and 7-digit toll calls).
- Prefix Mark 3 — Send a 1 on all toll calls and keep or insert the NPA to ensure that all toll calls are 10-digit calls. Note that a dialed Prefix digit 1 for a 7-digit call makes it a toll call and, hence, NPA is also inserted in this case.
- Prefix Mark 4 — Always suppress a Prefix digit 1.

NOTE:
This capability is required, for example, when routing ISDN-PRI calls to an AT&T 4ESS. If the prefix digit 1 were not suppressed, then the 4ESS would reach calls.

Which of the five possible treatments of the 1 prefix digit should be administered on a given route is based on the characteristics of the distant office. Prefix Mark 0 prevents the system from sending a 1 prefix digit for 10-digit FNPA calls. However, the system leaves a user-dialed prefix digit 1 for 7-digit HNPA calls and 10-digit calls that are not administered as FNPA or HNPA types in the ARS Routing Table.

Prefix Mark 1 causes the system to send a 1 prefix on all 10-digit FNPA calls.

With Prefix Marks 2 and 3, the decision is based on whether the call is a toll call. Toll Lists are provided in the system to furnish this information. A Toll List simply indicates if the office code associated with the call constitutes a toll call from the interconnecting office (not from the local system). Up to 32 Toll Lists are provided. The applicable list number, if any, for the call is given in the Routing Pattern.

Prefix Marks are only applicable on 7- or 10-digit DDD public network calls. Requirements for outpulsing a 1 are specified via Prefix Marks and go into effect when the call accesses is outpulsed. Digit 1 outpulsing only applies to calls administered as ‘‘fnpa’’ or ‘‘hnpa’’ in the ARS Digit Analysis table.

NPA Deletion and Insertion

Each public network route in the ARS Routing Pattern contains an indication of the NPA of the distant end of the trunk group. If this NPA is the same as the NPA associated with the call, the NPA is deleted prior to outpulsing unless the Prefix Mark is 3 and the call is a toll call in the associated Toll List.

NPA deletion and insertion only applies to calls administered as ‘‘fnpa’’ or ‘‘hnpa’’ in the ARS Digit Analysis table.

The NPA is inserted on 7-digit calls if the distant NPA is different from the home NPA or if the Prefix Mark is 3 and the call is a toll call in the associated Toll List.

The preceding paragraphs describe NPA deletion or insertion when the call is an ARS 7- or 10-digit DDD call. An ARS call accessing a tandem trunk is another example of NPA insertion. If the call is a 7-digit ARS call, the system inserts the home NPA before sending the call to the tandem trunk. Therefore, all ARS calls accessing a tandem trunk are 10-digit calls. Whether or not the digit 1 is sent on a tandem call is determined by the prefix rules. This enables the system to distinguish between ARS calls and the 7-digit on-network calls.

IDDD and Service Code Dialing

ARS can provide individual Routing Patterns for each type of call. An ARS call can be processed via the RHNPA table. This is particularly useful on international calls, since the RHNPA table can be used on the country code. Thus, call routing can be determined according to the called country, rather than handling all international calls alike.

Considerations

ARS provides the most-preferred usage of public network facilities available at a system.

Up to 254 Routing Patterns, shared with AAR, can be provided.

Up to 32 Toll Lists can be provided.

Up to 32 RHNPA tables, shared with AAR, can be provided.

ARS and AAR Digit Analysis tables together with the Toll Analysis table can have up to 2,000 entries.

Internal memory resources used for ARS Digit Analysis are shared by ARS, AAR, Digit Conversion, and Toll Analysis features. A Percent Full field on the ARS and AAR Digit Analysis screens indicate how many of these resources have been used.

If a customer changes ARS routing assignments, it is the customer’s responsibility to notify the RSC network designer and the SCO technician of the changes to receive their continued support.

Interactions

The following features interact with the ARS feature.

- AAR

  ARS and AAR can access the same trunk groups and share the same Routing Patterns, toll lists, and RHNPA tables. ARS calls may be converted to AAR calls.

- GRS

  Generalized Route Selection (GRS) works with ARS to provide call routing over the appropriate trunking facilities. Routing is determined by the type of call being made. With GRS, calls may be routed differently than they would with justARS. For details on GRS, see "GRS" on page 4-56 in this chapter.

- CDR

  An ARS call using a trunk group marked for CDR is indicated by the dialed access code and by a Condition Code. The dialed number is recorded as the called number. Subnet Trunking does not affect CDR.

  If CDR generation is administered for a trunk group assigned to a Routing Pattern, data will be collected for all calls routed through the trunk group.

Administration

ARS is initially assigned on a per-system basis by an AT&T service technician. After the feature is activated, the following items are administered by either the system administrator or the service technician:

- ARS Access Code 1 (one to three digits)
- ARS Access Code 2 (one to three digits)
- ARS Digit Analysis Table
- ARS Digit Conversion Table
- Up to 32 RHNPA Tables
- Up to 32 Toll Lists
- Up to 254 ARS Routing Patterns
- Trunk Groups to be used with ARS
- Whether or not the system returns dial tone after the ARSFAC is dialed on trunk calls

Hardware and Software Requirements

ARS may be used on a stand-alone system or may be an integral part of a private network. No additional hardware is required for a stand-alone system. A private network may require additional tie trunks and TN748B Tone Detector circuit packs. These additions are, however, cost effective when compared to the alternatives for call routing.

BONDing

Bandwidth on demand interoperability group (BONDing) is a standardized form of inverse multiplexing that is an alternative for gaining higher bandwidths between 112k and 384k without the use of ISDN-PRI multirate or H0 channels.

Description

As an alternative to the special hardware and software required for ISDN-PRI for higher bandwidths, an ANSI BONDing inverse multiplexing standard has been established. There are three BONDing modes outlined in the BONDing standard:

- Mode 1

  Provides user data rates in multiples of the bearer rate (56k/64k). This is the BONDing mode currently supported by the AT&T MCU.

- Transparent mode

  The default mode used when BONDing framing is not detected by a specific timeout period.

- Mode 3

  Includes an in-band monitoring function. Uses 7 B-channels instead of 6 B-channels for 384k. The AT&T MCU does not currently support this mode.

Considerations

To ensure proper operation of the BONDing link, the first call must be successfully connected before the AT&T MCU connects the remaining aggregated calls.

H0 calls placed to a BONDed Px64 MCU conference port are automatically dropped.

Two-channel calls placed to a BONDed MCU conference port are added to the conference as audio-only endpoints.

All channels in a BONDed call are required to terminate on the same TN787D circuit pack, which may affect the system port capacity.

Interactions

The following features interact with the BONDing feature.

- Low-/High-Speed Interworking

  BONDed ports cannot coexist with interworking ports in the same conference.

- Audio Add-On

  A maximum of six Audio Add-On endpoints are allowed for each BONDed conference.

- Cascading

  The inter-AT&T MCU link must be specified on the conference record as a specific BONDed bandwidth cascade. If transparent mode is invoked in the cascade link, the AT&T MCUs connected by the link do not share video, and the participants on one AT&T MCU are added to the conference as audio-only endpoints. If a bandwidth between 112k and 336k is specified for the link, the cascade must join the conference at a BONDed mode1 bandwidth of the same bandwidth or as a BONDed transparent mode link. If 384k is specified for the link, the cascade must join the conference at an H0 bandwidth, a BONDed mode 1 384k bandwidth, or as a BONDed transparent mode link.

- Dial-Out

  The Dial-Out feature dials one channel first and then all remaining channels simultaneously when an endpoint is designated as a BONDed endpoint. The calls can all be the same number or all different numbers. Different numbers are supplied from the endpoint equipment through the BONDing protocol.

- Dynamic Resizing

  To join an active H0 conference, a BONDed endpoint may only join if it is running at the same transfer rate.

Administration

The following administration is required for BONDing.

- Conference Record form

  Identify extensions that correspond to BONDing conferees.

- Availability of Ports form

  Use the BONDing option to identify the available BONDing ports.

- Optional Features form

  Enter the number for maximum BONDing port capacity.

Hardware and Software Requirements

See Table 1-1 for feature availability by model. Release 2.0 or later of the AT&T MCU is required for BONDing. The TN787D circuit pack is required for BONDed calls.

Basic/Enhanced Service Flag

Description

There are two user-selectable flags you can use for identifying an AT&T MCU to endpoint interface.

- Basic/enhanced service
- Application compliance

The basic/enhanced service flag is set to identify endpoints that can use the full set of conferencing messages. Some endpoints do not operate properly when other endpoints (or the MCU) attempt to use the full set of messages. If an endpoint cannot operate with the full message set, set the basic/enhanced service flag to basic.

The application compliance flag identifies whether endpoints in a conference can operate properly with MCS/MLP applications.

Considerations

None.

Interactions

The basic/enhanced service and application compliance flags interact with the dynamic resizing feature as follows. The endpoint should be set to enhanced for data sharing with Vistium Release 2 endpoints.

Administration

Translations must be performed to use this feature.

Hardware and Software Requirements

See Table 1-1 for feature availability by model. Only release 3.0 or later versions of the AT&T MCU can provide the service and compliance flags.

BRI/DCP Direct Connect Interface

Description

This feature allows the user to connect Basic Rate Interface (BRI) or Digital Communications Protocol (DCP) endpoints directly to the MCU without involving a public or private network, PBX, or MUX. Up to 12 Direct Connect endpoints can be administered.

Considerations

BRI Direct Connect endpoints support data only. Voice capabilities are not provided via the BRI interface.

The endpoints directly connected to the MCU via BRI or DCP can participate in point-to-point conferences as well as multipoint conferences through the MCU.

The feature requires 48 timeslots and therefore reduces the system port capacity. Therefore, for example, a 16-port BONDed system, which is a maxed-out system, does not have the timeslots to support BRI/DCP Direct Connect endpoints. Also, CRS supports the feature but deducts 48 timeslots from its calculated port capacities.

Interactions

All H.320 and T.120 features supported on the AT&T-MCU are supported by this feature. However, refer to the last paragraph in the previous section for some important notes.

Administration

DCP Direct Connect endpoints are administered in increments of four with a maximum capacity of four and a default value of zero. BRI Direct Connect endpoints are administered in increments of four with a maximum capacity of 12 and a default value of 0.

The following fields in the System-Parameters Customer-Options form must be populated:

- Max. BRI Direct Connect Port Capacity
- Max. DCP Direct Connect Port Capacity

Also, the Type field in the Processor Interface Data module form must be populated.

Hardware and Software Requirements

This feature is supported between an AT&T-MCU running R3.0 (or greater) software and H.320 endpoints.

The TN556 ISDN Line circuit pack, the TN771 Maintenance/Test circuit pack, and the TN778 Packet Control circuit pack are required in every system that contains BRI Direct Connect endpoints. A maximum of one of each of these circuit pack types is supported. Also, a maximum of two TN754 Digital Line circuit packs is supported to allow for the maximum DCP Direct Connect port capacity of four.

Endpoints that are directly connected to the MCU via BRI terminate on the TN556 circuit pack. Endpoints that are directly connected to the MCU via DCP terminate on the TN754 circuit pack. The High-Speed Link (HSL) is the data module used to connect endpoints to the MCU directly via DCP. For 112k and 128k connections, two HSLs connect to the MCU per endpoint.

Business Card

Description

The Business Card feature of the WW model AT&T MCU allows endpoints to share up to 1000 characters of business card information with other endpoints. This can be useful in a conference to display information such as name, title, address, and phone numbers of conference participants during the conference.

Considerations

None.

Interactions

None.

Administration

None.

Hardware and Software Requirements

See Table 1-1 for feature availability by model.

Call Detail Recording

NOTE:
''Call Detail Recording (CDR)'' is also referred to as ''Station Message Detail Recording (SMDR).''

Description

The CDR feature records detailed call information on all incoming and outgoing calls on specified trunk groups and sends this information to a CDR output device. The CDR output device provides a detailed printout that can be used by the telecommunications manager or system administrator to compute call costs, allocate charges, analyze calling patterns, detect unauthorized calls, and keep track of unnecessary calls.

For 2-channel conferences, two records with identical Conference Billing ID fields are stored. For 384K (H0) conferences, a single record is stored on a per-call basis.

Call detail information is provided on trunk groups that are administered for CDR. CDR provides detailed call information for the following types of calls:

- Outgoing Calls — calls originated by Administered Connection/MCU extension, going out on a trunk group.
- Incoming Calls — calls incoming on a trunk group and terminating at an MCU extension.
- Ineffective Call Attempt — (a) unavailable incoming or outgoing trunks due to trunk usage allocation for ISDN Call-By-Call Service Selection trunks and (b) incoming calls rejected by the AT&T MCU due to network specific feature (NSF) mismatch.

You have the option of turning off CDR generation for specific trunk group(s).

NOTE:
Some call accounting systems do not support all the call information offered by CDR. See your account executive for details.

Resource Limitation on CDR Records

If new calls come in when the CDR link is down and the buffer is full (the AT&T MCU has a 180 record maximum), the earliest records are overwritten.

Set Time and Date

The system clock must be set for daylight savings time (if appropriate) when the time changes. Changing the time and date ensures that CDR records have the correct time and date for the records being kept. The time and date can be changed using the AT&T MCU management terminal (MCU-MT).

NOTE:
If the time is changed while calls are in progress, the actual call durations for these calls are not reflected in the CDR record. A ‘‘9999’’ gets output in the CDR duration field. We recommend that you not make date and time changes when there are active calls.

CDR Data Formats

This part covers two types of formats sent to the CDR output device: date record and call detail formats.

Date Record Format

The date record format is shown in Table 4-5.

Table 4-5. Date Record Format to Printer and Expanded Version

ASCII Character Position	Data Field Description
01-02	Month*
03	Space
04-05	Day*
06	Carriage Return
07	Line Feed
08-10	null

* Leading zero added if needed.

Call Detail Record Format

The call detail record format provides detailed information concerning a call. Table 4-6 summarizes the DS1 robbed bit CDR direct output from the system to the printer. Table 4-7 summarizes the 24-word ISDN expanded CDR record format. Call detail records are generated during processing of the call and are sent to the CDR output device in ASCII format when the call is disconnected.

Even though the MCU outputs the record in ASCII character representation, the terminology ‘‘word’’ is used here to mean the ASCII characters.

Table 4-6. DS1 RB CDR Direct Output Format

ASCII Character Position	Data Field Description
01	Time Hour (tens)
02	Time Hour (units)
03	Time Minute (tens)
04	Time Minute (units)
05	Space
06	Duration Hour
07	Duration Minute (tens)
08	Duration Minute (units)
09	Duration Minute (tenths)
10	Space
11	Condition Code
12	Space
13-15	Access Code Dialed*
16	Space
17-19	Access Code Used*
20	Space
21-35	Dialed Number*
36	Space
37-41	Calling Number*
42	Space
43-57	Conference Billing Identification information*
58	Space
59-65	Authorization Code
66-69	Space
70	FRL
71	Space
72	IXC
73	Space
74-76	Incoming Circuit† (hundreds, tens, units)
77	Space

Table 4-6. DS1 RB CDR Direct Output Format — *Continued*

ASCII Character Position	Data Field Description
78-80	Outgoing Circuit[†] (hundreds, tens, units)
81	Space
82	Feature Flag
83	Carriage Return
84	Line Feed

* Data is right-justified and padded with blanks.

† Data is right-justified and padded iwht 0s.

Table 4-7. 24-Word ISDN Expanded CDR Record Format

ASCII Position	Data Field Description
01	Time Hours (tens)
02	Time Hours (units)
03	Time Minutes (tens)
04	Time Minutes (units)
05	Space
06	Duration Hours (units)
07	Duration Minutes (tens)
08	Duration Minutes (units)
09	Duration Minutes (tenths)
10	Space
11	Condition Code
12	Space
13-16	Access Code Dialed*
17	Space
18-21	Access Code Used*
22	Space
23-37	Dialed Number* (The MCU extension in the AT&T MCU)
38	Space
39-48	Calling Number* (TAC of the trunk group for DS1 robbed bit signaling; CPN/BN, if available, for ISDN)
49	Space
50-64	Conference Billing Identification information*
65	Space
66-72	Authorization Code*
73-76	Space
77	FRL
78	Space
79-81	Incoming Circuit ID* (hundreds, tens, units)
82	Space
83-85	Outgoing Circuit ID* (hundreds, tens, units)
86	Space

Table 4-7. 24-Word ISDN Expanded CDR Record Format — *Continued*

ASCII Position	Data Field Description
87	Feature Flag
88	Space
91	Space
92-95	Incoming Trunk Group Access Code*
96-99	Space
100-102	INS*
103-107	Space
108	BCC
109-120	Space
121-122	Bandwidth
123-130	Space
131	Carriage Return
132	Line Feed
133-135	Null

* Data is right-justified and padded with blanks

Call Detail Record Fields

The following list describes the CDR data collected for each call and the number of digits in each field. All information is right-justified in the respective field, unless otherwise indicated.

- Access Code Dialed (four digits on 24-word formats)

  NOTE:
  This field is used only for outgoing calls.

  This field can be the ARS access code, AAR access code, or the access code of a specific trunk group.

- Access Code Used (four digits on 24-word formats)

  NOTE:
  This field is used only for outgoing calls.

This field contains the access code of the actual trunk group that the call was routed over (often the trunk group used is different from the trunk group access code dialed). This field always shows the access code of the used trunk group, even if it is the same as the dialed access code.

- Billing Identification (up to 15 digits)

  NOTE:
  This field contains Account Code.

  This is a numeric field, and contains a number that uniquely identifies a conference. Information in this field is right justified. The billing ID is optional and is administered on the conference form. It allows the system administrator to associate conferences with projects or account numbers.

  If the AT&T MCU cannot allocate resources for a conference, but the trunk call is answered, a CDR report for that call is generated. However, the billing identification field is left blank.

- Authorization Code (seven digits)

  This field contains the four- to seven-digit authorization code used to make the call. This applies to outgoing calls only.

- Bearer Capability Class (BCC) (one digit)

  This field contains the BCC for ISDN calls, identifying the type of an ISDN call. It distinguishes between different types of data. Either of the following digits may appear in this field.

  — 1 = Mode 1 (112k synchronous data)

  — 4 = Mode 0 (128k data clear)

  — w = Multirate bandwidth (128k, 192k, 256k, 320k, 384k, 768k, 1472k, 1536k, or 1920k)

- Bandwidth (two digits)(24-word Record only)

  Used to capture the bandwidth of the multirate bandwidth calls. Bandwidth is expressed as the number of DS0s or 128k channels comprising a call, and is stored in the two-digit bandwidth field (1 or 6).

- Calling Number (up to five, and 10 digits in 24-word format)

  — For *outgoing calls*, this field contains the AT&T MCU extension.

  — For *incoming calls*, this field contains the TAC of the trunk group used for the call (in the standard 18-word formats). In the 24-word format, the calling number field is 10 digits and contains the CPN/BN information, if provided, on incoming ISDN calls. If the CPN information is not output, the field is blank.

- Carriage Return (one character)

  The ASCII carriage return character followed by a line feed is used to terminate CDR records.

- Condition Code (one character)

  The codes in this field reflect special events relating to the call. Table 4-8 describes the condition code mapping.

Table 4-8. Condition Code Mapping

Condition Codes	Description
4	Identifies an extremely long call (10 hours or more) or an extremely high message count for TSC (9999 messages or more). On a call exceeding 10 hours, a call record with this condition code and a duration entry of 9 hours, 59 minutes, and 1 to 9 tenths of a minute is produced after the first period. A similar call record with this condition code is produced after each succeeding 10-hour period. When the call does terminate, a final call record with a different condition code identifying the call type is produced.
7	Identifies calls served by the AAR or ARS Selection feature.
9	Identifies an incoming call.
A	Identifies an outgoing call.
E	Identifies an ineffective call attempt due to trunks not being available. This also identifies an ISDN Call-By-Call Service Selection call that is unsuccessful because of an administered trunk usage allocation plan.
F	Identifies an ineffective call attempt because of either (a) insufficient calling privileges of the originator (assigned per FRL), (b) ISDN calls rejected by the switch due to an NSF mismatch, or (c) an authorization mismatch which prevents the completion of a data call.

NOTE:
When more than one condition applies to a call, the overriding code is shown in the following table.

When two condition codes apply on the same call, one will override the other. The following matrix, Table 4-9, defines the overrides. To illustrate how to use this matrix, assume that condition codes 4 and 9 apply to the same call. The matrix contains four horizontal rows (4, 9, E, and F) and four vertical columns (4, 9, E, and F). To find the condition code that overrides, look at the point of intersection between row 4 and column 9. In this case, condition code 4 overrides. This can also be found by looking at the point where row 9 and column 4 intersect.

Table 4-9. Condition Code Override Matrix

	Condition Code			
	4	**9**	**E**	**F**
4	NA	4	NA	NA
9	4	NA	E	F
E	NA	E	NA	NA
F	NA	F	NA	NA

- Dialed Number (up to 15 digits)

  For incoming calls, this field contains the AT&T MCU extension that corresponds to the dial-out number. If more than 15 digits are dialed, the least significant digits are truncated.

  NOTE:
  Dial out allows 22 digits in the dialed number field. The same truncation rules apply so that, as an example, for the dial-out number 90114567890123456, the dialed number field of the CDR record would show 901145678901234.

  The # sign (“E”) may be printed in this field:

  — When the user dials a feature access code that starts with a #

  — When the user dials # at the end of digit dialing (for example, for WATS and IDDD calls)

  — When the inter-digit timeout occurs before the answer supervision timeout, even if the user has not dialed the # sign.

- Duration (four digits)

  All calls are timed. The timing is recorded in hours (0 through 9), minutes (00 through 59), and to the nearest tenth of a minute (0 through 9).

- FRL (one digit)

  FRLs, numbered zero through seven, are associated with the AAR and ARS features and define calling privileges. The information contained in this field is as follows:

  — If the call is an outgoing call, this field contains the FRL of the originator.

  — If the call is an incoming call, this field contains the FRL assigned to the incoming trunk group.

— If the call is an incoming tandem tie trunk call, this field contains either the FRL assigned to the tandem tie trunk or the Traveling Class Mark (TCM) sent with the tandem tie trunk call, depending on which was used to complete the call. On ISDN calls, this field always contains the TCM, if it was received.

— The CDR System Parameters can be administered to have ‘‘Disconnect Information in Place of FRL.’’ For trunk CDR, the following call disconnect data is printed in this field in place of the FRL data:

Data	Meaning (for calls)
0	Don’t know who dropped first
1	We dropped first
2	The CO dropped first
3	Maintenance got the trunk

- Feature Flag (one digit)

  The digit in this field indicates whether or not the switch has received answer supervision from the network and whether the call was a voice or data call.

  — A **4** in this field indicates a voice call with network answer supervision.

  — A **0** in this field indicates a voice call without network answer supervision.

  — A **5** in this field indicates a data call with network answer supervision.

  — A **1** in this field indicates a data call without network answer supervision.

  Answer Supervision is indicated for non-interworked ISDN calls.

  The answer supervision flag is interpreted as follows:

  — For ISDN trunks, if the answer supervision field contains a **5**, the call interworked with non-ISDN trunks and the duration was calculated but does not have the degree of accuracy of a strictly ISDN call. Thus, this field shows whether the call was interworked or went through a strictly ISDN network.

- Incoming TAC (four digits) (24-Word Records Only)

  This field contains the access code of the incoming trunk group.

- ISDN Network Services (INS) (three digits)

This field specifies the INS requested for a call. This field applies only to ISDN calls. Each Network Specific Facility is translated into an INS according to Table 4-10.

Table 4-10. Network Specific Facility to INS Mapping

Network Specific Facility	INS Value
Network Operator	324
Presubscribed Common Carrier Operator	325
Software Defined Data Network (SDDN)	352
INWATS	355
Maximum Banded WATS	356
AT&T Long Distance Service	358
ACCUNET Digital Service	357
International 800	359
Multiquest	367

- IXC Code (three or four digits with an ISDN format)
  - — Non-ISDN Formats

    IXC codes, numbered 1 through 15 (1 through F hexadecimal), are associated with the AAR and ARS features and depict the carrier used on the call. This information is sent to the CDR output device in ASCII code as a hexadecimal representation (for example, ASCII ‘‘F’’ equals ‘‘15’’).

    An IXC access number is used to access a specific common carrier for a call. In the US, this number is of the form 10XXX, 950 — 1XXX, or NXX — XXXX, where N is any digit 2 through 9 and X is any digit 0 through 9. The IXC access numbers applicable at a given location are associated with an IXC code on the IXC form. When ARS is used, and a routing pattern inserts one of the administered IXC codes, the associated IXC code is recorded. If no IXC access number is used, a 0 is recorded. In this case, either an IXC carrier is not used on the call or the carrier is selected at the CO. A one-character index (1 through 15 and through F hexadecimal) corresponding to the administered IXC code’s index in the first page of the IXC form is generated when that matching IXC code is used on an ARS call. If none of the administered IXC codes is used, a zero is recorded.

IXC codes on the first page of the IXC form can be any one of the following :

- Any number matching the second page of the IXC form
- A 7-digit number of the form 950-XXXX, where ‘‘X’’ is any digit 0 through 9
- Any 8 to 11-digit number

— ISDN Formats

With an ISDN record format, this field is a three-digit field that identifies the actual IXC used on an ISDN call. This information is determined from the routing pattern administration. On AAR and ARS calls, the three-digit IXC value is administered in the routing pattern for all ISDN calls. If a user dials an IXC code with a 10 XXX format as administered on the IXC Codes form, the CDR device will put only the last three digits in the CDR record. If a user dials a seven-digit IXC code, this field will contain a zero.

- Incoming Circuit Identification (three digits)

  This field contains the member number of a trunk within a trunk group used for an incoming call.

- Line Feed (one character)

  The ASCII line feed character followed by a carriage return is used to terminate CDR records.

- Null (one character)

  The NULL is used to terminate and divide CDR Records (usually in triplets) when needed by the receiving adjunct.

- Outgoing Circuit Identification (three digits)

  For outgoing calls, this field contains the member number of the trunk within a trunk group used. This field is blank for incoming calls.

- Space (one to forty characters)

  The ASCII blank character is used to separate other CDR fields or to blank fill unused record locations.

CDR Output Devices

System printer is the only output device currently supported by the AT&T MCU. A standard 232C interface is provided by the system’s processor circuit pack. This allows for direct connection of the CDR output device to the system. If this port is not used, additional interface equipment is required as described in the Hardware and Software Requirements part of this feature description.

If the CDR link is down for more than a minute, some data may be lost. However, the most recent 180 records are stored by the AT&T MCU if the link goes down.

When the link comes back up, these records are output on a ‘‘first-in, first-out’’ basis.

If the CDR buffer is full, the AT&T MCU overwrites the earliest CDR records.

The time stamp on calls recorded by CDR is normally applied at the end of the call.

The following paragraphs give a brief description of the printer as CDR output device.

Printer

An 80- or 132-column (character) printer can be connected as an CDR output device. The printer prints CDR records in a two-line format. No data processing or reports are provided. The 18-word CDR records sent to the printer are 84 bytes or 672 bits long. The 24-word CDR records sent to the printer are 135 bytes or 1,080 bits long.

CDR can be used as a polling device or can be buffered to the following equipment:

- Host computer
- Tape drive
- ASCII printer

Considerations

Capacities

The system can store up to 180 CDR records (when the link is down), which are sent to the output devices when the link comes back up.

DS-1 Call CDR Records

CDR records of DS-1 calls are generated only if the answer supervision timeout is exceeded or if the call is answered at the far end. Therefore, more accurate CDR records for DS-1 facilities can be obtained by setting the answer supervision timeout field on the DS-1 tie trunk form to the highest possible value (250 seconds).

Interactions

For 2-channel conferences, two records with identical Conference Billing ID fields are stored. For 128k, 192k, 256k, 320k, 384k (H0), 768k, 1472k, 1536k (H11), and 1920k conferences, a single record is stored on a per-call basis. For BONDed 336K or BONDed 384K (H0) calls, six records with identical conference billing ID fields are stored. The following interaction discussions assume CDR is activated.

- AAR and ARS

  CDR records the following information for ARS:

  — Fact that an ARS call was made

  — Calling extension number

  — FRL of the calling extension

  — Called number

  — Type of trunk group used for the ARS call

  — Time of call completion

  — Call duration (how long the parties talked)

  — IXC code, if any

  If CDR is suppressed for the trunk group actually used on an ARS call, an CDR record is not generated; otherwise, Condition Code 7 applies. The ARS access code is recorded in the Access Code Dialed field and the trunk access code for the trunk group actually used is recorded in the Access Code Used field.

  If an AAR call is placed to a busy trunk group and CDR is suppressed for that trunk group, the user hears reorder tone and the CDR output shows an ineffective call attempt.

- Call-By-Call Service Selection

  When a successful call is made on a Call-By-Call Service Selection trunk, the network specific facility used on the call is translated into an INS number and recorded in the INS field of the CDR record. If a Call-By-Call Service Selection call is unsuccessful because of an administered trunk usage allocation plan, the INS number is recorded in the INS field of the report with a condition code of ‘‘E.’’

- Intercept Treatment

  If an outgoing or tandem call is routed to Intercept Treatment, the number dialed by the calling party is recorded as the dialed number, and Condition Code F is recorded.

- ISDN

  When specific answer supervision is received from the network, an indication is sent to the CDR device to this effect. If an ISDN call has been interworked, the CDR record will not record the call as having answer supervision.

- Private Network Access

  Private Network Access calls will be recorded if CDR is administered for this trunk group. Private Network calls will be recorded if either an incoming and/or outgoing tie trunk is assigned CDR.

- UDP

  If one user calls another user via a Uniform Dial Plan extension number, and the trunk group used has CDR assigned, CDR records the following information:

  — Condition Code — 7

  — Access Code Dialed — blank

  — Access Code Used — trunk access code of trunk used

  — Dialed Digits — Uniform Dial Plan extension

Administration

CDR is administered by the system administrator. The command is: ‘‘change sys-param features.” The following items can be administered.

System Parameters

- Printer should be specified as the type of CDR output device.
- Whether standard or ISDN formats are used.
- The speed at which the CDR device connected to the direct RS-232C interface on the processor circuit pack will operate (300, 1200, 2400, 4800, or 9600 baud rate).
- Whether the reason for disconnect is recorded instead of the FRL.
- CDR can be suppressed for Ineffective Call Attempts. Ineffective call attempts include calls to incoming or outgoing trunks that are unavailable due to trunk usage allocation for ISDN Call-By-Call Service Selection trunks and incoming calls rejected by the switch due to NSF mismatch.
- Primary output layout: expanded or printer.
- Primary output destination:EIA or the assigned extension.
- ISDN layout?: A yes or no field.
- Data Format: either day/month or month/day format.

Date and Time

The date and time should always be updated for events such as a leap year, daylight savings time, or a system restart after a power failure. If a time of day is not administered, CDR records will not be generated.

IXC Codes

- IXC access numbers
- Name of IXC (optional)

Data Modules and Modems

The CDR output devices can be connected to a modular processor data module (MPDM), Trunk Data Module, or a modem. The following items must be administered:

- A netcon channel must be assigned using a data module form and entering data-channel or netcon channel for the type. This channel provides a path for CDR data from the Switch Processing Element to the time-division bus.
- If the CDR output device is connected to an MPDM, administer an MPDM form.
- If the CDR output device is connected to a Trunk Data Module, administer a Trunk Data Module form.

If the EIA port on the Processor Interface circuit pack is used by the output device, the CDR output device extension should be administered as ‘‘eia.’’

Hardware and Software Requirements

Hardware requirements depend on the type of output device used for CDR. The CDR output device can be connected directly to the processor circuit pack which provides a standard RS-232C interface. If the output device is connected via a 7400B data module (as is the case for a remote CDR printer), the 7400B must be administered so that 7400B result codes do not interfere with CDR data output.

Call-By-Call Service Selection

Description

Call-By-Call Service Selection allows a single ISDN-PRI trunk group to carry calls to more than one service or facility.

NOTE:
The Call-by-Call Service is only applicable for Country Protocol option 1 (US).

Without Call-By-Call Service Selection, each trunk group must be dedicated to a specific service or facility. Call-By-Call Service Selection eliminates this requirement by allowing a variety of services to use a single trunk group. These services are specified on a call-by-call basis. Trunking efficiency is immediately obtained with Call-By-Call Service Selection by distributing traffic over the total number of available trunks.

Services Used with Call-By-Call Service Selection

The services used on Call-By-Call Service Selection calls are assigned after an ISDN-PRI trunk group is assigned a service type of Call-By-Call Service Selection. A Call-By-Call Service Selection trunk group can be administered to carry calls to many services. The services relevant to the AT&T MCU are as follows:

- ACCUNET Digital Service — AT&T's digital network services for various high-volume, high-speed data transmission requirements.
- SDDN — An AT&T offering that provides a virtual private network using the public-switched network. SDDN can carry voice and data between customer locations as well as off-net locations.
- Other user-defined services — New service types can be assigned as they are developed and defined.

A Call-By-Call service selection example is shown in the following figure.

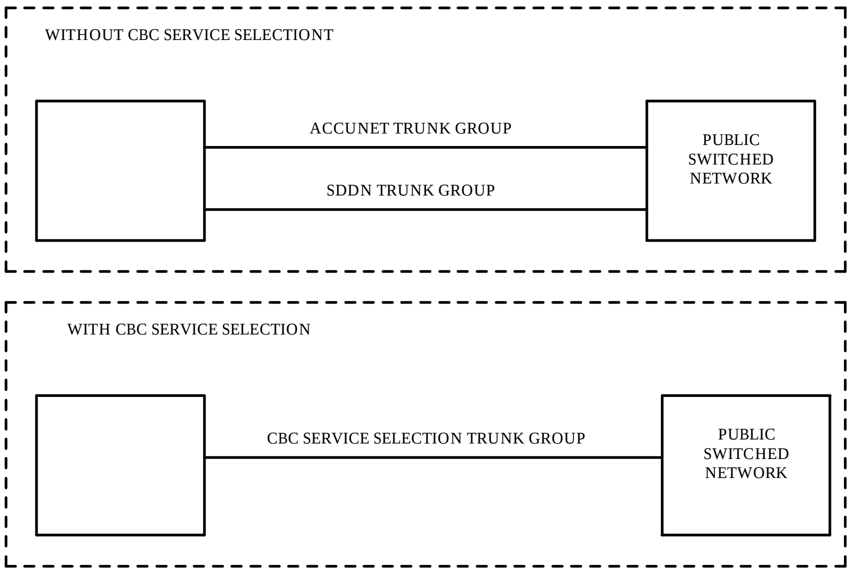


Figure 4-5. Call-By-Call Service Selection Example

ISDN-PRI Messages and Information Elements used for Call-By-Call Service Selection

Although the technical details of ISDN-PRI messages and information elements are not critical to implementing the ISDN-PRI application, the following details may aid in the understanding of some readers and are, therefore, included in this description.

Call-By-Call Service Selection allows the system to specify one of the preceding service types on a call-by-call basis. This is done via a SETUP message that indicates the intent of the originating system to initiate a call using the specified service or facility. The SETUP message contains units called information elements that specify call-related information. The information elements used with Call-By-Call Service Selection are as follows:

- Network Specific Facility — indicates which facilities or services are to be used to complete the call.

  The system also checks all incoming ISDN-PRI calls for the presence of a Network Specific Facility information element. If this information element is present, the system makes sure that the requested service is compatible with the administration of the trunk. If the requested service is not compatible with administration, the switch rejects the call.

Usage Allocation Plan

Optional Usage Allocation Plans (UAPs) may be assigned to provide more control over a Call-By-Call Service Selection trunk group. Up to three Usage Allocation Plans can be assigned for each Call-By-Call Service Selection trunk group. A Usage Allocation Plan allows the customer to set the following:

- A maximum number of trunk group members that each specific service can use at any given time.
- A minimum number of trunk group members that always is available for each specific service.

The sum of the allocation plan maximums may exceed the total number of trunk group members. This ensures that all trunk group members are not dominated by a specific service, yet allows for periodic fluctuations in demand. Also, the sum of the allocation plan minimums may not exceed the total number of trunk group members.

If a UAP has been defined for a Call-By-Call Service Selection trunk group, and the type of the incoming call exceeds one of the plan's limits, the system rejects the call, even if a trunk is available.

Considerations

Call-By-Call Service Selection provides the following benefits:

- Cost Reduction — since many services share the same trunks, the total number of trunks can be reduced.
- Improved Service — Call-By-Call Service Selection trunks can reduce the probability of features and services being blocked.
- Simplified Networking — network engineering is simplified because analysis of trunking needs can be done based on total traffic instead of on a per-service basis.
- The ability to respond to changes in a more timely fashion. The network does not have to be consulted because of the flexibility provided by the usage allocation plans.
- Measurement of Call-By-Call Service Selection calls.

Interactions

The following features interact with the Call-By-Call Service Selection feature.

- AAR

  Call-By-Call Service Selection uses the same routing tables and routing preferences that are used by AAR.
- ARS

Call-By-Call Service Selection uses the same routing tables and routing preferences that are used by ARS.

- GRS

  Call-By-Call Service Selection uses the same routing tables and routing preferences that are used by GRS.

- CDR

  On successful incoming Call-By-Call Service Selection calls, the Network Specific Facility specified by the call's Network Specific Facility Information Element is recorded by CDR. CDR refers to this information as the INS (ISDN Network Service).

  When a Call-By-Call Service Selection call is rejected because of a trunk group usage allocation plan, CDR records the cause as an ineffective call attempt.

Administration

Call-By-Call Service Selection is administered by the system administrator on a per trunk group basis. The following items require administration:

- ISDN-PRI Trunk Group — must be administered with a Service Type of Call-By-Call Service Selection. The trunk group administration also includes the following:
  - Incoming Call Handling Treatment
  - Whether or not UAPs are required
  - UAPs
  - UAP Assignment Schedule
  - Group Member Assignments
- ARS or AAR Routing Patterns — Routing Patterns can be administered to include a Network Specific Facility and/or IXC.
- Network Specific Facilities Encoding — new Network Specific Facilities can be added as needed by the system technician.

Hardware and Software Requirements

A TN767D DS1 circuit pack is required for assignment of a signaling link and up to 23 ISDN-PRI Trunk Group members. A TN2207 circuit pack is required for E1 interfaces for assignment of a signaling link and up to 31 trunk group members.

The DS1 provides 24 ports, and the E1 provides 32 ports. A TN768 Tone-Clock circuit pack is required to provide synchronization for the DS1 circuit pack. A TN765 Processor Interface circuit pack is required in conjunction with the DS1 circuit pack for ISDN Link Administration.

Cascading

Description

Cascading provides the means to connect two AT&T MCUs via a switched PX64 or BONDing link to form a *cascaded* conference. For all practical purposes, each AT&T MCU treats the cascade connection as if it were simply another endpoint in the conference. All the features and services available for non-cascaded conferences are available in cascaded conferences as well. Cascading provides some particular benefits:

- Through cascading, network charges can be greatly reduced for a conference between geographically distant endpoints. Without cascading, all endpoints must call into one AT&T MCU that may be distant from some of the endpoints. This can mean many endpoints, each paying a network toll for each connection to the AT&T MCU. With cascading, the endpoints can connect on an AT&T MCU that is geographically close to them and then request a single line for a cascade call to another AT&T MCU being used by other participants. This can be particularly useful for cross-continent conferences.
- Cascading allows larger conference sizes since the cascaded AT&T MCUs pool resources for as many as 46 multimedia endpoints (23 per AT&T MCU) and as many as 12audio-only endpoints (six per MCU).

Considerations

In a cascaded conference, one AT&T MCU is referred to as primary and the other secondary. Any given AT&T MCU can be primary in one cascaded conference yet secondary in another simultaneous cascaded conference.

Additionally, each AT&T MCU in a cascaded conference honors its own stop time. The first AT&T MCU to reach its individual stop time is dropped from the conference despite the stop time of the second AT&T MCU.

The times for the cascaded MCUs must be within one minute of each other.

Interactions

The cascade feature interacts with the following features.

- CDR

  Each AT&T MCU in the cascade outputs its own CDR records for calls in the conference.
- Dial-Out

  One AT&T MCU in the cascade uses the Dial-Out feature to call the second AT&T MCU to provide the cascading connection.

Administration

It is recommended that the following fields be the same for both AT&T MCUs in a prearranged conference.

- Conference Number
- Conference Name
- Billing ID
- Entry/Exit Tones

Ideally, these fields should be (but need not be) identical for dynamically joined cascades. However, for two independent conferences to be dynamically joined, the following fields must be identical for each.

- Password

  All AT&T MCUs in the cascade must have the same password or no password.

- Number of Channels and Bandwidth

  The effective rate of the conference
  [112k, 128k, 192k, 256k, 320k, 384k, 768k, 1472k, 1536k, or 1920k].

- Conference Mode

  The selected conference control mode must be the same for all AT&T MCUs involved.

To ensure that the dial-in AT&T MCU is ready to receive the connection that is initiated by the dial-out AT&T MCU, the dial-in AT&T MCU should be administered such that the conference is active before the dial-out AT&T MCU.

Hardware and Software Requirements

See Table 4-1 for feature availability by model.

Class of Restriction

Description

Class of Restriction (COR) defines up to 64 different classes of call origination and termination privileges. Systems may have only a single COR, one with no restrictions, or many CORs up to 64 as necessary to effect the desired restrictions.

A COR is assigned to each of the following:

- Maintenance alarm terminals
- Data module
- Trunk group

Use of CORs can be categorized as follows:

- Calling party restrictions (that is, the extension for the maintenance alarm terminal)
- Called party restrictions
- Facility Access Trunk Test
- Fully Restricted Service
- Restriction Override
- Restricted Call List
- Unrestricted Call List
- Miscellaneous restriction groups
- Selective denial of public network calling through a CCSA or EPSCS network
- An ARS or AAR FRL for control of call routing

Features assignable as calling party restrictions are as follows:

- Origination Restriction
- Outward Restriction
- All-Toll Restriction
- TAC-Toll Restriction

Features assignable as called party restrictions are as follows:

- Inward Restriction
- Manual Terminating Line Restriction
- Termination Restriction

- Public Restricted

Use of CORs

CORs can be used to assign a variety of restrictions to a variety of facilities. The types of restrictions that can be assigned are discussed in the following paragraphs. The Unrestricted Call List field is displayed in the Class of Restriction form only if the calling party restriction is administered as TAC-Toll or All-Toll.

Calling Party and Called Party Restrictions

Calling party restrictions prevent specified users from placing certain calls or accessing certain features. Features assignable as calling party restrictions are Code Restriction, Origination Restriction, Outward Restriction, and Toll Restriction. These individual features are fully described elsewhere in this chapter. A brief description is given here:

- Outward Restriction — prevents callers at specified voice terminals from accessing the Public Network Access feature. Calls can be placed to other voice terminal users and to tie trunks.
- Origination Restriction — prevents callers at specified voice terminals from originating calls. Voice terminal users can, however, receive calls.
- TAC-Toll Restriction — prevents callers at specified voice terminals from making trunk access calls to certain toll areas as defined on the system's administered Toll Analysis form, unless the number is on an Unrestricted Call List associated with the caller's COR. This restriction applies to calls made using trunk access codes of CO or FX trunk groups. See the Restriction — Toll feature description elsewhere in this document for more details.
- All-Toll Restriction — identical to the TAC-Toll Restriction described above, except this restriction also applies to ARS calls. See the Restriction-Toll feature description elsewhere in this manual for more details.

Called party restrictions prevent specified users from receiving certain calls. Features assignable as called party restrictions are Inward Restriction, Manual Terminating Line Restriction, Termination Restriction, and Public Restriction. These individual features are fully described elsewhere in this chapter. A brief description is given here:

- Inward Restriction — restricts users at specified voice terminals from receiving public network, access endpoint-originated, and access endpoint-extended calls.
- Manual Terminating Line Restriction — restricts users at specified voice terminals from receiving calls other than those from an access endpoint.
- Termination Restriction — restricts users of specified terminals from receiving any calls.
- Public Restriction — restricts users of specified voice terminals from receiving direct public network calls.

The field labeled `Calling Party Restriction` and the field labeled `Called Party Restriction` are both administered as none. However, the field `Calling Party Restriction` could be administered as any of the other previously described calling party restrictions. Likewise, the field for `Called Party Restriction` could be administered as any of the other previously described called party restrictions. Including ‘‘none’’ as a choice of restrictions, as many as 20 combinations of calling and called party restrictions are possible. However, it is unlikely that all 20 combinations are needed in any one situation. Therefore, only the required ones should be established.

Calling and called party restrictions are the basis for all CORs. In cases where no restrictions are needed, a single COR could be assigned with calling and called party restrictions of ‘‘none.’’ This same COR could be used for unrestricted trunk groups, terminating extension groups, UCD groups, DDC groups, and data modules.

The following are typical examples of calling and called party restrictions that may be assigned to a COR:

- Long-distance calling is limited by All-Toll Restriction, but there are no restrictions on incoming calls.
  - — Calling party restriction=All-Toll
  - — Called party restriction=None

The called party restriction is checked only at the called terminal, module, attendant console, zone, or group.

Each COR is established as needed and is arbitrarily identified by a number. (See Appendix A, ‘‘System Hardware and Software Capacity Limits for Limitations.’’)

Facility Access Trunk Test

This field on the COR form is used to grant a user permission to make Facility Test Calls to access trunks. A y in this field allows users with this COR to make these calls. An n in this field causes users with this COR to receive intercept treatment when they attempt to make these calls. For more information on Facility Test Calls, see the Facility Test Calls feature elsewhere in this chapter.

Fully Restricted Service

Denies the specified voice terminal access to public network trunks for either incoming or outgoing completion. Note that Restriction Override should be set to No.

Restricted Call List

A Restricted Call List is assigned to the system by the system administrator. This call list is made up of specific digit strings that cannot be dialed from facilities that are restricted by the call list. The `Restricted Call List` field on the COR form is used to determine which facilities are restricted by this call list. If a COR has this field administered as y, facilities with that COR cannot be used to dial a number that matches one of the digit strings in the Restricted Call List.

Unrestricted Call List

Ten Unrestricted Call Lists are assigned to the system by the system administrator. Each `Unrestricted Call List` is made up of specific digit strings that can be dialed from facilities associated with the call list. The Unrestricted Call List fields on the COR form are used to determine which facilities have access to each of the 10 call lists. If a COR has any of these 10 fields administered with an Unrestricted Call List number, facilities with that COR can call any numbers contained in the assigned list(s) (unless the number is included in the Restricted Call List, which overrides the Unrestricted Call List).

Selective Denial of Public Network Calling Through a CCSA or EPSCS Network (APLT)

Public network calling via the private CCSA or EPSCS network (commonly referred to as off-network calling) is optional on a per-private network basis. If off-network calling is not provided, then the APLT field can be ignored. If off-network calling is provided, then permission or denial to access the off-network capability is set via the APLT field. Users assigned a COR that has APLT set to **n** (no) can use off-network calling. Users assigned a COR that has APLT set to **y** (yes) cannot. If there is a need for both yes and no choices in a system, separate CORs must be assigned to reflect this.

The field labeled APLT is preset as y. This means that a facility with this COR is not allowed to access CCSA or EPSCS off-network capabilities for public network calling. An n in this field would indicate that the facility can access CCSA or EPSCS off-network capabilities.

ARS/AAR FRL for Control of Call Routing

If the system does not use AAR or ARS to determine the most preferred routing of calls, then the FRL field can be ignored. If AAR or ARS is used, then an FRL is used to either allow or deny access to certain routes. The FRL for the outgoing (trunk) side of the call is provided in the AAR or ARS Routing Pattern. Although each outgoing trunk group has a COR and each COR has an FRL, this FRL is not used unless the trunk group is the originator of the call. Call routing is determined by a comparison of the FRLs in the AAR or ARS Routing Pattern and the FRL in the COR of the call originator.

The **FRL** field is preset to **7**. However, this field can have a value of 0 through 7. An originating FRL of 0 has the least calling privileges, whereas an originating FRL of 7 has the most calling privileges. Each route (for number of routes, see ‘‘System Hardware and Software Capacity Limits’’) in each of the AAR or ARS Routing Patterns also has an FRL. These route FRLs can also have a value of 0 through 7. A route FRL of 0 is the least restrictive, whereas a route FRL of 7 is the most restrictive. An FRL of 0 is checked before the other routes in a given ARS routing pattern. To access a route, the originating FRL must be greater than or equal to the route FRL. Determination of appropriate FRL values must be made with respect to the outgoing routes from a specific system and the desired levels of calling privileges. This is part of AAR or ARS customization. The FRL of the call originator is contained in the COR assigned. The **FRL** field in a COR assigned to an outgoing trunk group is never checked and should be ignored.

Assuming AAR and/or ARS has been customized for a system, the system administrator must establish unique CORs for each of the up to eight levels of ARS calling privileges that is used in the system. However, these CORs must maintain the desired restrictions dictated by the other fields on the screen form. The simplest case is a COR specifying no restriction. Ordinarily, this COR can be assigned to all unrestricted users. However, if some subset(s) of these users requires different FRLs, separate CORs must be established for each different FRL required.

For a detailed description of AAR, ARS, and FRLs, refer to the individual feature descriptions given elsewhere in this chapter.

COR Example

CORs could be used to prevent certain uses of the maintenance alarm terminal for anything other than its intended use.

Considerations

COR provides the means to consolidate assignment and administration of the various restriction features available with the system.

All items associated with a COR are distinct and separate. A unique COR must exist for each needed combination of FRLs, CCSA/EPSCS off-network restrictions, calling party restrictions, called party restrictions, and miscellaneous restrictions. Up to 64 CORs can be established, as required, to provide the needed combinations.

Interactions

The following features interact with the Class of Restriction feature.

- AAR or ARS

  Originating FRLs are assigned via a COR. Termination and Miscellaneous Restrictions do not apply to ARS or AAR calls.

Administration

COR is administered by the system administrator. For each COR that is assigned, the following items must be administered:

- Access to Malicious Call Trace

- COR Number
- FRL
- Permission to access EPSCS or CCSA off-net facilities
- Calling Party Restriction
- Called Party Restriction
- Permission to call other CORs
- Forced Entry of account codes for CDR (yes or no)
- Partitioned Group Number
- Priority Queuing (yes or no)
- Service Observing (yes or no)
- Time of Day Plan Number
- Direct Agent Calling
- Facility Access Trunk Test
- Fully Restricted Service
- Restricted Call List
- Unrestricted Call List

Assignment of Restrictions

A COR is assigned to each of the following:

Voice Terminals

All voice terminals must be assigned a COR. The same COR may be assigned to all voice terminals or a unique COR may be assigned to a particular voice terminal or group of voice terminals. This COR applies individually to each voice terminal and is independent of all other COR applications, such as Miscellaneous Restriction groups or UCD groups.

The main items of concern for individual voice terminals are calling party restrictions and called party restrictions (discussed previously under ‘‘Use of CORs’’). If no restrictions are needed for a certain group of voice terminals, ‘‘none’’ can be specified for both calling party and called party restrictions. If it is desired to restrict a group of voice terminals from making outside calls, a COR specifying a calling party restriction of ‘‘outward’’ should be established.

Additionally, miscellaneous restrictions, restrictions to CCSA and EPSCS off-network calling capabilities, and FRLs also apply. A separate COR must be established for each unique set of restrictions.

Trunk Groups

Each trunk group is assigned a COR. Trunk groups are assigned CORs mainly for the use of miscellaneous restrictions.

Calling party and called party restrictions should be ‘‘none.’’ Whether or not a CO or FX trunk group is restricted is specified on the trunk group form used during implementation.

CO and FX trunk groups default to being toll restricted for TAC calls. Toll Restriction for TAC calls can be disabled for certain CO or FX trunk groups on the trunk form.

Data Module

Each data module is assigned a COR.

Hardware and Software Requirements

No additional hardware or software is required.

Conference Redial Flag

Description

This feature allows a user to ask the reservations agent to redial the call. This can occur for any of the following reasons:

- The user failed password validation three times and wants to try again.
- The user missed the password entry window.
- The user disconnected from the conference and wants to rejoin.

There may be other legitimate reasons as well. With the redial flag set, a failed call receives the same redial treatment as any other dial-out call.

Considerations

For an endpoint that has one connection established successfully, the redial applies only to the second connection.

Interactions

None.

Administration

Set the redial flag from blank to yes to enable redial.

Hardware and Software Requirements

See Table 4-1 for feature availability by model.

Conference Reservation System

Description

The CRS feature is an optional software package used on a personal computer or personal computer network to view and add AT&T MCU information. With this feature, one or more reservations agents can perform system administration and conference reservation functions.

There are single-user and multiuser versions available for the CRS software. The multi-user version allows a company such as a service provider to subdivide (or *partition*) logically the MCUs to which it is attached so that several reservations agents and administrators can work within their partition as if on a separate MCU.

Considerations

To use the CRS software, you must supply the personal computer or network system as described in the *AT&T Conference Reservation System User's Manual*, 555-230-520.

Interactions

CRS supports a number of MCU features, including (among others) Cascading, Dynamic Resizing and Dedicated Access. See the *AT&T Conference Reservation System User's Manual*, 555-230-520 for details.

Administration

An AT&T MCU-ST is administered as an additional AT&T MCU-MT. CRS-specific administration, including MCU partitioning, is described in the *AT&T Conference Reservation System User's Manual*, 555-230-520.

Hardware and Software Requirements

See Table 4-1 for feature availability by model.

See *AT&T Conference Reservation System User’s Manual*, 555-230-520 for hardware and software requirements.

DS1 Trunk Service

Description

DS1 trunk service provides a digital interface for the following:

- Robbed-Bit DS1 Tie Trunks
- ISDN-PRI Trunks
- CAS DS1 Tie Trunks

Tie Trunks

Robbed-Bit DS1 tie trunks permit data calling between the AT&T MCU and another switch, can connect the system with other digital switches, and can be used to dynamically access data SDN services.

The TN767D DS1 circuit pack is used to support Robbed-Bit DS1 tie trunks in the Robbed-Bit Signaling mode.

DS1 transmission facilities in the US and Canadian DS1 format is a 1.544-M digital signal consisting of a 1.536-M signal multiplexed with an 8-k framing signal.

E1 transmission facilities format is a 2.048-M digital signal and requires the TN2207 universal DS1 circuit pack. E1 supports CAS signaling used in the United Kingdom.

Considerations

Each DS1 circuit pack can support up to 24 trunks: 24 trunks for transmission in the Robbed-Bit Signaling mode or 23 trunks for clear data transmission, and one trunk to transmit signaling for the other 23 trunks in the signaling mode or ISDN-PRI signaling mode.

The system can support a maximum of 25 DS1 circuit packs.

The TN2207 circuit pack is used for E1 trunks to provide 32 ports and includes an E1 interface cable.

Interactions

None.

Administration

DS1 trunks are assigned on a per-line basis by the system administrator. The following items require administration:

- DS1 Circuit Pack — assign the circuit pack to the system before administration of the associated trunks.
- Synchronization Plan — administer to provide synchronization between the switch's DS1 circuitry and the digital facilities to which the switch is connected.
- Trunk Groups — associate the trunks to groups, if desired.

Hardware and Software Requirements

One TN767D DS1 circuit pack is required for every 24 trunks using Robbed-Bit Signaling or for every 23 trunks using ISDN signaling.

A TN768 circuit pack is required to provide synchronization for theDS1 trunks.

For E1 (32 port) trunks, use the TN2207 universal DS1 circuit pack.

No additional software is required.

Dedicated Access

Description

Dedicated Access allows endpoints to participate in multipoint conferences via nonsignaled T1/E1 facilities. To accomplish this, the AT&T MCU is connected either through a Digital Access Crosspoint System (DACS) multiplexer (MUX) or directly to the H.320 endpoints via DS1 facilities. The AT&T MCU supports up to 25 DS1 connections for non-signaling interfaces.

Considerations

Each Dedicated Access endpoint is assigned a fixed number of DS0s (1-6) that are always identified with that endpoint. To connect in a conference, the endpoint uses one or more (depending on the conference bandwidth) of those assigned DS0s. Bandwidths of 128k, 192k, 256k, 320k, 384k, 768k, 1472k, 1536k, and 1920k are supported for Dedicated Access.

Interactions

Audio Add-On, cascade links, and BONDed facilities have to be accessed via switched facilities but can participate in a conference if at the same bandwidth as the Dedicated Access endpoints.

Administration

Dedicated access endpoint extensions and conference bandwidth for a given endpoint are administered on the Wideband Access Endpoint (WAE) Form.

Hardware and Software Requirements

See Table 4-1 for feature availability by model.

Dial-Out

Description

The Dial-Out feature allows the AT&T MCU to place outgoing calls automatically to conference participants using an Administered Connection. This is in addition to the basic AT&T MCU capability that allows conferees to call the AT&T MCU to join a conference.

A conference convener can use the dial-out feature in order to convene a conference automatically. At the conference start time, the AT&T MCU automatically dials all conference participants simultaneously.

Considerations

To establish the dial-out links, the AT&T MCU automatically translates a dial-out administered connection for each conferee number. This administered connection remains in place until the conference is completed. When the conference is completed, the AT&T MCU automatically untranslates the administered connection and makes it available for reuse.

If one of the conference participant numbers is a cascade link (that is, another AT&T MCU), the dialing AT&T MCU attempts to connect to the second AT&T MCU first and then calls all the other endpoints simultaneously. If the dial-out call fails, the AT&T MCU uses the failure cause code to determine its next action depending on whether the link is a cascade or a non-cascade link and the type of failure. The number of retries pertaining to post connection (answer) failures is administered on the System-Parameters Features form.

Cascade Link Retry Strategy

The retry strategy that dial out uses for a cascade link is as follows.

- If the failure is the first failure reported, retry immediately.
- If the failure occurred after a connection was established, retry the administered number of times, then raise a warning alarm on the administered connection and stop retries.
- For all other causes:

  If the alarm threshold has been reached, raise a warning alarm on the administered connection.

  Retry the connection in approximately one minute.

Since a cascade link can represent many conferees, the retry attempts for a cascade link are only halted after the administered number of post-connection retries are attempted. If the link is for a dedicated conference, the assumption is made that the link must always remain up. Therefore, retries are attempted no matter what the failure cause unless there is an administration mismatch between the two cascaded AT&T MCUs or the administered number of post-connection retries is reached.

Non-Cascade Link Retry Strategy

The retry strategy that dial out uses for a non-cascade link is as follows:

- If the failure is a far-end disconnect, stop trying to connect.
- If this is the first failure, retry immediately.
- If the failure occurred after a connection was established, retry the administered number of times, then raise a warning alarm on the administered connection and stop retries.
- For all other causes:

  Log an error in the hardware error log.

  If the alarm threshold has been reached, raise a warning alarm on the administered connection.

  Retry the connection in approximately one minute.

Interactions

The following features interact with the dial-out feature.

- AAR/ARS

  Dial out can use the AAR/ARS feature to route outgoing calls over the proper trunk facilities by including the AAR/ARS feature access codes in the destination number(s) of a conferee.
- BONDing

  Dial out can originate BONDed calls.
- Call-by-Call Service Selection

  Dial out can use the call-by-call service selection feature to route outgoing calls over specific ISDN-PRI networks.
- Cascading

  Cascading uses the dial-out feature to establish the inter-MCU link.
- CDR

  If the dial-out feature uses a trunk group with the CDR feature on, the origination extension (MCU extension), destination number, and conference billing ID are reported.
- Class of Restriction

The system administrator can prevent stations connected to the AT&T MCU from making outgoing calls by administering the stations with an Outward Restriction COR.

- DS1 Trunk Service

  Robbed-bit trunks can be used as a network facility in routing dial-out calls.

- ISDN-PRI

  ISDN-PRI trunks can be used as a network facility in routing dial-out calls. In addition, the MCU extension is sent as the calling identifier if enabled on the trunk group form.

- System Measurements

  All trunks used for dial-out calls are measured as with any other trunk use.

- System Status Report

  The status AC command is supported for dial-out administered connections.

Administration

Administration for this feature is required for outgoing trunk groups and AAR/ARS routing patterns. Dial-out numbers must be administered for each conference where they are to be used. If the conference is a single-channel high-speed conference, a single dial-out number is administered. If the conference is a 112k/128k conference, two dial-out numbers are administered. In either case, if ARS or AAR are to be utilized in the calls, the access codes for those features should be the first digit(s) of the dial-out number(s).

The maximum number of post-answer retries must be administered in the ‘‘Dial-out Post Answer Failure Retry Attempts’’ field of the System-Parameters Features form.

Hardware and Software Requirements

See Table 4-1 for feature availability by model. Dial out requires AT&T MCU software release 1.1 or later.

Dial Plan

Description

This feature enables the administration of the video port extensions used within the AT&T MCU to identify and route calls to video ports. It also enables the administration of extensions for entities such as the maintenance alarm terminal and an CDR printer port extension. In addition, feature access codes are defined by this feature.

Each video port is identified by a unique extension. As an example, this extension is used on the Conference Record Form and within the maintenance subsystem to identify the appropriate video port associated with a given call.

The dial plan, or first-digit table, established during administration for each system, provides information in conjunction with the AAR features on the AT&T MCU on what to do with digits received from the network or internal sources such as the maintenance alarm terminal. The following parameters define extensions and feature access codes (used internally by the AT&T MCU or the maintenance alarm terminal).

- Extension numbers

  Used to define video port extensions, maintenance alarm terminal extensions, and other extensions when applicable.

- Feature access codes

  A minimum of one digit is required but a maximum of three digits can be used. The * and # buttons can be used as the first digit of a feature access code.

- Trunk access codes

  A minimum of one digit. The digit can be any number in the range 1 through 9. For example, you could set 9 for local trunks and 8 for tie trunks.

Interactions

This feature controls extension number formats for video port and other extensions. It must be completed prior to using extensions and/or feature access codes on the system.

Administration

The system administrator administers the dial plan on a per-system basis. The following administration is required:

- Area code where the AT&T MCU is located (if in the United States). Blank if the AT&T MCU is located in Australia, Singapore, or the United Kingdom.
- Enter ‘‘y’’ in the “Uniform Dial Plan” field

Hardware and Software Requirements

No additional hardware or software is required.

Dynamic Resizing

Description

This feature allows a reservations agent to add and remove endpoints from a conference both before and during the conference. This capability includes the single Audio Add-On, cascade links, dial-out, and Meet Me endpoints. Using this feature, dial-out numbers can be removed and changed, and conference stop times can be changed. In addition, this feature allows dial-out numbers to be changed to dial-in numbers (and vice versa) and primary/secondary AT&T MCU types to be changed in a cascade link. It also allows the basic/enhanced service flag, redial flag, and interworking status during a conference to be changed..

Considerations

Dynamic Resizing is capable of using existing AT&T MCU resources only. For example, it cannot add more than six Audio Add-On endpoints.

Interactions

The Dynamic Resizing feature interacts as described in the individual feature descriptions.

Administration

Your AT&T MCU must be administered for Dynamic Resizing.

Hardware and Software Requirements

See Table 4-1 for feature availability by model.

Facility Test Calls (with Security Measures)

Description

This feature provides a maintenance alarm terminal user with the capability of making test calls to access specific trunks, DTMF receivers, time slots, and system tones. The test call is used to make sure the facility is operating properly. The maintenance alarm terminal user can make a test call by dialing an access code. AT&T remote maintenance personnel may also use this feature to make test calls.

Four types of Facility Test Calls can be made:

- Trunk test call

  Used to test specific tie trunks.

- DTMF receiver test call

  Accesses and tests the four DTMF receivers located on a Tone Detector circuit pack.

- Time slot test call

  Connects the maintenance alarm terminal user to a specific time slot located on the TDM buses or out-of-service time slots.

- System tone test call

  Connects the maintenance alarm terminal user to a specific system tone.

Security Measures

To help secure this feature from unauthorized use the following steps can be taken:

- Remove the code when not in use.
- Change the code from the factory default.
- Secure records of the code.
- Use COR to restrict which users can use the access code.

Consult the *GBCSystems Security Handbook*, 555-025-600, for additional steps to secure your system and find out about obtaining information regularly about security developments.

Considerations

If a user has a problem with a specific system facility, Facility Test Calls can be used to test that facility for proper operation.

The maintenance alarm terminal must be used to make test calls.

NOTE:
AT&T has designed the Facility Test Calls feature incorporated in this product that, when properly administered by the customer, will enable the customer to minimize the ability of unauthorized persons to gain access to the network. It is the customer's responsibility to take the appropriate steps to properly implement the features, evaluate and administer the various restriction levels, protect access codes and distribute them only to individuals who have been advised of the sensitive nature of the access information. Each authorized user should be instructed concerning the proper use and handling of access codes.

In rare instances, unauthorized individuals make connections to the telecommunications network through use of test call features. In such events, applicable tariffs require that the customer pay all network charges for traffic. AT&T cannot be responsible for such charges, and will not make any allowance or give any credit for charges that result from unauthorized access.

Interactions

None.

Administration

The Facility Test Calls feature is administered on a per-system basis by the system administrator. The Facility Test Calls access code must be assigned while performing tests and should be removed when not in use.

Hardware and Software Requirements

No additional hardware or software is required.

Facility and Non-Facility Associated Signaling

Description

This feature provides signaling for ISDN-PRI.

Facility Associated Signaling

Facility Associated Signaling (FAS) allows an ISDN-PRI DS1/E1 Interface D-channel to carry signaling information for only those B-channels located on the same DS1/E1 facility (circuit pack) as the D-channel.

Non-Facility Associated Signaling

Non-Facility Associated Signaling (NFAS) allows an ISDN-PRI DS1/E1 Interface D-channel (signaling channel) to convey signaling information for B-channels (video channels) on ISDN-PRI DS1/E1 facilities other than the one containing the D-channel. As a result, a D-channel can carry signaling information for numerous B-channels located on different DS1/E1 facilities. The AT&T MCU supports up to 10 T1s.

D-channel Backup

To improve reliability in the event of a signaling link failure, a backup D-channel may be administered. If a signaling link failure does occur, a switch to a backup D-channel will then take place.

D-channel Backup requires that one D-channel be administered as the Primary D-channel and that a second D-channel be administered as the Secondary D-channel. These assignments ensure that at certain times during D-channel Backup procedures that both D-channels are in the same state. This avoids the occurrence of both switches at each end of the DS1/E1 interface selecting the same D-channel to be put into service. In these cases, the Primary D-channel is given precedence over the Secondary D-channel.

The following figure shows a possible configuration involving three ISDN-PRIs between the AT&T MCU and a PBX or the public network. With DS1 (24 channel) interfaces, two of the ISDN-PRIs contain a D-channel and 23 B-channels, while the other ISDN-PRI contains 24 B-channels. One of the D-channels is the Primary D-channel, and the other is the Secondary D-channel. Together, this pair of D-channels will signal for all 70 (23+24+23) of the B-channels that are part of the three PRIs.

Since the D-channels are signaling for more than one DS1/E1 facility, the D-channel Backup feature requires the use of the NFAS feature. At any given time, one of the two D-channels will be carrying layer 3 signaling messages, while the other D-channel will be active at layer 2, but in a standby mode only. Any layer

3 messages received over the standby D-channel will be ignored. The two D-channels can provide signaling for only a predefined set of B-channels and cannot dynamically back up other D-channels on other interfaces.

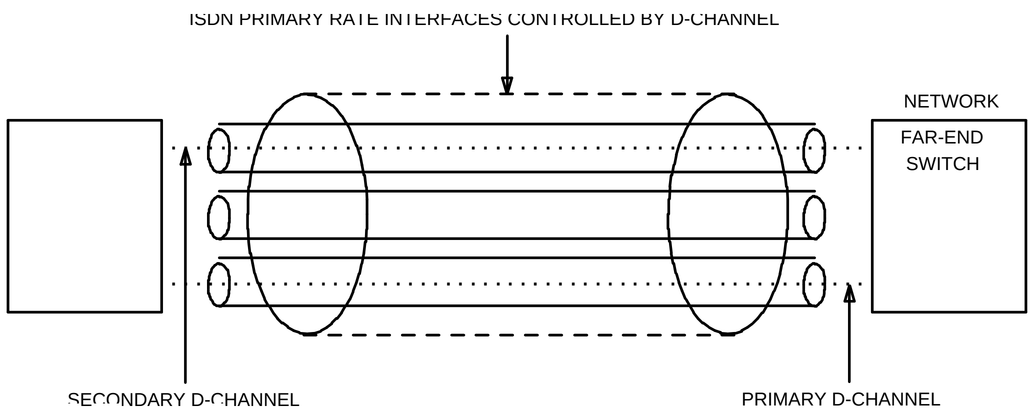


Figure 4-6. Example D-channel Backup with Three ISDN-PRIs

D-channel Backup Activation

D-channel Backup can be invoked in response to the following events:

- D-channel Failure

  If the signaling link fails on the active D-channel (D1) or the hardware carrying D1 fails, then the system will send a message over the standby D-channel (D2), which requests that D2 become the active D-channel. D2 then becomes the active D-channel and will carry all subsequent signaling messages. When the signaling link or hardware on D1 recovers from the failure, D1 becomes the standby D-channel.

- System Technician Commands

  If a system technician command requests that a D-channel switch-over take place, the first action taken by the system will be to tear down the signaling link on D1. After this has been completed, a message is sent on a D2 to request that D2 become the active D-channel. D2 then becomes the active D-channel and the switch-over will be completed.

Considerations

NFAS allows four H0 calls on a DS1/E1 that does not contain a D-channel.

Only those ISDN-PRI facilities that use the NFAS feature will be capable of providing the D-channel Backup feature. The reason for this limitation is that the two D-channels must be located on different PRI DS1/E1 facilities. As a result, the

D-channels must support NFAS so that they can signal for B-channels on different ISDN-PRI DS1/E1 facilities.

When a transition from one D-channel to another occurs (D-channel Backup is activated), all stable calls (calls that have been answered already) will be preserved. Some messages may be lost, resulting in a loss of call-related information, but the calls themselves will be maintained. The effect of the transition on unstable calls (those that have not been answered yet) is unpredictable since the results depend on which messages were lost and the contents of those messages.

NFAS is not supported in Australia, Singapore, or the United Kingdom.

Interactions

None.

Administration

The following provisioning and administration must be considered when implementing FAS and NFAS. Coordinate the following with the far-end switch for the DS1/E1 facilities to be used:

- Decide which DS1/E1 facilities will use FAS.
- Decide which of the remaining DS1/E1 facilities will carry D-channel signaling information on the 16th (E1) or 24th (DS1) channel. For those channels that have a D-channel Backup, D-channel pairs must be allocated.
- Define Signaling Groups (1 through 8). A Signaling Group is a group of B-channels for which a given D-channel (or D-channel pair) will carry the signaling information. Each Signaling Group must be designated as either a FAS or NFAS Signaling Group.
    - A FAS Signaling Group must contain all the ISDN B-channels on the DS1/E1 interface associated with the group’s D-channel, and cannot contain B-channels from any other DS1/E1 circuit pack. For 24-channel DS1 boards, some of the DS1 ports may use in-band (robbed-bit) signaling and be members in a tie trunk group rather than an ISDN trunk group. These tie trunks cannot be members of a Signaling Group.
    - There is no restriction on which DS1/E1 ports can belong to an NFAS Signaling Group. Normally, an NFAS Signaling Group consists of one or two D-channels and several complete DS1/E1 interfaces.

If a Signaling Group contains only a subset of a DS1/E1's B-channels (ports 1 through 12, for example), it is considered an NFAS Signaling Group, not a FAS Signaling Group. The remaining B-channels on the DS1/E1 will then be assigned as members of another NFAS Signaling Group.

- An Interface ID must be assigned to each DS1/E1 facility in an NFAS Signaling Group. For example, if the B-channels in a Signaling Group span three DS1/E1 facilities, a unique Interface ID must be assigned to each of the three DS1/E1 facilities. This designation is required to uniquely identify the same B-channel (port) number on each of the DS1/E1 facilities in the signaling group. Therefore, this designation must be agreed upon by both sides of the interface and administered prior to initialization.
- D-channel Backup involves two or more ISDN-PRI DS1/E1 facilities that interconnect the AT&T MCU to a PBX or to the network. Two D-channels must be present on the facilities. One of the D-channels is designated as Primary and the other as Secondary. This designation must be agreed upon by both sides of the interface and administered prior to initialization.

Hardware and Software Requirements

See "Integrated Services Digital Network — Primary Rate Interface" on page 4-82 in this chapter, for hardware and software requirements.

H.261 Annex D Still Image

Description

With this feature, you can transmit still images in the video channel of the H.221 format based upon procedures specified in H.261 Annex D. Still images can be transmitted at up to 4 times that of CIF images.

Administration

None.

Considerations

To transmit still images to other endpoints while in a voice activated switching conference without interruption, first request ‘‘see-me’’ and send the image then relinquish see-me control.

Interactions

None.

Hardware and Software Requirements

See "Integrated Services Digital Network — Primary Rate Interface" on page 4-82 in this chapter for feature availability by model.

Integrated Services Digital Network — Primary Rate Interface

Description

This feature allows connection of the system to an Integrated Services Digital Network (ISDN) by using a standard ISDN frame format called the Primary Rate Interface (PRI). The ISDN provides access to a variety of public and private network services and facilities. The AT&T MCU ISDN-PRI implementation is consistent with the International Telecommunications Union-Telecommunications (ITU-T) Recommendation Q.931 and Q.921.

The T1 is a digital transmission standard that in North America carries traffic at the digital signal level-1 (DS1) rate of 1.544 M. T1 facilities are also used in Japan and some Middle-Eastern countries. It consists of a 1.536-M signal multiplexed with an 8k framing channel.   The 1.536-M signal is divided into 24 channels (DS0s) of 128k each, numbered 1 through 24. The ‘‘D’’ (data) channel multiplexes signaling messages for the ‘‘B’’ (bearer) channels carrying data. When a D-channel is present, it occupies channel 24.

The E1 is a digital transmission standard that carries traffic at a rate of 2.048 M. It is used in Europe and elsewhere. The E1 facility is divided into 32 channels (DS0s) of 64 kpbs information numbered 0-31. Channel 0 is reserved for framing and synchronization information. When a D-channel is present, it occupies channel 16.

ISDN-PRI signaling in the system is supported by the TN767D (for 24 channels). The D-channel (signaling channel) is switched through the TN765 circuit pack.

With the ISDN-PRI, the system can interface with a wide range of other products including network switches and PBXs. These products include the following:

- Public Network switches (for example, 4ESS, 5ESS®, Northern Telecom™ DMS250, etc.)
- DEFINITY® Communications System Generic 2 and System 85 R2V4
- DEFINITY Communications System Generic 1
- DEFINITY Communications System Generic 3
- Other products that adhere to the ISDN-PRI signaling protocol

As an example of how the ISDN-PRI is used in private and public network configurations, see Figure 4-3 and Figure 4-4. As seen in these figures, the ISDN-PRI can be used to interface the AT&T MCU with a PBX.

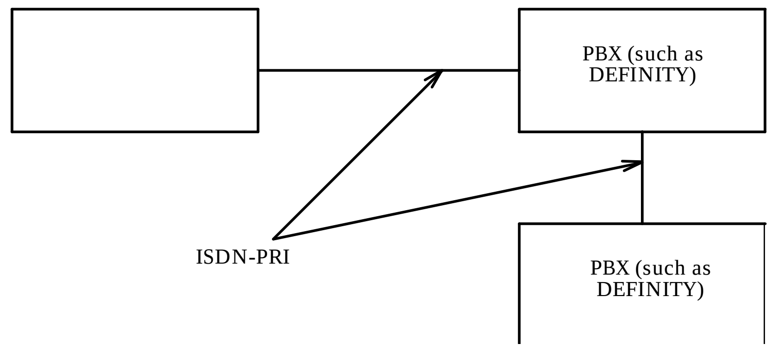


Figure 4-7. ISDN-PRI Private Network Configuration

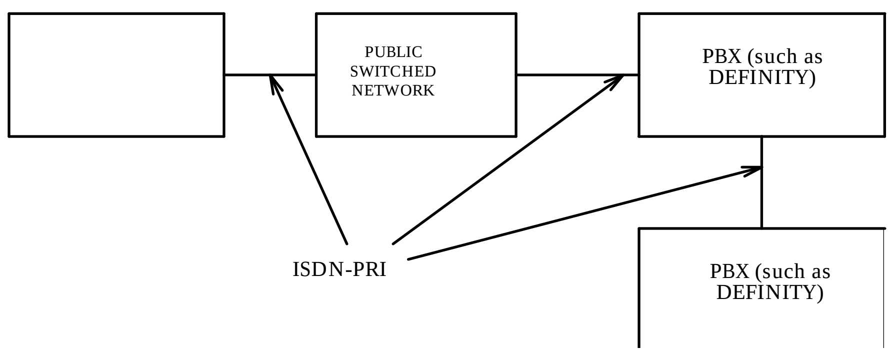


Figure 4-8. ISDN-PRI Public Network Configuration

ISDN-PRI Services

The ISDN-PRI provides the following services:

- Private Network Services
- Multirate bandwidths of 128k, 192k, 256k, 320k, 384k, 768k, 1472k, 1536k, and 1920k (both private and public networks) for video conferencing
- Call-By-Call Service Selection
- Access to AT&T Switched Network Services

Access to AT&T Switched Network Services

ISDN-PRI provides access to AT&T Switched Network Services such as Software Defined Data Network (SDDN), etc. An ISDN-PRI trunk group may be dedicated to a particular feature. Alternately, an ISDN-PRI Call-By-Call trunk group may provide access to several features. For a description of the ASN services accessible via ISDN-PRI (either via dedicated or Call-By-Call trunk groups), see "Call-By-Call Service Selection" on page 4-53 in this document.

ISDN-PRI Call Identification Display

Two types of identification numbers are provided with the ISDN-PRI. These identification numbers may be used in the various types of displays used with the ISDN-PRI. The two types of identification numbers are as follows:

- Billing Number (BN): The calling party's billing number that is provided to an inter-exchange network via Equal Access or Centralized Automatic Message Accounting (CAMA). This number is stored at either a local or network switch. If a customer is connected directly to the AT&T network, the BN is the customer's billing number stored in that network. If the CPN is not provided on an incoming ISDN-PRI call, the system uses the BN for the station identification number.

Private Network Services

In addition to providing access to switched public networks, the ISDN-PRI can provide private network services by connecting in an Electronic Tandem Network (ETN) configuration. This gives customers more efficient private networks that support new integrated voice and data services. Services are provided as follows:

- Multirate Bandwidth

  The primary function of the Multirate Bandwidth feature is to provide support for services that require large bandwidth, such as high-speed video conferencing at 128k, 192k, 256k, 320k, 384k, 768k, 1472k, 1536k, or 1920k. These services have traditionally been handled by dedicated facilities. With the Multirate Bandwidth feature, dedicated facilities are no longer a requirement for these large bandwidth services.

Call-By-Call Service Selection

Call-By-Call Service Selection allows the same ISDN-PRI trunk group to carry calls to a variety of services or facilities (such as a SDDN and ACCUNET). This feature is described in detail under the Call-By-Call Service Selection feature description.

Access to Software Defined Data Network (SDDN)

With ISDN-PRI, the SDDN service may be accessed. SDDN provides virtual private line connectivity via the AT&T switched network (4ESS switch). The services provided by SDDN include voice, data, and video applications. SDDN services complement the Software Defined Network (SDN) voice services.

Access to Switched Digital International (SDI)

SDI provides 128k unrestricted connectivity to international locations via the AT&T switched network. It is also the backbone for the AT&T International ISDN network. SDI complements the ACCUNET digital service already available to US locations. This service can be accessed using the Call-By-Call Service Selection feature.

Considerations

ISDN-PRI is required for video conferences at data rates greater than 112k. CBC and NFAS data channel backup are available if required.

Interactions

None.

Administration

ISDN-PRI is administered on a per-system basis by the system administrator. The following items require administration.

- Communication - Interface Link
- Communication - Interface Processor
- DS1 Circuit Pack (for DS1 and E1)
- DS1 Synchronization Plan (for DS1 and E1)

Processor Interface Circuit Pack

- ISDN-PRI Trunk Group
- GRS Routing Patterns
- Signaling Group (See "Facility and Non-Facility Associated Signaling" on page 4-78 in this document).

Hardware and Software Requirements

- A TN767D DS1 circuit pack is required for assignment of a T1 signaling link and up to 24 ISDN-PRI Trunk Group members. The DS1 provides up to 24 ports.
- A TN2207 DS1 circuit pack is required for assignment of an E1 signaling link and up to 31 ISDN-PRI Trunk Group members. The DS1 provides up to 31 ports.
- A TN768 Tone Clock circuit pack is required to provide synchronization for the DS1 circuit pack.

- A TN765 Processor Interface circuit pack is required for ISDN-PRI. The TN765 supports four ports (four D-channels).

One processor interface link is required per ISDN-PRI.

Low-/High-Speed Interworking

Low-/High-Speed Interworking allows lower-speed (56k and 64k) endpoints to join a higher-speed conference as Audio Add-On endpoints.

Description

A lower-speed endpoint can connect with a single channel to a single channel higher-speed conference as an audio-only endpoint. There is no limit to the number of low-speed endpoints that can join the conference in this way.

Considerations

Low-/High-Speed Interworking is allowed only for 56k and 64k endpoints. The low-speed endpoint connects with only one channel and only to high-speed single channel conferences. During a conference, this feature can be administered from off to on but not on to off.

Interactions

The Low-/High-Speed Interworking feature interacts with the following features.

- CDR

  CDRs for the call indicate the number of channels used by each call.

- Cascading

  Low-speed calls coming on a cascade link are blocked.

- BONDing

  The feature is not allowed in a BONDed conference.

- Dial-In/Dial-Out

  Only dial-in endpoints can use this feature.

- Passwords

  If passwords are administered for the conference/endpoint, the AT&T MCU requests a password from the low-speed endpoint even if that endpoint is joining as an audio-only endpoint.

- MCS/MLP

  Since low-speed endpoints join as audio-only, they cannot access the MCS/MLP features.

- Chair Control

  The MCU denies chair control to any audio-only endpoint.

- CRS

  If interworking is enabled for a conference, then CRS relaxes conditions for selecting endpoints but still does not allow BONDed ports since interworking is blocked in conferences with BONDed endpoints.

- Dynamic Resizing

  If dynamic resizing is enabled for a conference, then interworking can be administered from off to on during the conference. Interworking can never be administered from on to off during a conference.

Administration

Low-/High-Speed Interworking is enabled by administering it on the conference form for each conference.

Hardware and Software Requirements

See Table 4-1 for feature availability by model. Low-/High-Speed Interworking is available only on release 3.0 or later of the AT&T MCU.

Multipoint Communication Service/Multilayer Protocol (MCS/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.

Description

MCS/MLP is not directly used by conferees but is accessed via the use of other conference applications that use MCS/MLP primitives to access this feature via the AT&T MCU. MCS/MLP supports multipoint service for interactive, network-independent transport of data. Users within an MCS/MLP conference can set up dynamic sub-conferences where any one of the users can broadcast data.

The AT&T MCU provides the following types of MCS/MLP services:

- As many as 24 MCS/MLP connections per ESM
- Both static and dynamic channel IDs
- Normal send and uniform send of multicast data

- Multiple users per node

These services support MCS/MLP applications such as whiteboard, application sharing, and file transfer.

Considerations

The Expansion Services Module (ESM) is required to support MCS/MLP. Also, although MLP may coexist with any of the audio modes supported by the MCU, MLP requires the G.728 (LD-CELP) audio mode for operation.

Interactions

The MCS/MLP feature interacts with the following features:

- **Audio Mode**

  When MCS/MLP is used in a conference, the audio mode is set to G.728 LD-CELP.

  If MCS/MLP is disabled or enabled during a conference, the audio mode for the endpoints may be reevaluated and changed depending on the conference bandwidth and endpoint audio-mode capabilities. The following table describes the audio mode changes in an active conferences where MCS/MLP is enabled or disabled.

Table 4-11. Audio-Mode Change Based on MCS/MLP State During a Conference

Conference Bandwidth	MCS/MLP State Change	Audio-Mode Adjustment
2B	enabled to disabled	In a conference where all endpoints support G.728, the video bandwidth is increased. If only some endpoints support G.728, then G.711/audio-only endpoints are upgraded to G.711/video and G.728 endpoints are converted to G.711/video.
2B	disabled to enabled	Video bandwidth is decreased and an MCS/MLP channel is opened for endpoints that support G.728. Endpoints that do not support G.728 are changed from G.711/video to G.711/audio-only.

- **Dynamic Resizing**

  MCS/MLP endpoints can be added or dropped during a conference. However, MCS/MLP cannot be enabled or disabled for an active conference. The MCS/MLP option is set at the time the conference is reserved and cannot be changed dynamically.

- **Application Compliance Flag**

  When the application compliance flag is turned off for an endpoint that supports both var-MLP and G.728, the AT&T MCU blocks any data sends or receives to or from that endpoint.

- **Selected Communication Mode (SCM)**

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

Administration

MCS/MLP is activated on a per-system basis. To enable MCS/MLP, specific fields in a number of forms must be populated correctly. The following list identifies the appropriate forms, and it indicates how the fields therein should be populated.

- **System-Parameters Customer-Options form**

  The “Maximum Port Capacity: MCS/MLP” field should be set to the appropriate value in the range of **4** to **24** in intervals of four.

- **Conference Record form**

  The “MCS/MLP Data Mode” field must be set to **ww-pcs**. This entry indicates that only endpoints with the WorldWorx stack compliance ns-cap are allowed into the conference. Other endpoints have an MLP channel opened but are not connected to MCS.

  Also, the “App Comp” (“Application Compliant”) field must be set to **y**.

- **DS1 Circuit Pack form**

  If the “MMI Cabling Board” field is populated, the “MMI Interface” field appears and must be populated with **ESM** to indicate that the DS1 circuit pack is cabled to the ESM-MMI circuit pack. Entering this value into the field causes an additional form page to appear. Refer to the System Administration and Reports document for instructions on populating the fields that appear on this page.

- **Processor Channel Assignment form**

  The “Application” field must be populated with **ESM**. There are several requirements relevant to entering this value. For details, refer to the System Administration and Reports document.

Hardware and Software Requirements

MCS/MLP requires two TN787 MMI circuit packs. One MMI (H.221-MMI) is used to terminate the H.221 bit stream and the second one (ESM-MMI) performs the MLP bit rate adaptation. Also required are the TN765 Processor Interface and the TN2207 UDS1 circuit packs.

See Table 4-1 for feature availability by model. The MSM and ESM hardware is required for an AT&T MCU to use the MCS/MLP features. The MSM and ESM should be within 50 feet of each other.

Multirate Bandwidth

Description

The ISDN-PRI divides a T1 (E1 for international switches) trunk into 23 (31 for E1) information channels and one signaling channel for standard narrowband communication. Certain applications, such as video conferencing, require greater bandwidth, and it becomes necessary to aggregate several of those narrowband channels into one ‘‘wideband’’ channel to accommodate the extra bandwidth requirement. The Multirate Bandwidth feature provides an AT&T MCU with the ability to receive 128k, 192k, 256k, 320k, 384k, 768k, 1472k, 1536k, or 1920k calls.

Technical Description

The Multirate Bandwidth feature provides end-to-end connectivity between endpoints according to the ITU-T and ANSI standards for 128k, 192k, 256k, 320k, 384k, 768k, 1472k, 1536k, or 1920k calls.

NOTE:
Typically, a T1 facility provides for 23 B-channels and 1 signaling while an E1 facility provides 30 B-channels and 1 D-channel. If however, Non-Facility Associated Signaling (NFAS) is used, a group of T1/E1 facilities can be configured to share a single signaling channel allowing all but one of the T1/E1 facilities to be configured with an extra B-channel. The AT&T MCU supports up to 25 T1/E1 facilities.

An AT&T MCU can be configured for Multirate Bandwidth to support multipoint conferences at 128k, 192k, 256k, 320k, 384k, 768k, 1472k, 1536k, or 1920k. Each B-channel can provide 64k transmission rates. There are two service types for encoding the aggregation of channels, both of which are supported by the AT&T MCU.

Nx64 allows the aggregation of any channel groupings, while 384k (H0) and 1536K (H11) have fixed channel assignments.

Channel Allocation

Multirate bandwidth channel allocation is performed using one of three allocation algorithms: fixed, flexible, or floating. This subject is discussed in greater detail in the Implementation Guide, but in brief:

- Fixed allocation provides contiguous channel aggregation and the starting channel is constrained to a predetermined starting point. (Used only for 384k (H0), 1536K (H11), and H12 calls.)
- Flexible allocation allows an Multirate Bandwidth call to occupy non-contiguous positions within a single T1 or E1 facility.

- Floating allocation enforces contiguous channel aggregation but the position of the first channel is not constrained like it is in fixed allocation.

Considerations

Blocking is the most pertinent trunking consideration when using Multirate Bandwidth.

Blocking

Blocking occurs when the number of B-channels required to make a call are not available. Narrowband (128k) calls require two channels and are usually not blocked. However, blocking occurs for Multirate Bandwidth calls when bandwidth is not available in the appropriate format (that is, fixed, floating, or flexible, see earlier description).

Interactions

The following features interact with the Multirate Bandwidth feature:

- ACA

  Treats Multirate Bandwidth calls as single trunk calls so that only a single ACA referral call is made if an ACA referral call is required. The referral call is on the lowest B-channel associated with the Multirate Bandwidth call.

- CDR

  Multirate Bandwidth calls trigger CDR records containing the Multirate Bandwidth Bearer Capability Class (BCC) and the call bandwidth.

- ISDN-PRI

  Standard ISDN-PRI interfaces can be administered for Multirate Bandwidth switching.

Administration

Multirate bandwidths are administered on a per-system basis via the Optional Features form. Also, the Trunk Group (ISDN-PRI) form must be administered for Multirate Bandwidth service.

Trunk Groups

Trunk group administration is required to specify the Multirate Bandwidth services that are to be supported by each trunk group. The parameters 384k (H0), 1536K (H11), and NXDS0 must be administered to identify the algorithm that the trunk group uses to provide Multirate Bandwidth service.

Hardware and Software Requirements

The TN767D or TN2207 circuit pack must be used for any facility that carries Wdeband traffic. The only other additional hardware or software required for Multirate Bandwidth is that the PRI endpoints must adhere to the ISDN-PRI wideband interface requirements and to PRI line-side requirements.

The TN2207 is required for E1 facilities.

Networking and Call Handling

Description

The AT&T MCU provides a flexible and robust networking interface. This networking interface consists of the network trunks (DS1 and PRI) that provide the physical connection to the network (private or public) and the call handling software that processes the incoming and dial-out calls. Additionally, the network interface supports flexible call handling such that each video port supports variable bandwidths (transmission rates) on a per-conference basis.   Therefore, no hardware nor software modifications are required to the system beyond the setting of the bandwidth parameter on the Conference Administration form. As an example, video port 51000 can be administered in a 128k conference that lasts from 10:00 to 11:00. At 12:00, a new conference administered for 384k (H0) can reuse video port 51000 without any hardware or software modifications.

The AT&T MCU supports Robbed-bit DS-1 and ISDN-PRI trunks for connecting to the public switched network or a PBX (see the ISDN-PRI section). All video calls gain access to the MCU via these trunks. For each incoming call, the MCU expects to receive an MCU number (that is, a network number assigned to the MCU). The received MCU number can be used by the reservations agent (or system administrator) to help identify which endpoint originated the call and is always used by the MCU to identify the video port to which the call is to be connected. As with other equipment connecting to a network, it is expected that the network dial plan and call routing is already provisioned and in place prior to attempting video calls to the MCU. In the same way, outgoing (dial-out) requires the AT&T MCU dial plans, trunk groups, and routing be provisioned and in place prior to attempting a video call to the network.

Considerations

The objective of the Network/Call Handling Subsystem is straightforward in that calls are received from or sent to the network and connected to the proper video port within the MCU. Once the MCU is connected to the network, and the network administrator has provisioned the network dial plan, calls to the MCU can be made and the dial-out feature can be used to make calls. For each incoming video call, the MCU receives the incoming digits (that is, MCU number). (See the “Dial-Out” section in this chapter for details on outgoing call (dial-out) MCU numbers). This MCU number is used internally to route the call to the proper video port. Each video port is identified via its extension (that is, video port extension). Once internal processing of the MCU number is completed, the call is connected to a video port.

Interactions

To implement the function outlined above, the networking or call handling subsystem uses the following MCU features:

- DS-1 Trunk Service

  Enables administration of one to several DS-1 trunks groups.

- ISDN-PRI

  The ISDN-PRI feature enables the administration of the ISDN-PRI parameters for trunks administered for ISDN-PRI signaling.

- Dial Plan

  Enables the administration of the video port extensions.

- AAR

  AAR provides the call handling capability. MCU numbers for incoming calls are analyzed and, in turn, converted to the appropriate Video Port extension. The Video Port extension is then used to connect the call to the appropriate video port.

To handle incoming calls properly, the following steps must be completed:

1. Each trunk group must be administered to prepend an 8 on each MCU number received from the network

   NOTE:
   The network administrator is responsible for properly administering the network to deliver the entire MCU number (that is, network number).

   The network should be provisioned to deliver to the AT&T MCU the MCU numbers such that they correspond to the numbers distributed by the Reservations Agent and thus the number dialed by the end user. To join a 128k or 384k conference, a MCU number must have an administered bandwidth of 128k/384k. To join a 112k conference, a MCU number may have an administered bandwidth of 112k or 128k/384k.

2. The Dial Plan feature must be administered. Once completed, the Video Port extensions exist (4- or 5-digit extensions) and the AAR feature access code (8) is provisioned.

3. The AAR Digit Conversion Table must be administered such that each MCU number exists in the ‘‘Matching Pattern’’ field.

   NOTE:
   The full 7-, 10-, 4-digit (or whatever dial plan is implemented in the network and in turn passed to the MCU) number must be entered in this field. Directly across from each entered MCU number in the ‘‘Replacement String’’ field is a Video Port extension. The Video Port extension must correspond to the Dial Plan administered via the Dial Plan feature.

See "Dial-Out" on page 4-71 in this chapter for details of the requirements that need to be met to properly handle outgoing (dial-out) calls.

Administration

See individual feature.

Hardware and Software Requirements

See individual feature.

Notification Package

Includes the following:

- Terminal names feature
- Conference tones
- Video-switching mode notification
- Broadcaster notification
- Time-left notification tone
- Disconnect notification tone
- Play tone command

Description

The notification package is a collection of features. The Terminal Names feature is described in the "Terminal Names" on page 4-110 in this chapter. The conference tones are described in Chapter 8, "Technical Specifications". Broadcaster notification is a tone that sounds at a tones-capable endpoint when it becomes the broadcaster in a conference. The time-left notification tone is a “10-minute warning” tone to notify conference endpoints that the conference connection is about to end. The play conference tone command allows the WorldWorx PCS Meeting Reservation and Control System to command the MCU to play a conference termination warning tone. The tone played is the same tone that the MCU plays when 10 minutes are left in a reserved conference. Along with this tone, the MCU sends the number of minutes left in the conference. The play tones command can also be used to send only the minutes left without the tone to a single endpoint indicating to that participant that the conference is ending.

Considerations

None.

Interactions

None.

Hardware and Software Requirements

None.

Passwords

The passwords feature provides additional conference security by requiring a password entry for access to a conference on a per-conference or per-user basis.

Description

With per-user passwords set at the conference reservation, the AT&T MCU prompts each user identified for a password as they attempt to join the conference.

Audio-only conferees are given a single chance to provide the correct password and, if a password is incorrect, the endpoint is dropped from the conference. All other conferees are given three chances to enter the correct password before being dropped. In either of these cases, the Status Conference form shows “invalid password’’ as the drop reason for that endpoint.

Considerations

None.

Interactions

- Dial out

  When the AT&T MCU dials out for an endpoint that is set for auto-answer, the MCU prompts for the password after the endpoint answers. After a timeout with no endpoint entry (administered in the range 30-300 seconds) the AT&T MCU prompts a second time. If the endpoint still does not respond within the timeout, the AT&T MCU prompts a third time. If there is still no response from the endpoint, that endpoint is dropped from the conference. In this case, the reason for drop is set to ‘‘UIN/password timeout.’’

Administration

The timeout value for auto-answer endpoints must be set to a value in the range 30-300 seconds. The timeout value is the same for all conferences. The password flags for per-conference or per-user for each conference must be selected at reservation time. The password flag must be selected at each time a conference is reserved.

Hardware and Software Requirements

See Table 4-1 for feature availability by model. The passwords feature for non-audio-only endpoints is only available with MCU release 3.0 or later.

Rate Adaptation

Rate Adaptation provides a way for interaction of 2-channel (56k or 64k) endpoints in a conference.

Description

With Rate Adaptation, 56k endpoints can be included into a 64k conference. If a 56k endpoint joins a 64k conference, the 64k endpoints are dynamically altered to an effective rate of 56k. The 56k rate remains in effect until the conference ends.

Considerations

None.

Interactions

The Rate Adaptation feature interacts with the following:

- Dial-Out

  Dial-Out is capable of initiating calls using Rate Adaptation.

- Dynamic Resizing

  Rate Adaptation can be enabled (turned on) during a conference if Dynamic Resizing is enabled. It can never be changed from on to off during the conference.

Administration

The conference record must be administered to allow or disallow 56k or 64k rate adaptation in the conference.

Hardware and Software Requirements

None.

Recent Change History

Description

The Recent Change History feature allows the system administrator or reservations agent to view or print out a history report of the most recent administration and maintenance changes.   This report may be used for diagnostic or information purposes.

The system maintains a log in a software buffer of the most recent administration and maintenance commands, up to a maximum of 250 data commands. The log is called the transaction log. The commands must be ‘‘data affecting’’ and successfully entered to be saved in the transaction log. The ‘‘data affecting’’ commands are called data commands.

The transaction log can be displayed or printed as a report by entering the **list history** or **list history print** command at the Management Terminal, or a remote terminal by the following users:

- Local Customer Administrator
- Local System Technician
- Remote Customer Administrator
- Remote System Technician

Considerations

A maximum of 250 data commands are stored in the transaction log.

The Permission Administration Form shows the command permission categories that a user can access; for example, ‘‘Administer Features.’’ There are no permission restrictions for access to the Recent Change History reports. The local and remote customer and system technician administrators have unrestricted access.

Interactions

The following features interact with the Recent Change History feature.

- Other Users

  When a user requests a Recent Change History report, it takes a little time to read all the pages of the report. If during this time other users are entering data commands and altering the transaction log, the oldest entries in the transaction log may have been overwritten by the data commands entered by these other users.

- Set Time Command

  The use of the maintenance command **Set Time** to change the system clock could make the Recent Change History report look as if it were not in true last-in, first-out order.

Administration

None.

Hardware and Software Requirements

No additional hardware or software is required.

Report Scheduler and System Printer

Description

This feature allows the system administrator to schedule selected administration commands to be printed by an asynchronous printer. Reports are scheduled at 15-minute intervals for any combination of days of the week. Most **list**, **display**, or **test** commands may be scheduled.

Reports may be scheduled, changed, listed, and removed via the system's Management Terminal.

Considerations

With the Report Scheduler and System Printer, the system administrator can schedule most **list**, **test**, and **display** administration commands to be printed at various times on an asynchronous printer. By scheduling these reports to print automatically at the desired times, the system administrator saves valuable time that can be used to perform other administrative duties.

The system administrator can schedule a maximum of 50 individual reports. The system has a single asynchronous printer connection dedicated for use by the report scheduler. The system can be configured with other printers in the system such as those connected to the AT&T MCU-MT and the CDR printer. These are not used by the Report Scheduler feature.

Reports scheduled for the same time and day are printed according to their order in the Report Scheduler queue. The first report in the queue will be printed first.

To present the least possible impact on system performance, it is recommended that reports be scheduled at off-peak hours and staggered so they are not all scheduled to be printed at the same time.

Reports that are added to the scheduler queue and that are scheduled to be printed during the current time interval will not be printed until the next scheduled time.

If a system error is encountered while trying to print a scheduled report, the error will be printed on the report, just as it would be displayed for the same command on the Management Terminal screen.

Interactions

There is only one processor board EIA port available for asynchronous output. The port cannot be administered for both CDR and the Report Scheduler System Printer on the System-Parameter Feature form.

Administration

The system administrator may schedule, list, change, and remove the desired reports as previously described in this description. Before these procedures can be done, however, the system administrator must supply printer information on page 4 of the System-Parameters Features form by entering the following information:

- **Printer Extension:** ‘‘EIA’’ for the EIA port or a valid data module extension if the EIA port is not to be used.

  The system administrator must specify the printer link by selecting either the EIA port, if available, or a data-module extension. If the data-module extension is chosen, the system administrator must have administered the extension previously using the **add data-module** command.

- **EIA Device Bit Rate:** The speed of the printer (1200, 2400, 4800 or 9600 baud). The default is 1200.
- **Lines per Page:** The number of printed lines per page (24 to 132). The default is 60.

Hardware and Software Requirements

The asynchronous printer can be connected to the switch using either of the following methods:

- The printer can be connected directly to the EIA port on the switch’s processor board. In this case, the appropriate cable is required.
- The printer can be connected to the AT&T MCU with a 7400B data module and a port on a TN754B Digital Line circuit pack.

There is a single EIA port in the AT&T MCU. There may be contention between the CDR and the Report Scheduler feature for use of this port. If the Report Scheduler feature is using the EIA port and you would like to enable the CDR feature, it is recommended that you disconnect the system printer from the EIA port and use a data module for its connection, freeing the port for use by CDR.

An AT&T 475 or AT&T 572, or compatible printer that uses a serial interface may be used as the System Printer. A PC may be connected to the system printer port for collection of data; however, a serial interface on the PC must be provided for the connection.

Security Violation Notification

Description

This feature notifies the maintenance alarm terminal of an attempt to breach system administration access via an invalid login or Remote Access via an invalid barrier code. Information specific to access violations is available through the Monitor Security Violations status report.

Considerations

A maximum of one feature button (per system) may be assigned for each component of the security violation notification (SVN) feature [one for login security violations (LSVN) and one for Remote Access security violations (RSVN)].

An activated SVN feature button cannot be deleted by administration. The system will deny an attempt to delete an active SVN feature button.

Activation of an administered SVN feature button from the maintenance alarm terminal will be prohibited if the appropriate SVN feature component (either login or Remote Access) is not enabled in the Feature-Related System Parameters screen.

A maximum of one Referral Destination may be administered per AT&T MCU for each component of the SVN feature.

The Call Coverage, Call Forwarding, and Call Pickup (for example, within pickup group) features are not supported for SVN.

Unlike the ACA referral call, the SVN referral call is a priority call.

SVN does not currently support Speech Synthesis Circuit pack functionality.

Interactions

- ACA

  The originating extensions for login and remote access security violations cannot be the same as the originating extensions for ACA long and short holding time originating extensions. The destination extensions for both features can be the same.

Administration

The SVN feature is activated when either or both of the SVN Login Violation Notification Enabled or SVN Remote Access Violation Notification Enabled fields on the Feature-Related System Parameters form are enabled.

When the SVN Login Violation Notification Enabled field is enabled, the following additional fields appear on the Feature-Related System Parameters screen:

- Originating Extension

  This field requires the entry of an unassigned extension that is local to the switch and conforms to the dial plan for the purpose of originating and identifying SVN referral calls for Management Applications login violations.

  The originating extension initiates the referral call in the event of a login security violation. It also triggers the display of the appropriate alerting message on the referral destination's display module.

  NOTE:
  The SVN feature button does not have to reside on the referral destination station. It can be administered on the maintenance alarm terminal, but must be active if referral calls are to be placed.

- Referral Destination

  This field requires the entry of a previously administered maintenance alarm terminal. The destination may enable or disable the referral call through a feature button. The SVN feature buttons do not have to reside on the station(s) that is/are the referral destination(s). A referral call will not be made to the administered destination if its corresponding feature button is deactivated.

  The Referral Destination is the extension assigned to the maintenance alarm terminal that will receive the referral call in the event of a login security violation.

- Login Threshold

  This field requires the entry of the minimum number of invalid login attempts that will be permitted before a referral call is made. The value of this field, in conjunction with the Time Interval field, will determine whether a security violation has occurred.

- Time Interval

  This field requires the entry of the time period in which the login security violations must occur. The range for the time interval is one minute to eight hours (00:01 to 07:59), and is entered in the form xx:xx. For example, if you want the time interval to be one minute, you would enter 00:01. If you want the time interval to be seven and one-half hours, you would enter 07:30.

- Activating Station

  The Activating Station must have an LSVN call button to activate and deactivate this feature.

When the SVN Remote Access Violation Notification Enabled field is enabled, the following additional fields appear on the Feature-Related System Parameters screen:

- Originating Extension

  This field requires the entry of an unassigned extension that is local to the switch and conforms to the dial plan for the purpose of originating and identifying SVN referral calls for remote access barrier code violations.

  The originating extension initiates the referral call in the event of a login security violation. It also triggers the display of the appropriate alerting message on the referral destination's display module.

  NOTE:
  The SVN feature button does not have to reside on the referral destination station. It can be administered on the maintenance alarm terminal, but must be active if referral calls are to be placed.

One SVN feature button (per system) may be assigned for each component of the SVN feature (one for login security violations and one for Remote Access security violations).

Hardware and Software Requirements

No additional hardware or software is required.

Selected Communications Mode (SCM) Upgrades

Description

The SCM configures conferences as endpoints join using the highest common approach for optimum audio mode, video resolution, and minimum picture interval (MPI). This feature also automatically upgrades the audio mode, video resolution, and MPI as endpoints leave the conference.

Consider the following example. If all the endpoints in a given conference support G.728, the audio mode is automatically set to G.728. If an endpoint joins the conference but does not support G.728 (and the conference does not require data sharing), all endpoints are downgraded to the audio mode that the new endpoint supports. If the new endpoint subsequently disconnects from the conference, all the endpoints are upgraded again automatically back to G.728. The same is true for upgrade and downgrade of QCIF or CIF video resolution and MPI.

Administration

SCM upgrade can be administered on a per-conference basis.

Interactions

SCM upgrade interacts with the following features.

- Dynamic Resizing

  Dynamic resizing allows the automatic upgrading and downgrading during active conferences.

- Cascading

  SCM upgrade or downgrade is not allowed in a cascaded conference.

- Rate Adaptation

  SCM upgrade and downgrade are allowed in a conference that uses rate adaptation.

Considerations

None.

Hardware and Software Requirements

None.

Service Flags

Description

Endpoints from several vendors tend to crash whenever these endpoints receive certains standard BAS commands/caps. To prevent this problem the Basic/Enhance Service Flag in the Conference Record can be set to *basic* (default). This disables commands/caps that are known to cause problems. If no such problems are apparent, the flag should be set to *enhanced*, where appropriate. Also, the basic/enhanced service and application compliance flags allow the types of messages allowed in a given conference to be identified.

Some endpoints do not operate properly with MCS/MLP messaging and must be identified in the conference to prevent problems. If all the endpoints in a conference can use the MCS/MLP messaging, the application compliance flag is set to allow those messages.

Considerations

None.

Interactions

The service flags feature interacts with the following features:

- MCS/MLP

  The application compliance flag is pertinent only to systems that have the MCS/MLP feature option.

Administration

Each conference record must be administered at conference reservation for these flags.

Hardware and Software Requirements

Only the AT&T MCU Release 3.0 or later has the service flags options.

Status of WorldWorx Conference

Description

The **status ww-conference** command allows the WW model AT&T MCU to gain a status report of all the current conferences in the system. Static administration information is not included in this report, and fields are represented in integer format rather than characters.

Considerations

None.

Interactions

None.

Administration

None.

Hardware and Software Requirements

See Table 4-1 for feature availability by model.

System Measurements

Description

System measurements provide reports on items such as trunk group usage, hunt group usage and efficiency, maintenance alarm terminal activity and efficiency, and security violations.

Individual reports are available as detailed in *AT&T MultiPoint Control Unit (MCU) System Administration and Reports*, 555-027-727&.

All reports are on-demand reports. None are given automatically. Reports are available on the remote administration terminal. The reports can be printed if a printer is associated with the terminal. The reports can also be scheduled to print at the system printer via the Report Scheduler and System Printer feature.

Considerations

Reports provided by System Measurements contain data that is useful to determine group efficiency.

Traffic measurements are automatically accumulated by the system and are available on demand. However, reports are not archived. If needed, reports must be requested periodically. Obtaining a printed copy can aid in maintaining a history of the system traffic.

Detailed information of each call handled by a specific trunk group, if required, must be provided by the CDR feature. Processed CDR data can also provide detailed information on trunk group usage. However, if individual call details are not required for bill-back or cost-allocation, System Measurements should be considered as the means to determine and maintain trunk group efficiency.

Interactions

None.

Administration

Measurements for the Coverage Paths, Coverage Principals, Route Patterns, Trunk Group Hourly, and Wideband Trunk Group Hourly reports are only collected for objects that have been administered using the **change meas-selection** commands.

Hardware and Software Requirements

System administration terminals are required to monitor system measurements. A system printer is required to generate paper copies of the reports. No additional software is required.

System Status Report

Description

The system status report allows the user to view data associated with the maintenance alarm terminal, major and minor alarms, and traffic measurements. The information is displayed on the Management Terminal, and it presents a basic picture of the system condition. The report can only be displayed by the system administrator and maintenance personnel.

The Status Report is displayed by entering one of the commands listed below. Once the command is entered, the system continually displays the report until it is canceled.

NOTE:
Canceling a **monitor system view1** or **monitor system view2** report results in automatically logging the user off the system.

- **monitor system view1**

  This command displays the following information:

  — Activation status of the maintenance alarm terminal (updated every minute)

  — Maintenance status, which includes major and minor alarms for trunk ports, terminal ports, and all maintained objects in the system except terminals and trunks (updated every minute)

  — Traffic measurements for trunk groups, hunt groups, and the maintenance alarm terminal (updated every hour)

- **monitor system view2**

  This report is a subset of the view1 report and displays the same information listed for the view1 report except the last hour's measurement for the hunt groups.

Considerations

In addition to providing status reports, this feature also provides an indication that the administration terminal is functional. Any attempt to stop the ‘‘monitor system view1/view2’’ reports logs the administration terminal off the system. Therefore, no unauthorized administration can be performed.

Interactions

None.

Administration

None required.

Hardware and Software Requirements

No additional hardware or software is required.

Terminal Names

Description

Terminal names allows the MCU to query and pass endpoint terminal names as defined in the ANSI243 (ANSI version of H.243) recommendation. With the terminal names option, endpoints can identify themselves and other endpoints involved in a conference.

Considerations

None.

Interactions

None.

Administration

Terminal names can be administered on a per-conference basis.

Hardware and Software Requirements

See Table 4-1 for feature availability by model. Terminal names are available for Release 3.0 or later releases of the AT&T MCU.

Uniform Dial Plan

Description

The Uniform Dial Plan (UDP) provides a common 4- or 5-digit dial plan (specified by Dialplan administration) that can be shared among a group of AT&T MCUs. UDP maps an incoming 7-digit number into a local MCU number.

Interactions

The following features interact with the UDP feature.

- AAR

  After the system determines the RNX of the switch being called, AAR routes the call to the correct switch. The required subset of AAR is provided with the UDP software. If the AAR feature is provided in addition to the UDP, the seven-digit AAR number will provide the exact same routing as the UDP.

- Dial Plan

  All of the extension numbers on a switch are not necessarily part of the UDP. Any that do not belong to the UDP are handled by a regular, non-UDP dial plan associated with the local switch. When administering the dial plan and designating a group of extensions as UDP non-local, you can specify on the Dialplan form whether you want to search for local extensions first or last. This allows some flexibility in the changing of extensions from local to non-local. However, after the dial plan is changed to make an extension UDP, nothing can be administered with these extensions on the local switch.

Administration

The UDP is administered by the system administrator. The following items require administration:

- Whether UDP has 4- or 5-digit extension numbers PBX Codes (expands first one, two, three, or four digits of dialed extension to an RNX)
- AAR Analysis Table (used by AAR to route calls to the correct switch)
- Routing Patterns
- Node Number Routing (used to route ENP calls)
- Extension Codes and type(s) of conversion
- AAR Digit Conversion (to define conversions for AAR Code extensions or to define home location codes)

NOTE:
If the user changes the Uniform Dialing Plan field value from a ‘‘y’’ to an ‘‘n,’’ then a warning message is generated to inform them that this action causes all UDP extension codes to be lost. The same warning message is applied if the Plan Length field value is changed from a ‘‘4’’ to a ‘‘5’’ or from a ‘‘5’’ to a ‘‘4.’’

Hardware and Software Requirements

UDP software is required.

User Identification Number

Description

Considerations

The UIN feature of the WW model AT&T MCU allows endpoints to be assigned a unique identification number when they register for service. Using that number in conjunction with a password allows them access to conferences to provide an added level of security.

Interactions

The UIN feature and passwords feature together provide added security for conferences.

Administration

The UIN security feature can be administered (turned on or off) on a per-conference basis.

Hardware and Software Requirements

See Table 4-1 for feature availability by model.

Video Conference Control

Description

Video conference control is performed using one or more of the following methods:

- Voice-activated switching control (sometimes referred to as automatic control)
- Chair control (part of user control)
- Presentation (part of advanced control)
- Broadcast with auto-scan (part of advanced control)

A detailed description for each is provided in the following sections.

Voice-Activated Switching

A speaker's endpoint becomes the broadcasting endpoint. The broadcasting endpoint sees the last broadcaster. AT&T MCU software and hardware provides for suppression of a ''ping-pong'' effect to avoid excessive switching during a rapid conversation. As an example, imagine a four-party video conference between endpoints named Site A, Site B, Site C, and Site D. Site B is the first endpoint to speak. When Site B finishes speaking, Site A begins to speak. As Site A begins, the voice-activated switching function displays Site A to the other three endpoints, while Site B is still displayed on Site A's screen as depicted in the following figure.

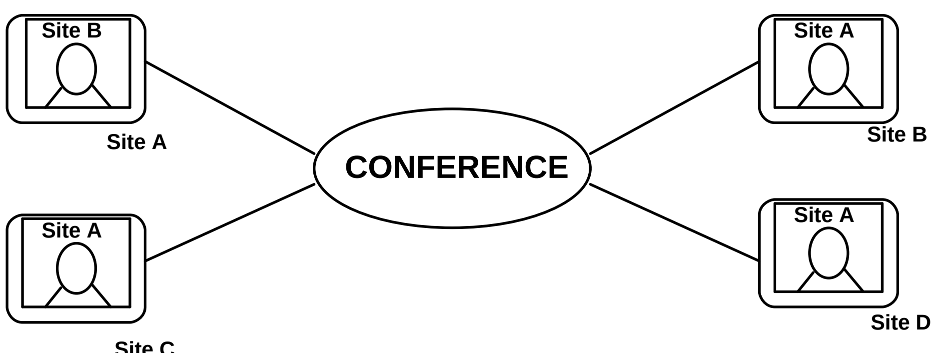


Figure 4-9. Previous Speaker=Site B

The following figure illustrates the scenario when the speaker at Site A relinquishes to the speaker at Site B. In this setting, Site B sees Site A (the previous speaker), while all the other endpoints see Site B (the current speaker).

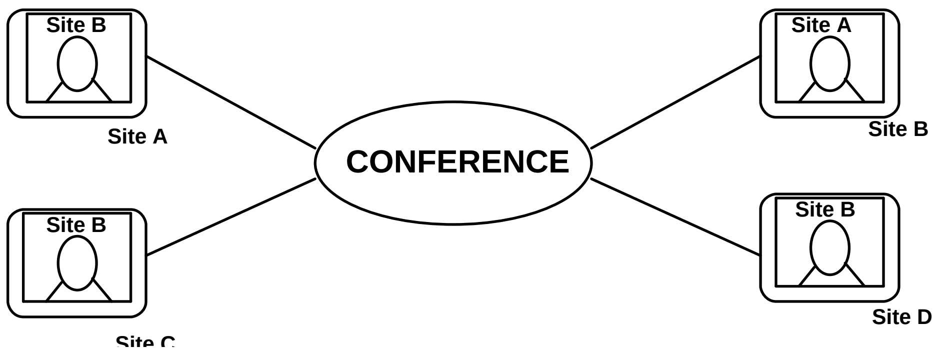


Figure 4-10. Previous Speaker=Site A

In the final voice activated switching example, Site D begins to speak after Site B finishes, as shown in the following figure. Site D sees Site B on their screen while all other endpoints see Site D.

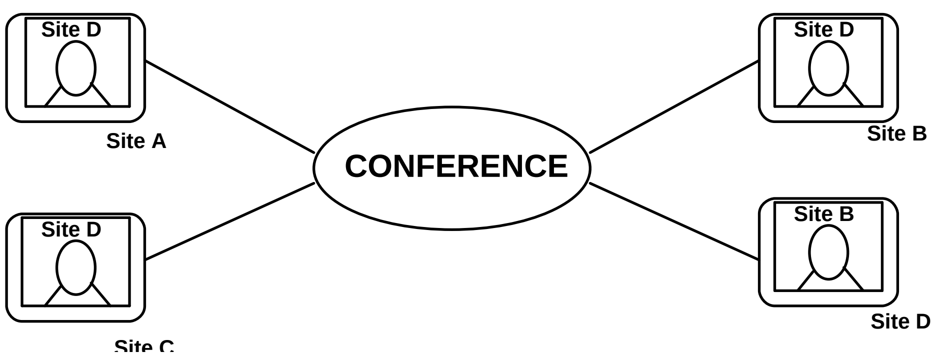


Figure 4-11. Previous Speaker=Site B

Contributor

This video function can be used for tasks such as sending high-resolution video still images to other endpoints. It allows a user to broadcast graphics to all the other conferees. To do so, the broadcasting endpoint uses a code to gain the broadcast control. After broadcasting, the endpoint sends a second code to relinquish control. This is sometimes called the ‘‘See Me’’ function.

Chair Control

This video function allows a conference chair to select which endpoint video is being broadcast to all the other endpoints. An endpoint with chair control can request and relinquish control and drop conferences. This function is not available on the DX model. The endpoint that claims ‘‘chair control’’ is the chair endpoint. In the following figure, Site A has the chair token and has selected Site C as the broadcast source while Site D is speaking. In this example, Site C (the broadcaster) is displayed at all endpoints except Site C. Site C sees Site D (the speaker).

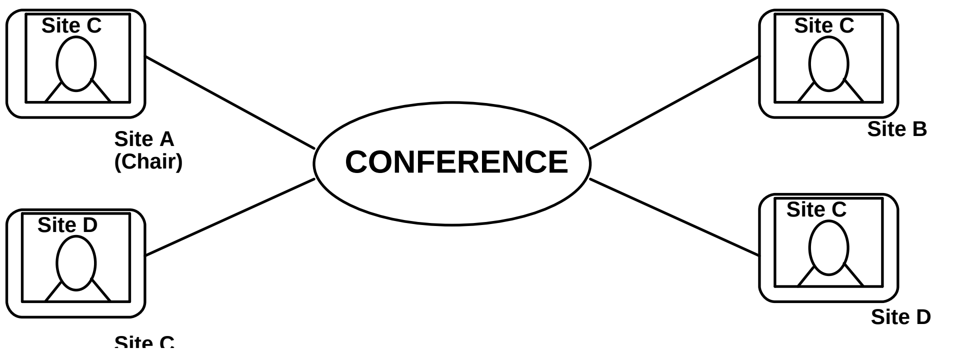


Figure 4-12. Broadcaster=Site C, Speaker=Site D

After a time, Site D finishes talking and Site C resumes talking. No change occurs to the video streams as shown in the following figure.

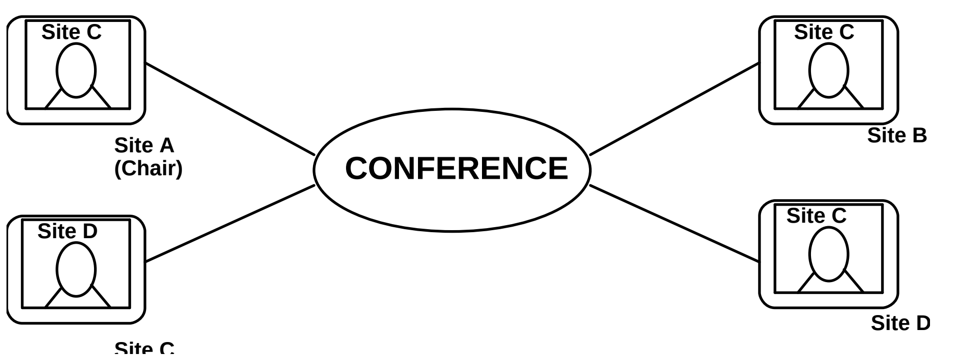


Figure 4-13. Broadcaster=Site C, Speaker=Site C

Site A begins to talk when Site C asks for responses. Now, as shown in the following figure, Site C sees Site A (the speaker) on their screen while all other endpoints continue to see Site C (the broadcaster).

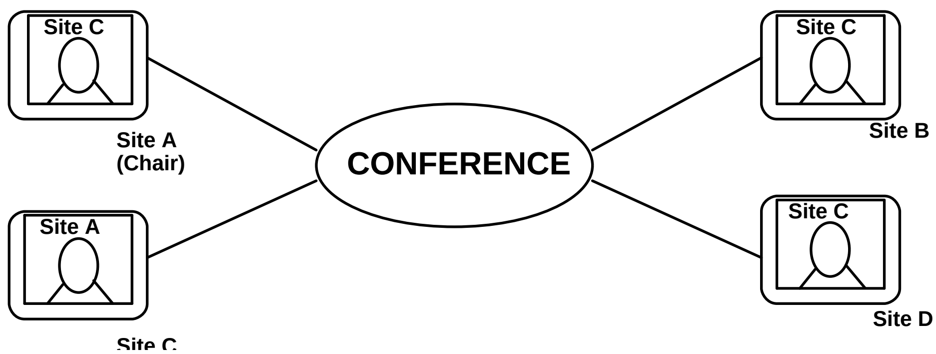


Figure 4-14. Broadcaster=Site C, Speaker=Site A

Presentation

This is an AT&T-provided function that does not require special endpoint support. It is a standard feature on the EX, GS, MS, VS, and WW models. It is an FX model option and not available on the DX or SX. This function configures the system to broadcast a single endpoint (*presenter*) to all other endpoints. The receiving endpoints are set to *voice-activated switching* so the broadcaster sees the endpoint selected by that function while all other endpoints see the broadcaster. One endpoint is administered as the broadcaster on the AT&T MCU prior to the conference. In the following figure, Site A is administered as the broadcaster while Site D is speaking. In this case, Site A (the broadcaster) sees Site D (the speaker) while all other endpoints see Site A.

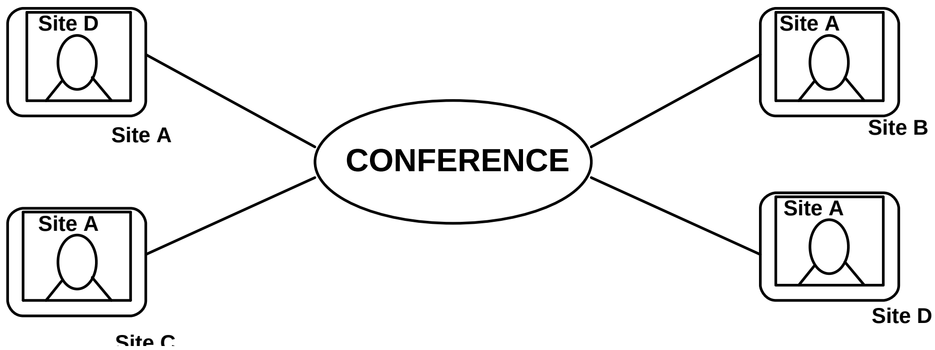


Figure 4-15. Speaker=Site D

The following figure illustrates what happens when Site D stops speaking and Site C begins. Site A remains the broadcaster and sees Site C (the speaker) while all other endpoints still see Site A.

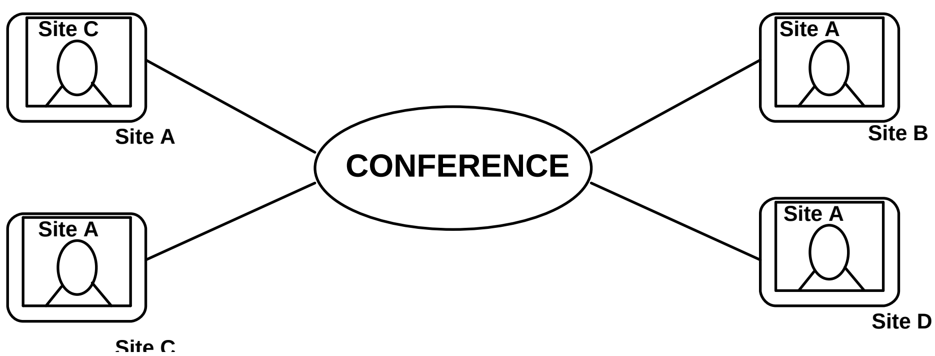


Figure 4-16. Speaker=Site C

Broadcast with Autoscan

This is an AT&T-provided function that does not require special endpoint support. Broadcast with auto-scan provides for a single endpoint to broadcast audio and video to all other endpoints. The broadcasting endpoint does not receive audio but sees video from each of the endpoints one at a time. The interval each endpoint is displayed is administrable on the AT&T MCU. In the following figure, Site A was identified as the broadcast endpoint when the conference was scheduled. The scanning period was also set during conference reservation. Site A views each endpoint in turn for the scanning period one at a time until the conference ends. All other endpoints see and hear Site A the entire time.

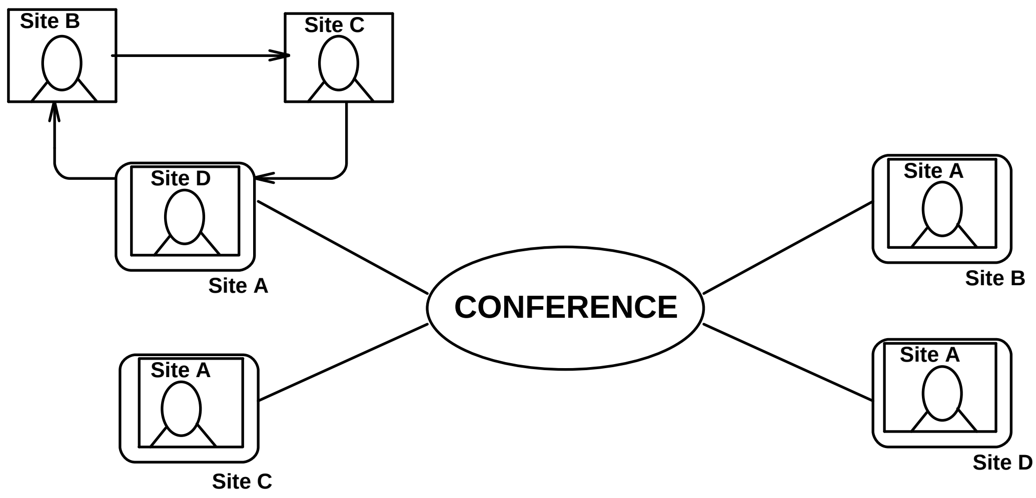


Figure 4-17. Broadcast with Autoscan

Considerations

As described in the detailed descriptions of video conference methods.

Interactions

None.

Administration

None.

Hardware and Software Requirements

See Table 4-1 for feature availability by model.

WorldWorx Data Compliance

Description

In a WorldWorx MCU, endpoints are expected to announce their ability to conform to WorldWorx data sharing. This is a slightly different approach from the application compliance flag, which identifies endpoints from within the MCU rather than having endpoints identify themselves. If endpoints do not identify themselves as WorldWorx data-compliant, they are not allowed to use the MCS/MLP data channel.

Administration

None.

Interactions

None.

Considerations

If an endpoint does not identify itself as WorldWorx data-compliant, the MCU opens an MCS/MLP channel but does not allow that channel to be used.

WorldWorx Service Indicator

Description

This feature provides the WorldWorx look and feel to endpoints in a conference. If this indicator is set, the MCU notifies endpoints through a special BAS code that they are connecting with a WorldWorx MCU.

Administration

None.

Interactions

None.

Considerations

None.

Power and Fans

5

The power and fans discussion in this chapter is divided into the following major sections:

- Facility power sources
- Multicarrier cabinet (MCC) power systems
- Single-carrier cabinet (SCC) and enhanced SCC (ESCC) power systems
- Lightning protection
- Sneak current protection
- Cabinet fan units

NOTE:
The information in this chapter that refers to the SCC is true of the ESCC as well unless specified otherwise.

The power and cooling information in this chapter can be used to plan for AT&T MCU installation.

Power Sources

An AT&T MCU can be configured for either AC or DC operation. Refer to the power source information that pertains to your configuration in the following sections.

AC Power Sources

Power feeders from a dedicated AC power source (usually located outside the building where the system is installed) are connected to an AC load. The feeders do not power other equipment. The AC load distributes the power to receptacles. The power cord from the AC power distribution unit in each multicarrier cabinet and AC power supply in each single-carrier cabinet is plugged into a receptacle.

Either of the following AC power sources is acceptable for powering the AC load:

- Single-phase 240VAC 60Hz that supplies 120VAC or 240VAC, which is shown in Figure 5-1.

  This source has three wires plus ground: two hot wires, one neutral wire, and one ground wire. A hot wire has a voltage with respect to the neutral wire.

- Three-phase, Y, 208VAC 60Hz that supplies 120VAC or 208VAC, which is shown in Figure 5-2.

  This source has four wires plus ground: three hot wires, one neutral wire, and one ground wire.

Figure 5-1. Single-Phase 240VAC Source

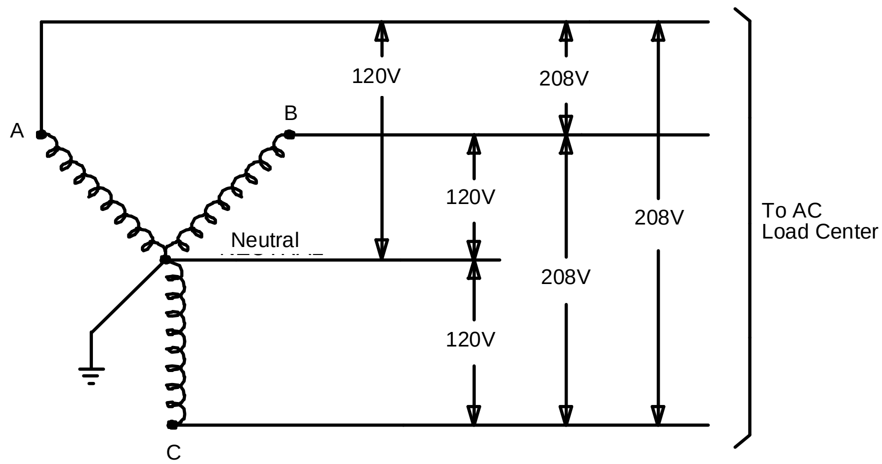


Figure 5-2. Three-Phase Y 208VAC Source

Table 5-1 summarizes the 60 Hz AC power sources and power units that can supply power to an AC load. A National Electrical Manufacturers Association (NEMA) receptacle (identified by ''R'' in the table) is connected to the wires from the AC load. The AC power cord from the power input of the distribution unit or power supply is plugged into a receptacle.

Table 5-1. AC Power Sources Used for MCC and SCC Systems

Unit Type	Power Input	Power Sources
AC power distribution (J58890CE-2-L9)	120VAC at a NEMA5-50R	One phase lead of 240VAC, one phase of 208VAC from Three-Phase Y 208VAC source for 208VAC input, or 240VAC from Single-Phase 240VAC source for 240VAC input
AC power distribution (J58890CE-2-L10)	208/240VAC at a NEMA L14-30R	One phase of 208VAC for 208VAC input, or single-phase, 240VAC for 240VAC input
AC power supply (WP-91153-L3) in a single carrier cabinet	120VAC at a NEMA 5-20R	One phase lead of 240VAC, or one phase lead of three-phase, Y, 208VAC

DC Power

DC-powered cabinets require a -42.5VDC to -52.5VDC source at up to 75A. All non-US installations of MCCs require this DC source. In addition, DC-powered systems require an opto-islotator (106005242) for the AC power components of the system.

Fused Current Drains and Power Cord Plugs

Table 5-2 lists fused current drains and power cord plugs of AC-powered cabinets.

Table 5-2. Fused Current Drains and Power Cord Plugs of AC-Powered Cabinets

Cabinet	Fused Current Drain (Amps)	Power Cord Plug
MCC (120VAC)	50	NEMA 5-50P
MCC (208/240VAC)	30	NEMA L14-30P
ESCC (120VAC)	20	NEMA 5-20P
SCC (120VAC)	20	NEMA 5-20P

Table 5-3 lists fused current drains of DC-powered cabinets.

Table 5-3. Fused Current Drains of DC-Powered Cabinets

Cabinet	Fused Current Drain (Amps)
Multicarrier cabinet	75
Single-carrier cabinet	25

Multicarrier Cabinet Power System

An MCC power system consists of the following parts:

- AC or DC power distribution unit in the bottom (G position) of each cabinet, and cabling that distributes AC and DC output voltages to power unit circuit packs in the carriers. Non-US installations use only the DC power distribution unit.
- Power unit circuit packs in the carriers, which supply DC power to the circuit pack slots. The "Circuit Packs" on page 3-9 in Chapter 3 describes the power unit circuit packs.

  Table 5-4 lists the power inputs and power outputs of each power distribution unit that can reside in an MCC. A power cord with a NEMA plug is attached to each unit.

  In the table, the AC power input voltage wires are in the power cord going to the unit.

Table 5-4. Power Distribution Unit Inputs and Outputs in MCCs

Unit Type	Power Input	Power Outputs
AC power distribution (J58890CE-2-L9)	120VAC, 60Hz, 30A, three wires: one hot, one neutral, one ground, and a NEMA5-50P plug	120VAC (normal), 144VDC (optional, emergency), and 75VAC to 100VAC at 20Hz from the ring generator
AC power distribution (J58890CE-2-L10)	208/240VAC, 60Hz, 20A, four wires: two hot, one neutral, one ground, and a NEMA L14-30R plug	Same as above
DC power distribution (J58890CF-2-L9)	-48VDC at up to 45A	-48VDC, and 75VAC to 100VAC at 20Hz from the ring generator

Table 5-5 lists the input and output voltages of power unit circuit packs in the carriers of multicarrier cabinets.

Table 5-5. Carrier Power Unit Inputs and Outputs in Multicarrier Cabinets

	Power Inputs			DC Power Outputs		
Unit Type	**120 VAC**	**144 VDC**	**-48 VDC**	**+5VDC at 60A**	**-5VDC at 6A**	**-48VDC at 8A**
AC 631DA1	Yes	Yes	No	Yes	No	No
AC 631DB1	Yes	Yes	No	No	Yes	Yes
DC 644A	No	No	Yes	Yes	No	No
DC 645B	No	No	Yes	No	Yes	Yes

Power Distribution in a Multicarrier Cabinet

Figure 5-3 shows AC power distribution in multicarrier cabinets. Five conductor power distribution cables on each side of the cabinet connect the power distribution unit to the power unit circuit packs in each of the five carriers. The cables carry 120VAC during normal operation and 144VDC from optional batteries during power backup operation when AC power fails.

Another cable connects 120VAC from the power distribution unit to the battery charger.

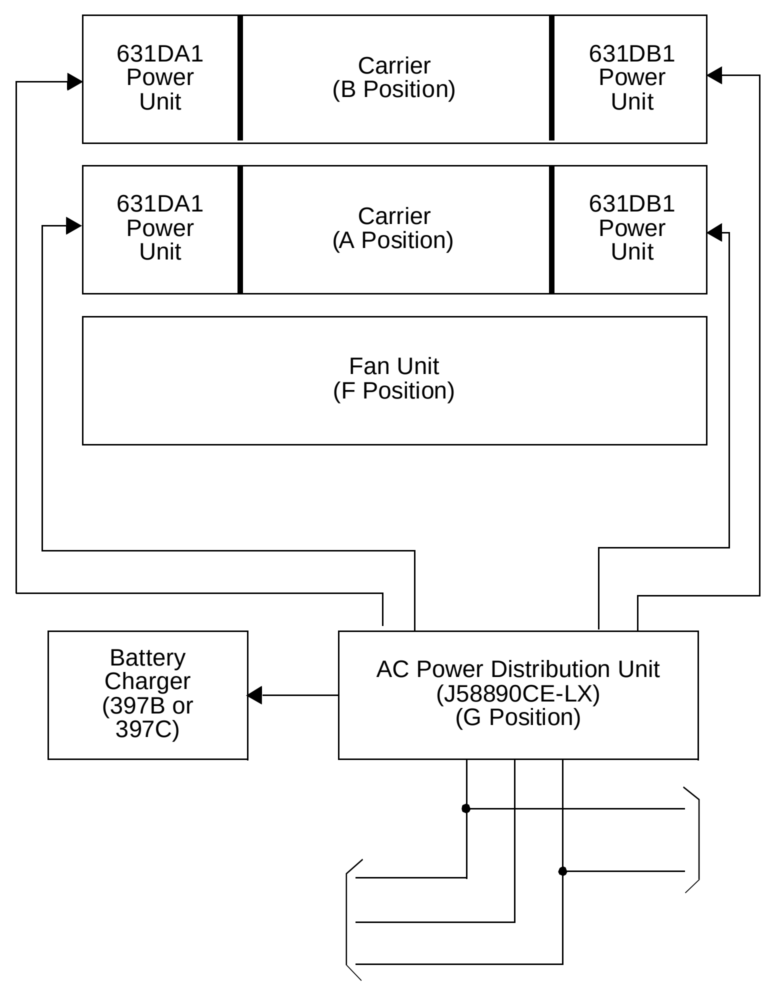


Figure 5-3. AC Power Distribution in Multicarrier Cabinets

NOTE:
The “LX” in the AC power distribution unit identifier refers to the L-number that identifies a specific unit type.

Figure 5-4 shows DC power distribution in multicarrier cabinets. Five conductor cables on each side connect the DC power distribution unit to the power unit circuit packs in the carriers.

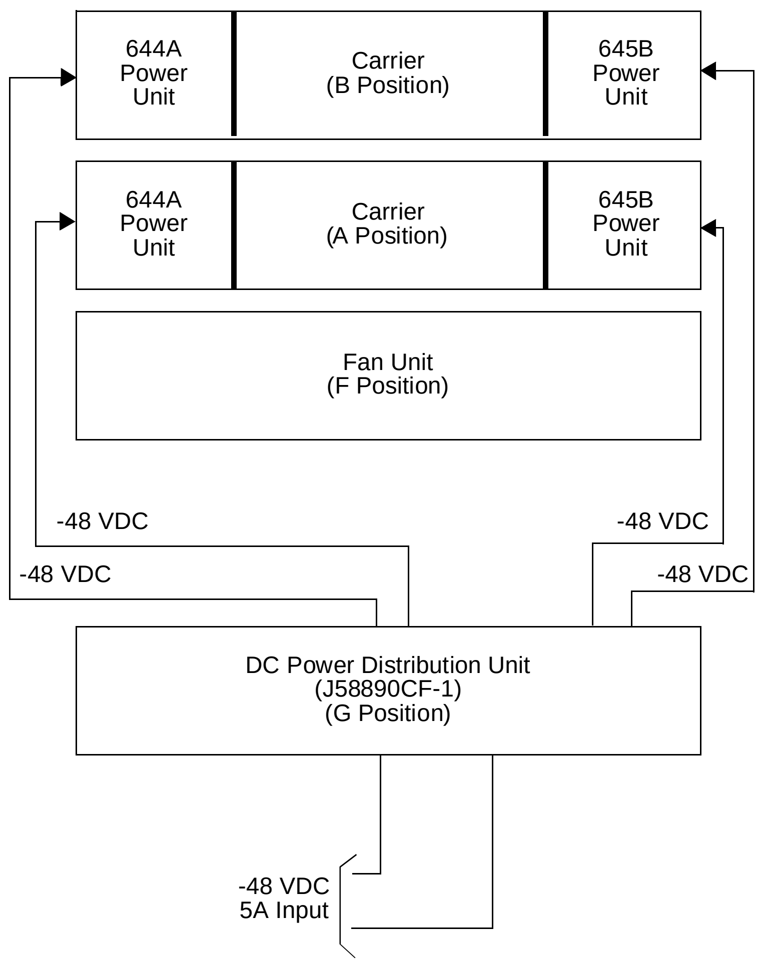


Figure 5-4. DC Power Distribution in Multicarrier Cabinets

AC Power Distribution Unit (J58890CE-1) in a Multicarrier Cabinet

Figure 5-5 shows the AC power distribution unit, which is located at the bottom of each multicarrier cabinet. Five conductor cables connect the unit to the power units in the carriers. These cables carry 120VAC to the carriers during normal operation and optional 144VDC during emergency operation when the AC input power has failed.

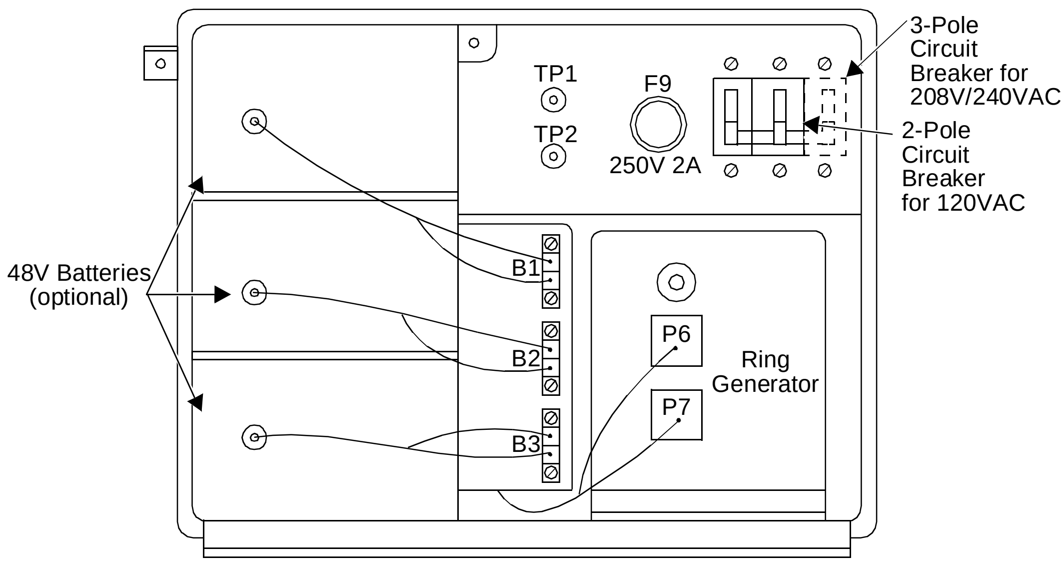


Figure 5-5. AC Power Distribution Unit (J58890CE-1) without Front Cover

The unit contains the following electrical components:

- Circuit breaker
- Three optional 48VDC batteries (KS-21906, L9), which are used only without an AT&T Uninterruptible Power Supply (UPS) powered cabinet and provide backup power to the cabinet
- DC power relay (used only without a UPS)
- Electromagnetic interference (EMI) filter
- Optional battery charger (397B or 397C power unit), used without AT&T UPS-powered cabinets
- Ring generator (124B2)
- 20-A fuses

Circuit Breaker

The circuit breaker protects the AC input power to the cabinet and serves as the main AC input disconnect switch. The circuit breaker has two poles for 120VAC or three poles for 208VAC/240VAC. When the cabinet overheats, the circuit breaker automatically opens and removes the AC power input.

48VDC Batteries

The three 48VDC batteries (used only without an AT&T UPS) are connected in series to produce nominally 144VDC, which is fused at 20A. The batteries are trickle-charged from the battery charger.

Battery Charger

When AC power is restored after an outage, the battery charger (used only without an AT&T UPS) converts a 120VAC input to DC voltage that recharges the batteries. The charger should charge the batteries within 30 hours.

DC Power Relay

This relay disconnects the batteries (used only without an AT&T UPS) from a system when AC power is being used. This relay also disconnects the batteries if power fails for more than 10 minutes. This protects the batteries from being over-discharged.

EMI Filters

The EMI filters suppress noise voltage on the AC input line to the unit.

Ring Generator

The ring generator converts the -48VDC input to a 75VAC to 100VAC, 20Hz AC output. The analog line circuit packs use this AC voltage output to ring voice terminals. The AC outputs are routed from the ring generator to port carriers, expansion control carriers, and control carriers.

20-A Fuses

20-A fuses protect the power on each cable going from the AC power distribution unit to power units in the carriers.

Power Backup

When AC power fails, the three 48VDC batteries (used without a UPS) power the system for the following times:

- 10 seconds for a port carrier
- 10 minutes for the control carrier

Uninterruptible Power Supply

An external UPS (which has a longer backup time than holdover batteries) can replace the batteries and battery charger. A UPS is connected from the AC power source to a cabinet’s AC power cord. When AC power fails, the UPS senses the failure and then supplies its own AC power to the cabinet.

DC Power Distribution Unit (J58890CF-1) in a Multicarrier Cabinet

Figure 5-6 shows the DC power distribution unit, which is located at the bottom of the multicarrier cabinet. Five conductor cables connect the unit to the power units in the carriers. These cables carry -48VDC to the carriers. 20A circuit breakers protect the power on each cable.

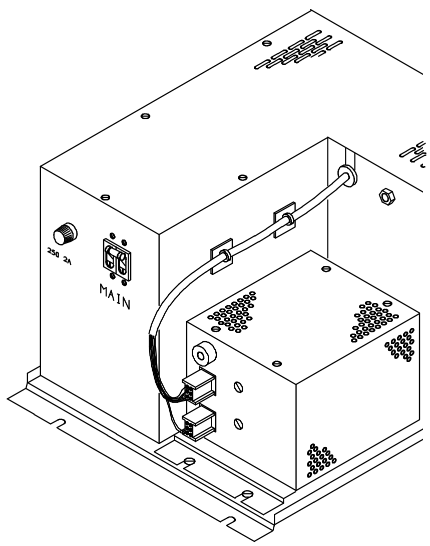


Figure 5-6. DC Power Distribution Unit (J58890CF-1)

The unit contains the following electrical components:

- Ring generator
- Filter circuits
- Circuit breakers
- Terminal blocks

Ring Generator

The ring generator converts the -48VDC input to a 75VAC to 100VAC, 20 Hz AC output. The analog line circuit packs use this AC voltage output to ring voice terminals. The AC outputs are routed from the ring generator to the port and expansion interface carriers and the control carrier.

Filter Capacitor Circuits

One filter capacitor circuit is in a list-5 unit.

Circuit Breakers

The main circuit breaker is located on the front of the unit and serves as the main DC input disconnect switch. When the cabinet overheats, the circuit breaker automatically opens and removes the DC power input.

The circuit breakers that control power to the carriers and filter circuits are located at the rear of the unit. In the CF list-5, five circuit breakers control the power units in the carriers. One circuit breaker controls power to the filter circuits.

Terminal Blocks

Terminal blocks on the rear of the unit are used to connect -48VDC from the DC power source.

AC Power and Ground Wiring for Multicarrier Cabinets

Figure 5-7 shows typical AC power and ground wiring for an MCC AT&T MCU. A 6-gauge ground wire connects the cabinet ground block to the system single-point ground bar on the customer’s AC power panel.

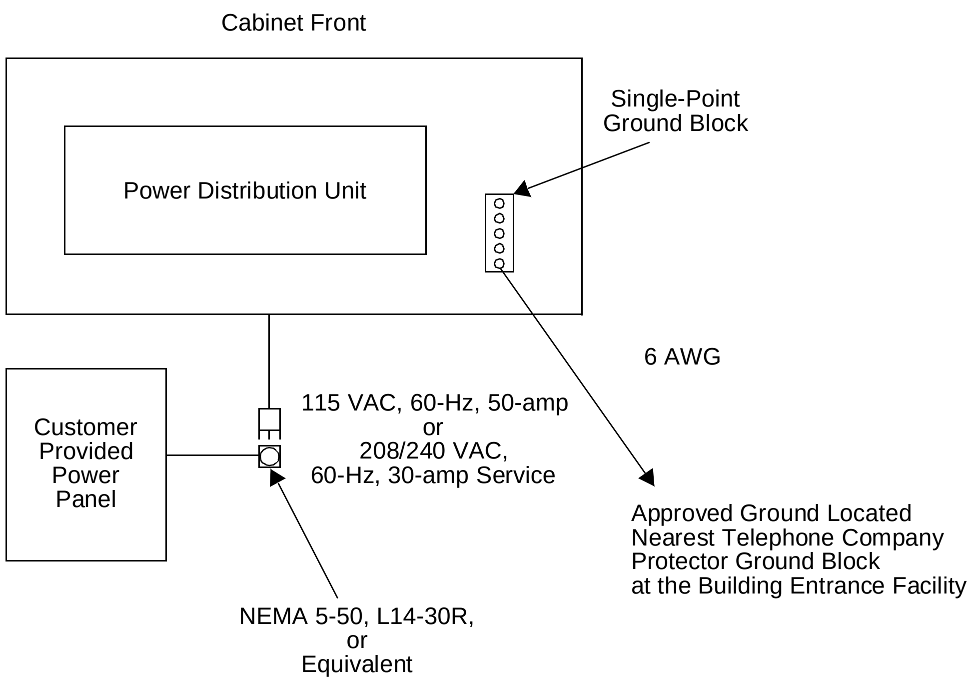


Figure 5-7. AC MCC Power and Ground Wiring

DC Power and Ground Wiring for Multicarrier Cabinets

Figure 5-8 shows power and ground wiring for a typical DC-powered multicarrier cabinet. The power and ground leads are routed through ductwork or underneath the cabinets.

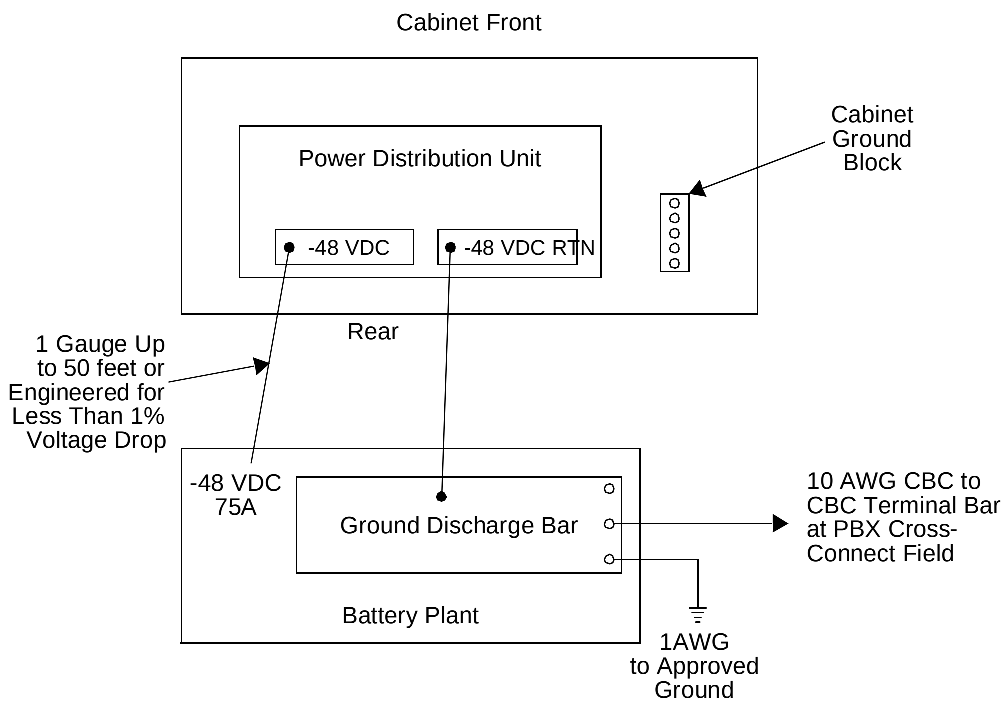


Figure 5-8. DC MCC Power and Ground Wiring

Single-Carrier Cabinet Power System

Each single-carrier cabinet has one AC or one DC power supply, which distributes DC power and ringing voltage to the circuit pack slots in the cabinet.

AC Power Supply (WP-91153) in a Single-Carrier Cabinet

The WP-91153 is a circuit pack used in the single-carrier cabinets of an AC powered system. See Chapter 3, "Components" for details.

DC Power Supply (676B) in a Single-Carrier Cabinet

The 676B is a circuit pack used in the single-carrier cabinets of a DC powered system. See Chapter 3, "Components" for details.

Uninterruptible Power Supply

An external UPS (which has a longer backup time than holdover batteries) can replace the batteries. The UPS is connected from the AC power source to the cabinet's AC power cord. When AC power fails, the UPS senses the failure and supplies its own AC power to the cabinet.

DC Power and Ground Wiring for Single-Carrier Cabinets

Each cabinet requires a separate DC power input. Figure 5-9 shows typical DC-powered and grounded single carrier cabinets. A ground wire is connected to the ground block in the bottom cabinet. The wire is routed to the battery plant where it is terminated on the ground discharge bar. An approved external ground is terminated on the ground discharge bar. An optional third cabinet can be connected via the next available receptacle in the distribution unit. Additional cabinets can be connected with additional power receptacles as the ones shown.

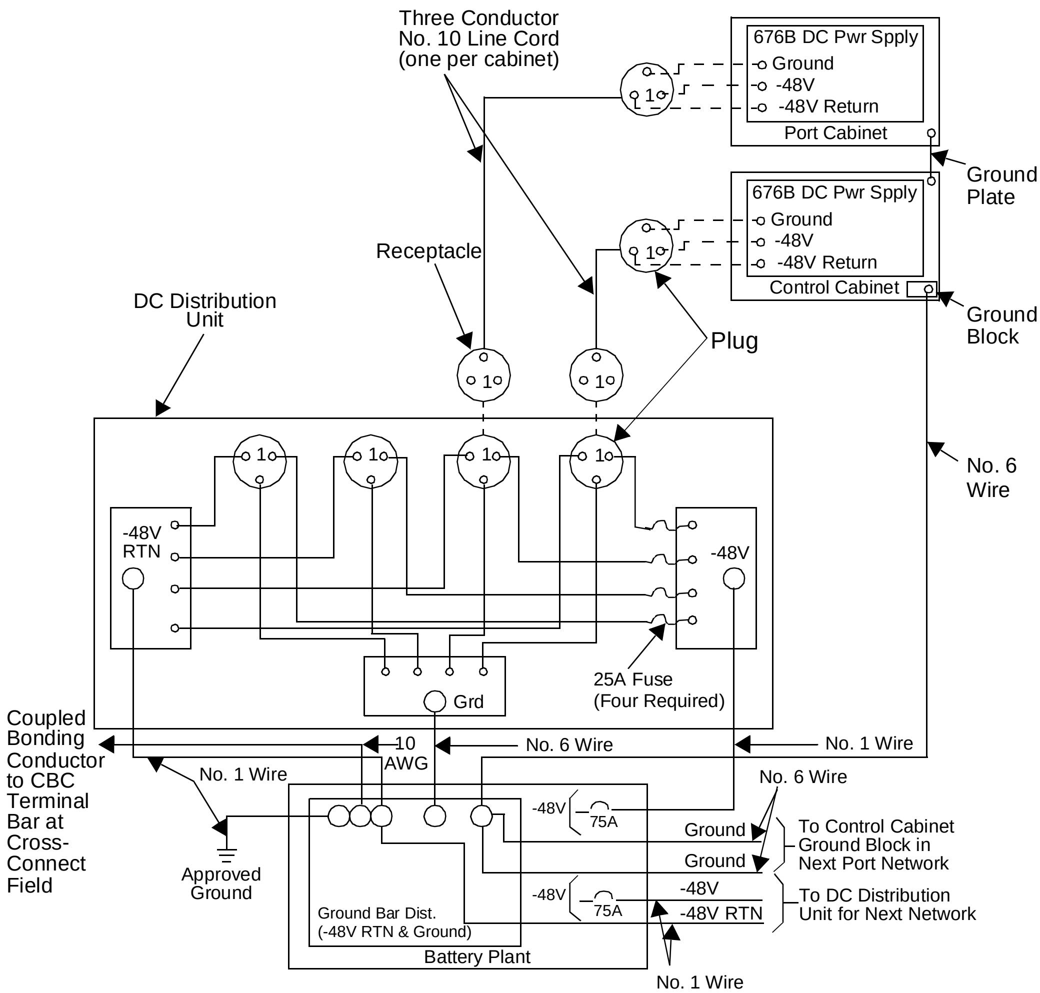


Figure 5-9. Typical SCC DC Power and Ground Wiring

Lightning Protection

A coupled bonding conductor (CBC) in multicarrier cabinet and single-carrier cabinet ground wiring protects the system from lightning. A CBC runs adjacent to wires in a cable and causes mutual coupling between the CBC and the wires. The mutual coupling reduces the potential differences that result from lightning surges. A CBC can be one of the following:

- 10-AWG ground wire
- Continuous cable sheath surrounding wires within a cable

- Six unused pairs of wire within a cable, which are twisted and soldered together

The CBC is connected from the cabinet single-point ground bar in an AC-powered cabinet or the ground discharge bar in a DC-powered cabinet to the CBC terminal bar at the PBX cross-connect field.

Each convenience outlet is fused at 5A. A dedicated AT&T MCU management terminal (MCU-MT) is usually plugged into one of the convenience outlets.

Figure 5-10 shows CBC grounding in an AC-powered cabinet. A minimum one-foot spacing is maintained between CBC and other power and grounding leads. The system single point ground terminal block is located on the AC load or AC protector cabinet.

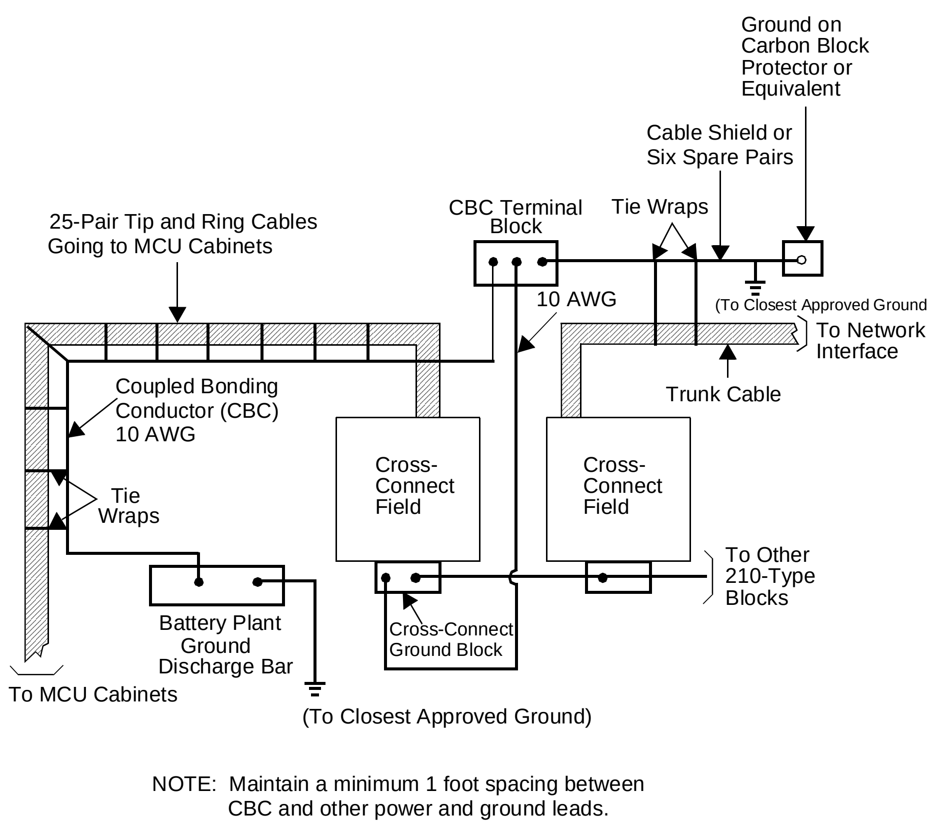


Figure 5-10. CBC Grounding in an AC-Powered Cabinet

Sneak Current Protection

Sneak fuses protect the building wiring and circuit packs from ‘‘foreign potential’’ by providing a current interruption capability. Sneak fuse panels, when provided, are installed on the switch side of the network interface. All incoming and outgoing trunks and off-premises station lines pass through the sneak fuses. Sneak current protection is required for installations in Canada. Sneak fuses must be CSA-certified.

Cabinet Fan Units

The fan units for MCC and SCC or ESCC cabinets provide for cooling the circuits in the respective cabinet. Refer to the section that pertains to your cabinet type.

Multicarrier Cabinet Fan Unit (ED-67077-30)

Figure 5-11 shows a cross-sectional side view of the fan unit in a multicarrier cabinet. The figure shows: fan and air filter positions; air flow directions from the front and rear fans.

The fan unit number is ED-67077-30, G5.

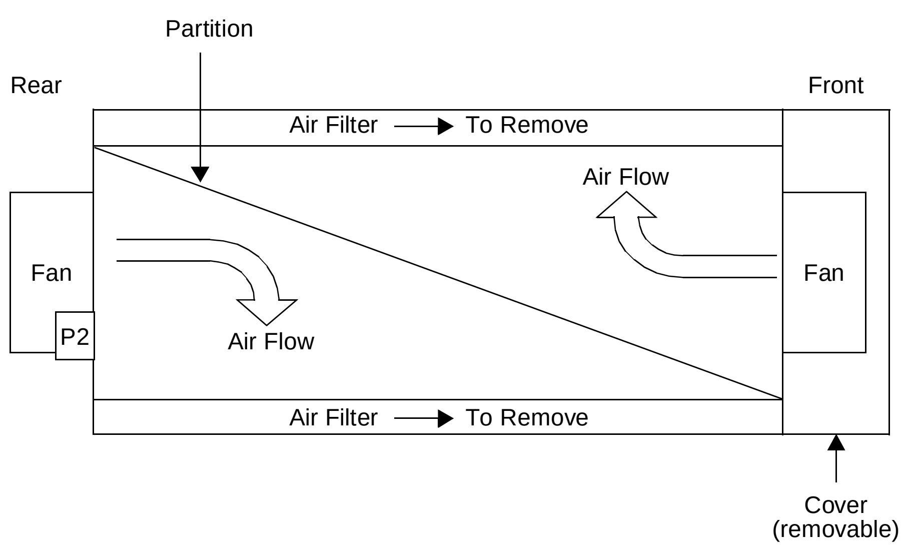


Figure 5-11. Fan Unit (ED-67077-30 G5) in Multicarrier Cabinet

The fan unit is mounted in cabinet ‘‘F’’ position and consists of the following equipment:

- Fans — six (three in the front and three in the rear) that operate at three speeds
- Air filters (403326820) — (removable foam type) one filter is located above the fan unit and one filter below it
- Thermistor sensors — four that monitor the cabinet temperature. Three sensors are inside the top of the cabinet and one sensor is inside the bottom of the cabinet. The top sensors affect the speeds of the front fans and temperature alarms, while the bottom sensor affects the speed of the rear fans.
- A speed control and thermal alarm circuit in each fan monitors the thermal sensors. When a sensor indicates a change in cabinet temperature, the circuit in a fan changes that fan’s speed accordingly.
- A P2 connector on the lower right rear side of the unit, which connects the fans to a power cable that supplies:
  - -48VDC to each fan
  - +5VDC to the speed control and thermal alarm circuit in each fan
  - Temperature sensor signals to the each fan. One pair of wires goes to each fan circuit.
  - Alarm signals from each fan. One pair of wires goes to each fan circuit.

The power cable branches from the cabinet harness on the right rear side of the cabinet.

The fans receive -48VDC from the 631DB1 AC power unit located on the front right side of the ‘‘A’’ position carrier.

Each fan circuit sends a major or minor alarm to the processor circuit pack for the following reasons:

- Reduced air flow in the cabinet

A minor alarm is sent when there is a fan failure. A major alarm is sent if the exhaust temperature reaches 1490F (650C). If the exhaust temperature reaches 1580F (700 C), the system shuts down.

Single-Carrier Cabinet Fan Unit

The SCC and ESCC fan unit (406411272) contains four fans mounted at the top rear of the cabinet. An air filter (405302159) is located below the fan unit. Air flows down through the filter over the circuit packs. This filter can be removed and cleaned or replaced when the cabinet door is removed. If the cabinet temperature reaches 1580F (700C), the temperature sensor in the power supply causes the system to shut down.

System Connections

6

The AT&T MultiPoint Control Unit (MCU) supports only digital connections. Figure 6-1 illustrates the logical trunk, network, and peripheral connections of a typical AT&T MCU system.

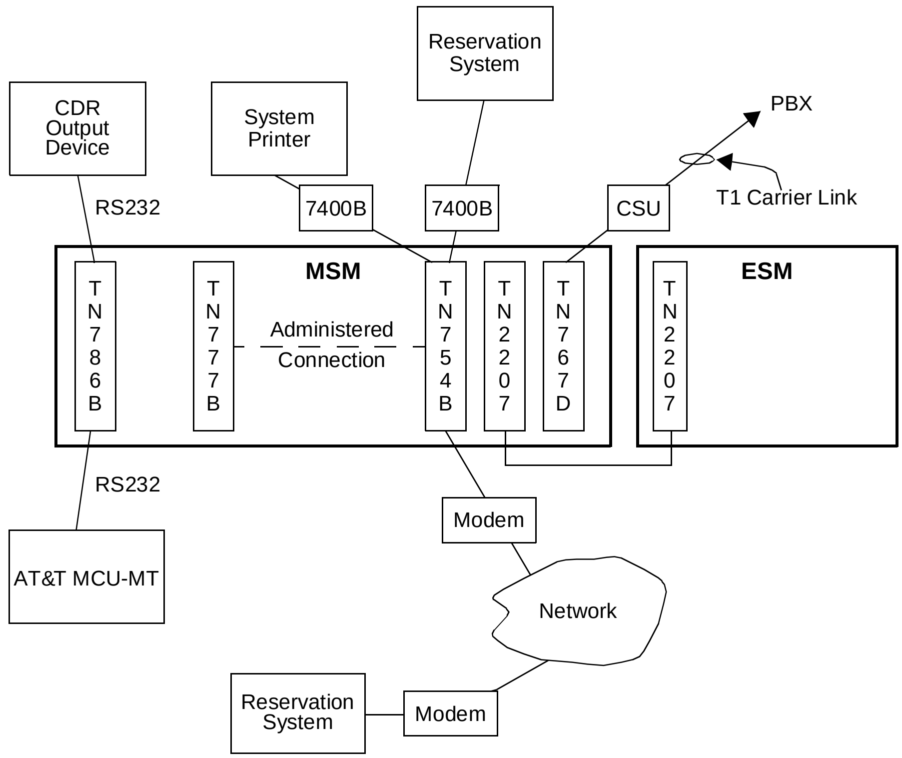


Figure 6-1. Logical Connections of a Typical AT&T MCU System (WW Model)

The following sections describe the AT&T MCU connection requirements.

Connection to the WorldWorx PBX

The AT&T MCU connects directly to the WorldWorx PBX via a PRI connection with NFAS.

CBC Application

CBC service selection allows the AT&T MCU to select, on a call-by-call basis, the services and/or features for a particular call. AT&T MCU-supported CBC provides ACCUNET and SDDN services integrated on a single transmission facility with flexible assignment of trunks to services.

See Chapter 4, "Feature Descriptions" for more information on CBC service.

NFAS Application

NFAS is a method that allows multiple T1 links to share a single D-channel on one of the spans to form an ISDN-PRI. In this manner, one T1 link is configured with 23 B-channels and one D-channel, while the other spans are configured for 24 B-channels each.

One weakness in NFAS is if the D-channel fails, all the facilities that share that single channel are no longer operational. For that reason, the AT&T MCU supports *D-channel backup,* which keeps a second D-channel in *standby* ready to take over for the shared D-channel if it fails.

See Chapter 4, "Feature Descriptions" for more information on NFAS.

Peripheral Connections

The AT&T MCU provides for connections to peripherals including adjuncts. Adjuncts are optional external components not residing in the system that provide administrative and application functions. The following peripherals and adjuncts are discussed in the following sections:

- Data communications equipment (DCE) and data terminal equipment (DTE)
- Terminals
- Printers
- Administration adjuncts

DCE Connections

DCE includes devices such as data modules that are connected between the system and DTE. DCE does the following:

- Provides analog-to-digital and digital-to-digital interfaces between the system and DTE
- Converts protocols between the system and DTE, and isolates the system electrically from DTE

Data modules link DTE with a system’s digital communications protocol (DCP) digital ports. The system can be connected to 7400B data modules, which provide full-duplex asynchronous voice and data connectivity in DCP applications.

DTE Connections

DTE uses EIA RS232C or DCP interfaces. DTE for the AT&T MCU includes the following equipment types:

- AT&T MCU-MTs
- Printers (optional)

Management, Scheduling, and CRS Terminals

The AT&T MCU-MT and AT&T MCU-ST are 715 BCS terminals. The following terminals can be added to the system as additional AT&T MCU-STs or PC remote scheduling terminals:

- 4410
- 4425
- 513 BCT
- Personal computer with CRS software

The AT&T MCU-MT is connected to the system via the TN786B processor board. AT&T MCU-STs or PC remote scheduling terminals can be connected locally using a 7400B with a modem to the TN754B/TN777B pair. They can be connected remotely (over the public or private network) using a modem on both sides of the network and a null modem cable to the TN754B/TN777B pair.

Printers

The following printers can be connected to a system:

- System printer: models 443, 450, 460, 470, 475, 476, 477, 478, 479, 495, 5310, and 5320
- CDR printers

The system printer or CDR printer can be interfaced to the system via either the TN754B/TN777B pair (via a 7400B) or TN786B circuit pack. If a system printer is added to the system, the printer is interfaced to the TN754B/TN777B pair and the CDR device is connected via the TN786B.

Call Detail Recording Output Devices

The CDR interface collects call records produced by a system when calls are connected. Output is through an ASCII printer interface. Any output device compatible with a printer interface can be attached to the system for viewing, processing, or printing CDR records.

A CDR device can be connected to the system through either the TN786B or TN754B circuit pack. If a system printer is present, the CDR output device is connected via the TN786B.

Administered Connections

The AT&T MCU provides data communications by means of ‘‘administered connection’’ and access endpoint features. An administered connection is a connection between the system and a peripheral that is enabled by an entry from a terminal.

The administered connection feature automatically establishes an end-to-end connection between two access (data) endpoints. For example, administered connections are used to control the Dial-Out feature.

For a detailed description of administered connections, refer to *AT&T Network and Data Connectivity*, 555-025-201.

Environmental Requirements

7

Use the information in this chapter to determine floor area and wall area required for the system and associated peripheral equipment installed in the equipment room. Also included are specifications for temperature, humidity, air purity, and lighting levels in the equipment room.

NOTE:
The main MCU cabinet is referred to as the Multimedia Server Module (MSM). The supplementary MCU cabinet that serves as the data module is referred to as the Expansion Services Module (ESM). For instructions on installing the ESM, see the ESM supplemental documentation.

Floor Area

Floor area requirements in the equipment room vary between multicarrier cabinets and single carrier cabinets. Refer to the section that pertains to your cabinet type.

Multicarrier Cabinets

The following system equipment and optional peripheral equipment occupies the following floor area in the equipment room:

- System Cabinet and Cable Slack Manager — The system cabinet is 32 inches (81 cm) wide and 28 inches (71 cm) deep. The cabinet is 70 inches (1.8 m) high. The cable slack manager requires 38 inches between the cabinet and wall. Each cabinet (including the door opening) and cable slack manager occupy about 22 square feet (2 square m) of floor area. Allow at least 36 inches in front of the cabinet for door removal.

Single-Carrier Cabinets

The following system equipment and optional peripheral equipment occupy the following floor area in the equipment room:

- System Cabinet and Cable Slack Manager — The system cabinet is 27 inches (69 cm) wide and 22 inches (56 cm) deep. A single cabinet is about 20 inches (51 cm) high, a 2-cabinet system is 39 inches (99 cm) high, a three-cabinet system is 58 inches (1.5 m) high, and a four-cabinet system is 77 inches (2 m) high. The cable slack manager requires 38 inches between the cabinet and wall. The system cabinets and cable slack manager occupy about 8 square feet (.74 square m) of floor area. Allow at least 36 inches in front of the cabinet for door removal.

Floor Plans

Floor plans of the system and peripheral equipment vary depending on the size and shape of the equipment room and the extent of growth planned for the system. The wall behind a system cabinet must be clear of all objects (pictures, shelves, or windows) that are not required in the system installation. The entire area behind a cabinet must be reserved for the cross-connect field and the cable access panel (when provided). Also, room for system growth should be considered.

Figures 7-1 and 7-2 show typical floor plans. Power outlets should be located outside the cross-connect field area and must not be under switch control or shared with other equipment. Processor port networks require a special 120 volt, 60Hz, 15 Amp or 20 Amp power outlet (NEMA 5-15 or NEMA 5-20 receptacle or equivalent). The system must be properly grounded as described in Chapter 5, "Power and Fans". The trunk field may be located in the cross-connect field. Each SCC cabinet uses 10-foot B25A cables from positions A and B. Each MCC cabinet uses 15-foot B25A cables from positions A and B.

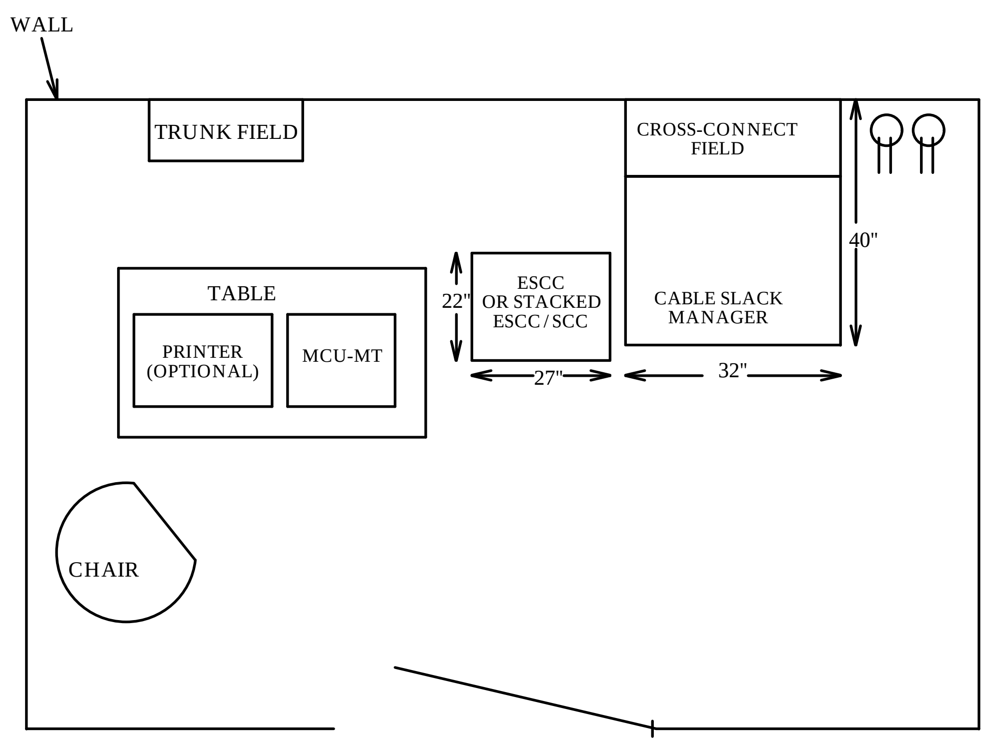


Figure 7-1. Typical SCC System Floor Plan

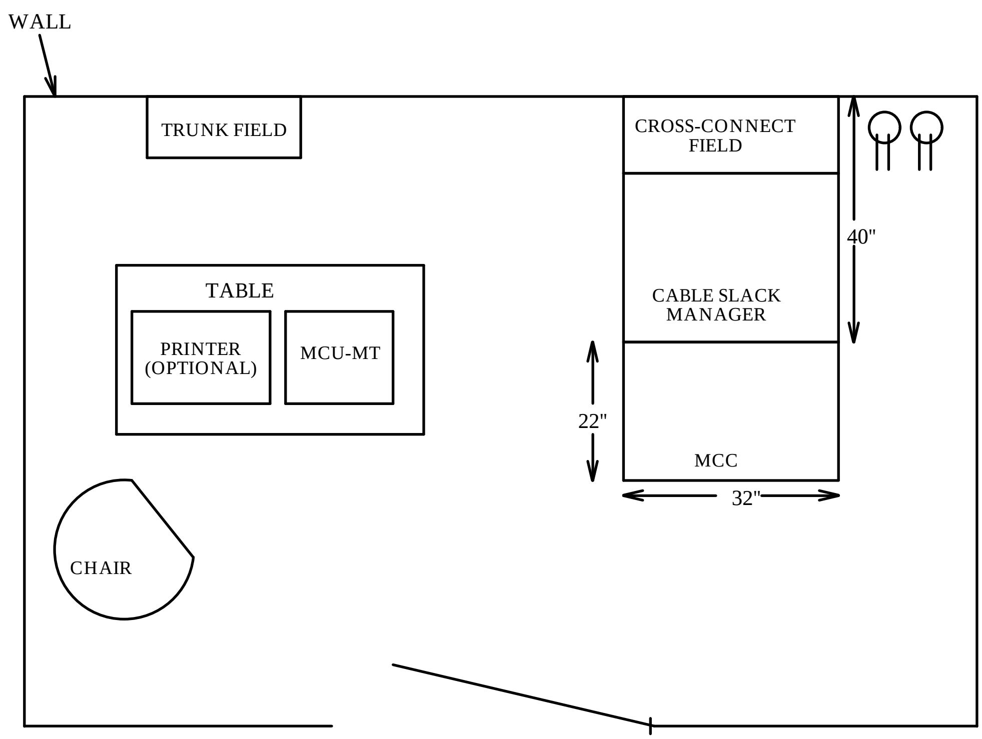
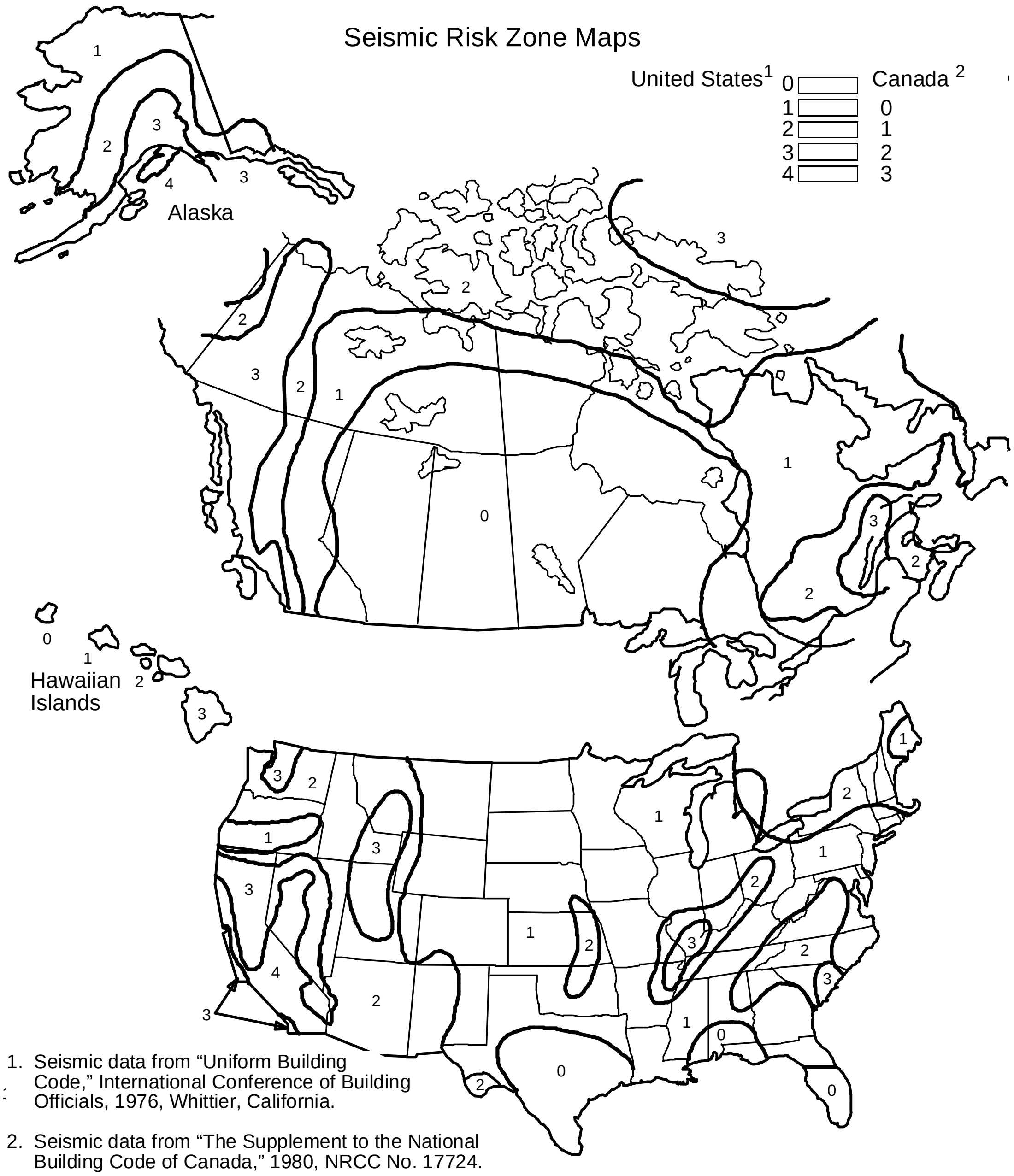


Figure 7-2. Typical MCC System Floor Plan

Earthquake Protection

When earthquake or disaster bracing is required by law or when local engineering feels that bracing is necessary, the cabinets can be bolted to the floor. Figure 7-3 shows US and Canadian earthquake zones where bracing may be needed. A greater susceptibility of an area to earthquakes is indicated by a higher number in the figure. In the United States, 0 represents the lowest susceptibility and 4 represents the highest. In Canada, 0 represents the lowest susceptibility and 3 represents the highest. An earthquake protection kit is available. See the *AT&T MultiPoint Control Unit Installation Quick Reference*, 555-027-723 for details.



1. Seismic data from “Uniform Building Code,” International Conference of Building Officials, 1976, Whittier, California.

2. Seismic data from “The Supplement to the National Building Code of Canada,” 1980, NRCC No. 17724.

Figure 7-3. United States and Canada Earthquake Environment

Desktop Area

The AT&T MCU-MT and AT&T MCU-ST terminals can be located in the equipment room and require area on a desk or table.

The terminals each require approximately 3.2 square feet (.3 square m) of area.

Optional Printers

The following AT&T printer documents include information on optional printers that require floor or desk area:

- 445 Printer 999-700-023
- 443 Printer 999-700-024
- 450 Printer 999-700-025
- 460 Printer 999-700-022
- 470 Printer and 475 Printer 999-300-285IS
- 572 Printer and 573 Printer 999-300-562

Wall Area

Wall area required in the equipment room depends on the type of cross-connect hardware installed, such as Z100-type (modular) or 110-type. The area required also depends on the size of the system.

If existing cross-connect hardware is reused, the space requirements and hardware requirements are detailed in the system floor plan.

Floor Loading

This section presents the floor-loading requirements for multicarrier cabinet and single-carrier cabinet systems.

Multicarrier Cabinets

The floor must have a commercial floor loading code of at least 50 pounds per square foot (242 kg per square meter). A fully loaded multicarrier cabinet weighs about 800 pounds (360 kg). Thus, a free maintenance area of at least 16 square feet (1.5 square m) is required for each cabinet.

Single-Carrier Cabinets

A single cabinet weighs about 130 pounds (59 kg), a fully loaded two-cabinet system weighs about 255 pounds (115 kg), a fully loaded three-cabinet system weighs about 380 pounds (171 kg), and a four-cabinet system weighs about 500 pounds (225 kg). The floor must have a commercial floor loading code of at least 50 pounds per square foot (242 kg per square meter).

Temperature and Humidity

The system equipment is installed in a well-ventilated area. Maximum equipment performance is obtained at an ambient temperature between 40 and 120 degrees Fahrenheit (4 and 49 degrees Celsius). The relative humidity range is as listed in Figure 7-1.

Table 7-1. Humidity Ranges by Temperature

Temperature	Acceptable Humidity
From 40 to 84 degrees Fahrenheit (4 to 29 degrees Celsius)	10 to 95 percent
Above 84 degrees Fahrenheit (29 degrees Celsius)	10 to 34 percent

Installations outside these limits may reduce system life or impede operation.

The system equipment can operate at the maximum short-term operational limits for a period not to exceed 72 consecutive hours or a total of more than 15 days in a year.

NOTE:
At altitudes above 5,000 feet (1,525m), the maximum short-term temperature limit is reduced by 1°F for each 1,000 feet (305m) of elevation above 5,000 feet (1,525m). At 10,000 feet (3,050m), for example, the maximum short-term temperature limit is 115°F.

Air Purity

The cabinet should not be installed in a place where the air may be contaminated by:

- Excessive dust, lint, carbon particles, paper fiber contaminants, or metallic contaminants
- Corrosive gases, such as sulfur and chlorine

Lighting

Lighting should be bright enough to allow personnel to perform their tasks properly. The recommended light intensity is 50 to 70 footcandles, which meets Occupational Safety and Health Act (OSHA) standards.

RF Noise

In most cases, noise is introduced into the system through trunk or station cables, or both. However, electromagnetic fields near the system control equipment may also cause noise in the system. Therefore, the system and cable runs should not be placed in areas where a high electromagnetic field strength exists. Radio transmitters (AM or FM), television stations, induction heaters, motors with commutators of 0.25 horsepower (187 watts) or greater, and similar equipment are leading causes of interference. Small tools with universal motors are generally not a problem when they operate on separate power lines. Motors without commutators generally do not cause interference.

Field strengths below 1.0 volt per meter are unlikely to cause interference. These weak fields can be measured by a tunable meter. Field strengths greater than 1.0 volt per meter can be measured with a broadband meter.

The field strength produced by radio transmitters can be estimated by dividing the square root of the emitted power in kilowatts by the distance from the antenna in kilometers. This yields the approximate field strength in volts per meter and is relatively accurate for distances greater than about half a wavelength (150 meters for a frequency of 1000 kHz).

Acoustic Noise Levels

Acoustic noise levels are given for multicarrier cabinets and single-carrier cabinets.

Multicarrier Cabinets

The noise produced by a system with a 5-Carrier Cabinet is 51, 53, and 56 dBA at low, medium, and high fan speeds, respectively, at a distance of five feet (1.5 m). If the system cabinet door is open, there is an additional 1 dBA of noise.

Single and Enhanced Single-Carrier Cabinets

The noise produced by the system is as follows at a distance of five feet (1.5 m):

- 1 cabinet—48 dBA
- 2 cabinets—50 dBA

If the system cabinet door is open, there is an additional 1 dBA of noise.

Technical Specifications

8

The sections in this chapter provide the technical specifications for AT&T MCU capabilities, performance, and feature capacities. The following specification are covered:

- Performance
- System hardware and software capacity limits
- Maximum port slot capacities
- Additional hardware to use features
- Allocation of station buttons
- Initialization and recovery
- Cabling distances
- DS1 remoting transmission distance
- Tones
- Indicator lamp signals
- Protocols
- Transmission characteristics
- Service codes
- Facility interface codes

Performance Specifications

Table 8-1 lists AT&T MCU response times.

Table 8-1. Response Time Listing

Attribute	Response Time
Call processing	Video switch time (three seconds)
System management	Four to six seconds mean response time
Maintenance	High-priority periodic tests must be completed within one hour. High-priority scheduled tests must be completed once each day, but not during the busy hour.
Booting and recovery	11 minutes

System Capacities

Table 8-2 lists the maximum capacities for AT&T MCU system software and hardware.

Table 8-2. System Capacities

Parameter	Capacity
Application Adjuncts	
Asynchronous links	5
CDR output device	1
System printer	1
Conference Reservation System (CRS)	1
ARS	
ARS patterns	254
Digits deleted for ARS	23
Digits inserted for ARS	36
Entries in HNPA and RHNPA tables	1,000
Inserted digit strings	450
FRLs	8
Trunk groups in an ARS pattern	6
Cabinets	
ESCC/SCC	3
MCC	1
Call appearances	
Maximum per extension	10
Minimum per extension	2
Call detail recording	
Number of CDRUs per system	1
Conferences	
Maximum simultaneous conferences per system	24
Data parameters	
Administered connections	128
Alphanumeric dialing	
Maximum number of entries	50
Characters per entry	22
Digital endpoints	24
Dial Plan	
Extensions	740
Number of characters in a name	15
Trunk dial access codes	
Number of access codes	97
Number of digits	1-3

Table 8-2. System Capacities — *Continued*

Parameter	Capacity
Port circuit pack slots	
Control carrier in an ESCC	14
Control carrier in a MCC	9
Port carrier (SCC or MCC)	14
NOTE: The number of port circuit packs is limited by the power supply and not by the number of slots.	
System administration	
Admin history file entries	250
Simultaneous administration commands	1
Simultaneous maintenance commands	1
Simultaneous SM sessions	3
Printer queue size	50
Time slots	
Total slots	512
Time slots for voice and date	483
Tone classifiers	
Tone detector boards	20
General-purpose tone detectors	40
Touch-tone receivers	80
TTR queue size	8
Trunks	
DS1/E1 circuit packs	25
Queue slots for trunks	128
PRI D-channels via PI	4
Total PRI D-channels	8
Trunk groups in system	64
Trunk members in a trunk group	99
Voice/Data terminals	
Total digital endpoints	24
Maximum number of button modules	8

Additional Feature-Required Hardware

Some features require specific additional hardware not included with a system that does not have the respective feature. Table 8-3 lists the hardware required for particular features.

Table 8-3. Additional Hardware Requirements by Feature

Feature	Required Hardware
Administered connections	■ TN754 digital line
Automatic circuit assurance	■ Maintenance alarm terminal with display
Automatic route selection in a private network	■ Additional TN748C or TN420C tone detectors ■ Possibly TN767D or TN2207
Bandwidth on demand interoperability group (BONDing)	■ A TN787D MMI to terminate BONDed calls
Call-by-call service selection	■ A TN767D or TN2207 DS1 interface for signaling link ■ As many as 23 ISDN-PRI trunk group members (99 with NFAS) ■ A TN768 or TN780 tone-clock ■ A processor interface
CRS	■ Customer must provide an appropriate PC
Data call setup	■ One TN754B digital line circuit pack port for each AT&T MCU management terminal (MCU-MT) and AT&T MCU scheduling terminal (MCU-ST)
Data-only off-premises extensions	■ TDM ■ One port on a TN754 digital line

Table 8-3. Additional Hardware Requirements by Feature

Feature	Required Hardware
DS1 tie trunk service	■ One TN767D or TN2207 DS1 interface ■ One TN768 or TN780 Tone Generator/Clock circuit pack for synchronization of the DS1 tie trunks
Report Scheduler/System Printer (asynchronous printer)	Any of the following: ■ EIA port on the Processor Circuit pack ■ MPDM or 7400A data module and a port on a TN754 digital line
Call Detail Recording	■ MPDM (with a TN754B Digital Line circuit pack) connected to a Call Detail Recording utility ■ MTDM (with a TN754B Digital Line circuit pack) connected to a host computer

Terminal Alarm Notification Buttons

When an alarm occurs, the green (status) lamp associated with the assigned button assumes a steady state.

The lamp can be turned off by pressing the button associated with the lighted alarm lamp. If the lamp is turned off, and the alarm has not been resolved by the time maintenance reschedules testing, the green (status) lamp resumes its steady state. The alarm types and the meaning for each type are described here:

NOTE:
The text of each button label appears in parentheses.

- Administered Connection Alarm (ac-alarm) lights if a locally administered connection (ADM-CONN) or a dial-out call has a Major, Minor, or Warning alarm active.
- DS1 Facility Alarm (ds1-alarm) — lights if a DS1-BD has an off-board Major, Minor, or Warning alarm active.

- Facility Access Alarm (trk-ac-alm) — lights when the facility access trunk test feature is used.
- Major Alarm (major-alrm) — lights if any Major alarm in the system is active.
- Minor Alarm (minor-alrm) — lights if any minor alarm in the system is active.
- PI Link Alarm (link-alarm) — lights if links (1 through 8) have a Major, Minor, or Warning alarm active.
- SMDR1-Alarm (smdr1-alrm) — lights if the Primary Link has a Major, Minor, or Warning alarm active.
- System Printer Alarm (pr-sys-alm) — lights if the System Printer (SYS-PRNT) has a Major, Minor, or Warning alarm active.
- Multimedia Interface Alarm (mmi-cp-alrm) — lights if the MMI-BD, MMI-PT, or MMI-LEV has an active Major, Minor, or Warning alarm.
- Voice Conditioner Alarm (vc-cp-alrm) — lights if the VC-BD, VC-DSPPT, or VC-LEV has an active Major, Minor, or Warning alarm.
- Warning Alarm (warn-alrm) — lights if any Warning alarm is active in the system.

Initialization and Recovery

The time needed to initialize a system or recover from a system reset depends on the system line size, features activated, trunks used, and adjuncts connected to the system. A system needs several minutes for initialization or recovery from a reset condition.

Cabling Distances

When the system layout is determined, maximum cabling distances to the system cabinet must be considered. Table 8-4 lists the allowable intrapremises cabling distances. In case of mixed wire sizes, the table columns for 26-gauge wire are used. These cabling distances are based on a minimum of -42.5 volt at the equipment connecting to the system.

Table 8-4. Allowable Intra-premises Cabling Distances

	24-Gauge Wire		26-Gauge Wire	
Equipment	**(0.5106 mm) Feet**	**(0.4049 mm) Meters**	**Feet**	**Meters**
7400B data module	5000	1524	4000	1219
Maintenance alarm terminals (7444D):				
Phantom powered	3400	1037	2200	670
Locally powered	5000	1524	4000	1219

Tones

The following sections discuss the tones used by the AT&T MCU.

Conference Progress Tones

The system can be administered to provide the optional conference tones. Table 8-5 lists the tones provided for conferencing.

Table 8-5. Conference Tones

Tone	Frequency	Pattern (ms)
Another party has joined the conference	400 Hz, 440 Hz, and 480 Hz	100 on for each then 700 off before next tones
A party has left the conference	480 Hz	500 on, at least 500 off
Ten minutes until end of conference	440 Hz	3000 on, at least 1000 off

Call-Progress Tones

Table 8-6 lists the call-progress tones generated by the AT&T MCU.

Table 8-6. Call-Progress Tones

Tone	Frequency	Pattern (ms)
Bridging warning tone*	440 Hz	1750 on, 12000 off, 650 on; repeated
Busy tone	480 Hz + 620 Hz	500 on, 500 off; repeated
Confirmation tone	350 Hz + 440 Hz	100 on, 100 off, 100 on, 100 off, 100 on filled by silence; not repeated followed by silence; not repeated
Continuous confirmation tone	350 Hz + 440 Hz	100 on, 100 off; repeated
Dial tone	350 Hz + 440 Hz	continuous
Intercept tone	440 Hz and 620 Hz	250 on (440Hz), 250 on (620Hz); repeated
Recall dial tone	350 Hz + 440 Hz	100 on, 100 off, 100 on, 100 off, 100 on, 100 off followed by continuous dial tone
Reorder tone	480 Hz + 620 Hz	250 on, 250 off; repeated

* Used with the Busy Verification and Executive Override features, and Service Observing feature when the warning tone is enabled.

Audible Ringing

The audible ringing tone generated by the AT&T MCU is 1200 ms when on and 4000 ms when off.

Indicator Lamp Signals

Table 8-7 lists the lamp signals generated by the system for the alarm terminal.

Table 8-7. Lamp Signals Generated by the System

Lamp Signal	Pattern (ms)
Dark	Off
Lighted	On
Flashing	500 on, 500 off; repeated
Fluttering	50 on, 50 off; repeated
Broken Flutter	5 cycles of 50 on, 50 off, followed by 500 off; repeated
Wink	350 on, 50 off; repeated

Protocols

Table 8-8 lists some of the protocols used in the system, with applications and maximum limitations.

Table 8-8. Protocols Used in the System

Protocols	Applications	Mamimum Data Rate	Maximum Distance
DCP	Digital switch to data endpoints	128k 3000 ft (915m) for voice	5000 ft (1524 m) for data
RS232C	PDM to AP Switch to administration terminal. PDM to host computer.	19.2k	50 ft (15.2 m)

The AT&T MCU supports the Px64 protocol suite as approved by the ITU-T in 1993 for point-to-point and multipoint operation. These include the standards described in Table 8-9.

Table 8-9. Px64 Protocol Standards

Standard	Description
H.221	Frame structure standard for videoconferencing
H.242	Videoconferencing standard specifying communications between audio-visual terminals
H.230	Frame synchronization control standard for videoconferencing
H.261	Standard for compressed-data syntax of encoded video in videoconferencing
H.231	Multipoint control unit standard for videoconferencing
H.243	Multipoint LSD standard for videoconferencing

In addition, the AT&T MCU supports H.261 Annex D still-image transmission. AT&T remains deeply involved in the development of this protocol and is working to standardize additional features.

Endpoint Compatibility

The AT&T MCU is designed to be comparable with any endpoint that adheres to the ITU-T Px64 standard. For a detailed list of known compatible endpoints, see Chapter 1, "Introducing the AT&T MCU".

Trunk Circuit Pack Specifications

Table 8-10 lists specifications of the trunk circuit packs.

Table 8-10. Trunk Circuit Pack Specifications

Trunk Type	Circuit Pack	Specifications
DS1 tie	TN767D or TN2207	Capacity: 24/32 trunks for robbed-bit data service. Speed: trunks at 56k/384K H0, one facility at 1.544 M Signaling:DS1 over 4-wire, and line-side signaling
ISDNPRI	TN767D or TN2207	Capacity: 24 trunks

Transmission Characteristics

The system transmission characteristics comply with the American National Standards Institute/Electronic Industries Association (ANSI/EIA) PBX standard RS464A (SP-1378A). Insertion loss in the system for trunk-to-trunk is 0 dB nominal. Table 8-11 lists the overload and crosstalk loss levels.

Table 8-11. Overload and Crosstalk

Overload level:	+3 dBm0
Crosstalk loss:	≥70 dB

The AT&T MCU limits the digital interface port-to-digital interface port echo path delay to ≤2 ms.

Service Codes

Service codes (for the US only) are issued by the Federal Communications Commission (FCC) to equipment manufacturers and registrants. These codes denote the type of registered terminal equipment and the protective characteristics of the premises wiring of the terminal equipment ports.

Private line service codes are as follows:

- 7.0Y — totally protected private communications (microwave) systems
- 7.0Z — partially protected private communications (microwave) systems
- 8.0X — port for ancillary equipment
- 9.0F — fully protected terminal equipment
- 9.0P — partially protected terminal equipment
- 9.0N — unprotected terminal equipment
- 9.0Y — totally protected terminal equipment

9 Maintenance and Support

The primary objective of maintenance in the system is to detect, report, and clear troubles as quickly as possible and with minimum disruption of normal service. Periodic tests, automatic software diagnostic programs, and fault detection hardware support the objective and allow most troubles to be traced to a circuit pack in the system.

The system hardware is maintained as a group of independent units that are separately replaceable. The units include circuit packs, power units, fans, and peripherals.

The two general categories within maintenance are: system-alarmed troubles and user-reported troubles. For alarmed troubles, both a remote maintenance facility (if provided), a local terminal, and any customer premises equipment (CPE) alarms are automatically alerted. Most alarms are also reported by lights on the circuit packs in the system.

The major part of maintenance is directed toward system-alarmed troubles. The system detects and reports most problems automatically. The system automatically retires alarms. After an alarmed trouble has been cleared, the system retests the previously faulty area. When the trouble is no longer detected, the alarm is removed. It is not necessary for personnel to retire alarms after a problem has been fixed. However, testing a fixed condition and manually retiring the alarm is faster than allowing the system to automatically retire the alarm.

Single Source for Help

For information on implementation, call your account representative.

Hardware Used for Maintenance

The following hardware is used in fault detection diagnosis and repair:

- Maintenance circuit located on the processor circuit pack functions as follows:
  - Sends alarm information to a terminal
  - Indicates system status by alarm lights
  - Monitors and controls the reset condition and operation of the processor element
  - Monitors and controls the power units
  - Provides direct access to a terminal
  - Provides an asynchronous modem that allows personnel to enter maintenance and administration commands at a remote terminal
  - Displays alarms remotely
- AT&T MCU-MT terminal provides a maintenance interface for personnel
- Multifunction alarm terminals — major, minor, and warning buttons can be administered
- Circuit pack lights indicate the following when lit:
  - Red (for alarm), which indicates the system has detected a fault in that circuit pack
  - Green (for test), which indicates the system is running tests on that circuit pack
  - Amber (for busy), which indicates that circuit pack is operating
- In-line error detection circuitry, which checks for correct operation

Tests

Maintenance tests are divided into two groups: periodic and demand. The periodic tests run automatically at fixed intervals on a specific schedule. The short tests run hourly, and the long tests run every 24 hours. Heavy call processing extends the interval of these tests.

Demand tests are run by the system when it detects a need for them or by personnel when required during troubleclearing activities. Demand tests include the periodic tests and others that are required only when trouble occurs. Some of the nonperiodic tests may be disruptive to system operation. Using a terminal, personnel can initiate the same tests that the system initiates, and the results are displayed on the terminal screen.

Procedures

If part of the system fails some of the periodic tests a preset number of times, the system automatically generates an alarm. This alarm alerts personnel that action is required to restore the system to a normal condition. The system supports three levels of alarms:

- Major alarms, which are failures that cause critical degradation of service and require immediate attention.
- Minor alarms, which are failures that cause marginal degradation of service while not rendering a crucial portion of the system inoperable. This condition requires action, but its consequences are not immediate. Problems that cause minor alarms might be impaired service in a few trunks or interference with one feature across the entire system.
- Warning alarms, which are failures that are localized and cause no noticeable degradation of service. Warning alarms are not reported to a remote location.

The system sends an alarm to any CPE device such as a light, an automatic dialer, a bell, or other CPE. The CPE alarm activation level field on the system parameters maintenance screen must be administered to indicate the alarm level (major, minor, warning, or none) that activates the CPE device. Some alarm levels are adjustable by the Set Options feature.

Error and Alarm Logs

A record of system errors is recorded on an error log, which can be displayed on a terminal. The log is useful for analyzing problems that have not caused an alarm or when alarms cannot be retired by replacement of hardware.

When errors result in alarms, the alarms are listed on another record called the alarm log. The alarm log can also be displayed on a terminal. If a number of alarms are active, the alarm log can be used to determine which alarms should be cleared first.

The alarm log and the error log list current unresolved conditions and past alarms and errors that provide a profile of system maintenance. Both logs are saved on flash ROM after a major system failure or restart.

Local and Remote Testing

A terminal connected directly to the system or a remote terminal can be used to do the following:

- Display error and alarm logs
- Test circuit packs
- Test system functions
- Turn off (busyout) and release system equipment
- Reset the system

Port Circuit Pack Replacement and Testing

A port circuit pack can be replaced without turning off power or interrupting service except in the area directly affected by the replacement. Verification tests are automatically run on the circuit pack when it is plugged in.

Reports

The traffic measurements and associated reports provide traffic monitoring and collection utilities for the AT&T MCU. Using the appropriate command(s), you can display or print traffic information collected by the AT&T MCU. For details on the Call Detail Recording (CDR) reports, see Chapter 4, "Feature Descriptions". The traffic reports are listed alphabetically by report title and their purposes are described in Table 9-1. See *AT&T MultiPoint Control Unit (MCU) System Administration and Reports*, 555-027-727 for details on system reports.

Table 9-1. Reports Description

Report Title	Description
ACA Measurements	Provides the audit trail list of short and long holding time calls that have been referred for having exceeded the established limits.
ACA Parameters	This report lists all trunk groups in the system and displays the current definitions (parameters) for long and short holding times.
ARS Measurements Selection	Allows you to select as many as 25 automatic route selection (ARS) patterns for measurement from the 640 patterns that are available.
ARS Pattern Measurements	Provides the usage measurements for each of the ARS patterns chosen on the ARS Measurements Selection form.
DS1 Facility Summary	Provides a summary listing of DS1 converter activity. This report is particularly useful for tracking synchronization errors that affect the stability of PX64 connections.
DS1 Facility Log	Lists the error information log for the DS1 converter.
Monitor System View1	Provides maintenance status, and last hour’s traffic data for hunt and trunk groups.
Monitor System View2	Provides maintenance status, and last hour’s traffic data for trunk groups.
Occupancy Busiest 3-Minute Intervals Measurements	Lists the busiest 3-minute interval measurements.
Occupancy Last-Hour Measurements	Provides a detailed view of the last-hour occupancy levels.
Performance Summary	Provides a summary of the peak-hour trunk blocking, automatic route selection traffic data, trunks out of service, and trunks not used.
Tone Receiver	Provides traffic data for the dual-tone multifrequency (DTMF), purpose receivers, and call classifiers.
Trunk Group Call-By-Call Measurements	Provides info on the last-hour traffic data for any specified call-by-call trunk group if the trunk group had an usage allocation plan administered for the last hour. This report can be used to monitor the trunk group and to determine if the usage allocation plan meets current needs.

Table 9-1. Reports Description — *Continued*

Report Title	Description
Trunk Group Performance	Provides both graphical and textual display of the peak-hour blocking for each trunk group.
Trunk Outage Measurements	Lists a maximum of five trunks (for each trunk group) that are out of measurements at the sample time.
Security Violations Summary	Provides a report that identifies the port-types from which invalid login attempts, invalid barrier code attempts, and invalid authorization code attempts are made.

Documents

For detailed maintenance information and procedures, refer to the optional *AT&T MultiPoint Control Unit (MCU) Maintenance*, 555-027-724.

Training

The AT&T MCU training includes the following:

- *AT&T MultiPoint Control Unit Technician Overview*, BO1069V
- *AT&T MultiPoint Control Unit (MCU) Installation Quick Reference*, 555-027-723

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.

American National Standards Institute
The principal national standard-forming body in the United States. ANSI sets international standards in many areas (for example, the Open Systems Interconnection Reference Model).

ANSI
See "American National Standards Institute."

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 MCU-MT
The local management terminal used for local administration and maintenance access to the system. The terminal used for local administration and maintenance access to the system. Also has access to the reservations agent's forms and commands.

AT&T MCU-ST
The Meeting Reservation and Control Terminal. This is the terminal used by the reservations agent to enter and view information on the AT&T MCU. What distinguishes this terminal from the AT&T MCU-MT used for other administration and maintenance tasks is that this terminal would probably be physically located on the reservation agent's desk and be dedicated to the agent's use. Other logins could use this terminal for other functions, however.

Audio Add-On (Integrated Audio Conferencing with Dynamic Echo Cancellation)
The ability to add a single nonPx64 endpoint as an audio-only endpoint in a conference.

Automatic Mode
A feature of the MCU that officers voice-activated conference mode.

B

Bandwidth
The amount of bandwidth allocated to a conference can be 112k, 128k, 384k, 768k, 1472k, 1536k, or 1920k.

B-channel
A bearer channel provided by the network from a video endpoint to the AT&T MCU.

BONDed/BONDing
An acronym for Bandwidth On Demand. A BONDed call is a collection of single 56k/64k calls that are aggregated at the destination endpoint for bandwidths as great as 2048k. The AT&T MCU supports BONDing bandwidths of 112k, 128k, 168k, 192k, 224k, 256k, 280k, 320k, 336k, and 384k.

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

Broadcast
The AT&T MCU is said to be "broadcasting" when it sends the same video/audio/data signal to two or more locations.

Broadcaster
A terminal is a "broadcaster" when its video/audio/data signal is being broadcast by the AT&T MCU to the other terminals on the conference.

C

Call Detail Recording (CDR)
A switch feature that utilizes software and hardware to record call data.

Cascading
Two or more AT&T MCUs connected to join two or more conferences.

CCITT
Consultative Committee for International Telephone and Telegraph (CCITT). See also ITU-T.

CDR
See "Call Detail Recording."

Chairperson Control
The video function that allows a conference chairperson to select which endpoint video is being broadcast to all the other endpoints. To use this function, the chairperson endpoint must conform to the ITU-T H.243 standard.

Contributor
The video function used for tasks such as sending high-resolution video still images to other endpoints. It allows a user to broadcast something to all the other conferees. To do so, the endpoints must support the ITU-T H.243 standard. This is sometimes called the "*See Me*" function.

Complete
A reserved conference has a status of complete when its stop time has passed and the last conferee has dropped from the conference.

Convener
The person who schedules a conference and is responsible for distributing the MCU numbers.

CRS
See AT&T Conference Reservation System.

D

DCP
See “Digital Communications Protocol.”

Dedicated
A conference call that reserves a specified number of AT&T MCU ports for a multipoint video conference at any time.

Digital Communications Protocol (DCP)
An AT&T proprietary protocol used to transmit both digitized voice.

Digital signal level 0 (DS0)
A single 64k voice channel in a T1 or E1 facility. It consists of eight bits in a T1 or E1 frame every 125 micro-seconds.

Digital signal level 1 (DS1)
A North American standard digital interface that operates at 1.544 Mb/second.

DS0
See “digital signal level 0.”

DS1
See “digital signal level 1.”

DX
An AT&T MCU model that is distinguishable by the number of available ports and the automatic conference mode.

Dynamic Resizing
The feature that allows a reservation agent to add or remove conference participants both before and during a conference.

E

E1
A digital transmission standard that carries traffic at the rate of 2.048 Mbps. The E1 facility is divided into 32 channels (DS0s) of 64k information. The channels are numbered from 0 to 31. Channel 0 is reserved for framing and synchronization information. Whenever a D-channel is present, it occupies channel 16.

Endpoint
A video CODEC, camera, personal computer, speakers, and other equipment for multimedia conferencing. There are two types of terminals: the conference controller and the conference participants.

Enhanced Single Carrier Cabinet
A structure designed to contain a control carrier. An SCC can be stacked on top of an ESCC to provide one port carrier and one control carrier.

ESCC
See “Enhanced Single Carrier Cabinet.”

EX
An AT&T MCU model that includes all the FX model capabilities with additional port capacities, additional standard features, and automatic control, user control, and advanced control conference modes.

Expansion Services Module (ESM)
A device consisting of its own industry-standard bus, commercially available microprocessors, and cards that interface to the MCU bus via an E1 connection.

F

Far-End
The video endpoint is sometimes referred to as being on the far-end of the connection.

FX
An AT&T MCU model that includes all the DX model capabilities with additional ports, additional standard features, and both automatic and advanced conference modes.

G

G.711
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
See LD-CELP.

H

H0
A one-channel 384k bandwidth call.

H11
A 1472k or 1536k bandwidth call.

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

H-Series
See Px64.

HDLC
See “High-Level Data Link Control.”

High-Level Data Link Control
An international bit-oriented protocol implemented at Level 2 of the Open System Interconnect (OSI) model.

I

Inactive
A conference status when a conference is reserved but has not begun.

Integrated Audio Conferencing with Dynamic Echo Cancellation (Audio Add-On)
The ability to add a single non-Px64 endpoint as an audio-only endpoint in a conference.

Integrated Services Digital Network Link-Access Procedure on the D-Channel (ISDN-LAPD)
A link-layer protocol on the ISDN-BRI and ISDN-PRI data-link layer (level 2). This protocol provides data transfer between two devices plus error and flow control on multiple logical links. It is used for signaling and low-speed packet data (X.25 and mode 3) on the signaling (D-) channel and for mode-3 data communications on a bearer (B-) channel.

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

ISDN
Integrated Services Digital Network, an open standard switching interface.

ITU-T
International Telecommunications Union-Telecommunications. Formerly known as the Consultative Committee for International Telephone and Telegraph (CCITT).

L

LAPD
See "Integrated Services Digital Network Link-Access Procedure on the D-Channel."

LD-CELP
Low-delay codebook excited linear prediction (LD-CELP). LD-CELP is a means of efficiently encoding audio in a smaller bandwidth (16k) than PCM while preserving a quality similar to PCM. This results in freeing up bandwidth for video, thus improving picture quality.

M

MCC
See "Multicarrier Cabinet."

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

MCU
See "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-MT
See “AT&T MCU-MT.”

MCU-ST
See “AT&T MCU-ST.”

MCU Number
A number that a conferee dials to join a multipoint video conference.

MSM
See “Multimedia Server Module.”

Multicarrier Cabinet
A structure that holds one control carrier and up to three port carriers.

Multimedia
Multimedia refers to the use of a variety of media, including audio, data, graphics, and full motion video.

Multimedia Server Module
Part of the MCU corresponding to the Processor Port Network (PPN) of the AT&T DEFINITY switch. The MSM and PPN contain the processor complex.

Multipoint
The multipoint conference involves two or more terminals.

Multipoint Control Unit
Equipment that provides high-quality multimedia conferencing with video endpoints that communicate via the ITUT-T Px64 standards.

Multirate Bandwidths
Bandwidths of one channel that use an ISDN-PRI facility from an endpoint that provides the appropriate size bandwidth. These bandwidths include 128k, 192k, 256k, 320k, 384k, 768k, 1536k, and 1920k.

N

Network Service
The telephone company that provides the trunks and MCU numbers to access the AT&T MCU.

Notification Tones
The entry, exit, and warning tones available with some of the AT&T MCU models.

O

OA&M
Operations, administration, and maintenance.

P

Paper-Based Scheduling System
A method of recording and tracking conference reservations to prevent overbooking of the AT&T MCU.

PC-Based Scheduling
A scheduling method that uses the AT&T Conference Reservation System to accept reservations and qualify video endpoints to participate in a multipoint video conference.

PCM
PCM is the standard means of encoding audio. It uses a large bandwidth and lessens the bandwidth available for video transmission, resulting in poorer picture quality.

Point-to-point
Two video endpoints communicating with each other.

Port
A port consists of the AT&T MCU resources necessary to support one conference endpoint. Therefore, a 24-port AT&T MCU can support 24 simultaneous conferees, although not necessarily on one conference.

Presentation
A video function that configures the system to broadcast a single endpoint (*presenter*) to all other endpoints. The broadcasting endpoints are set to *voice-activated switching* so the broadcaster sees the endpoint selected by that function, while all other endpoints see the broadcasting endpoint.

Presenter
The broadcasting endpoint in a video conference using the *presentation* function.

PRI
ISDN Primary Rate Interface, a trunk interface that provides 24 128k channels, one of which (D-channel) is used for signaling.

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. Endpoints conforming to Px64 are sometimes referred to as *H.230 terminals* or *320 terminals*.

R

RBS
See “Robbed-bit signaling.”

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

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

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.

SCC
See “Single Carrier Cabinet.”

SCM
See “Selected Communications Mode.”

Selected Communications Mode
Mode used by the MCU to configure multipoint conferences as endpoints are joined to the MCU. The SCM is used to represent the currently active collection of audio, video, data, and transfer modes and bit rate allocations provided to a conference. This collection may change over the course of a conference.

Status
The current state of a conference.

See Me
See *Contributor*.

Single Carrier Cabinet
A combined cabinet and carrier unit that contains one carrier. This structure is designed to contain a port carrier. An SCC can be stacked on top of an ESCC to provide one port carrier and one control carrier.

System Administrator
Person responsible for administration of the AT&T MCU’s trunks, dial plan, and time of day. The system administrator has access to the conference records but typically would not enter or modify these records, unless the system administrator is also performing the job of the *reservation agent*.

SX
Basic AT&T MCU model, which is available in two configurations: four multimedia ports, or six multimedia ports plus two Audio Add-On ports. The model supports one multimedia conference at a time at a 2B-channel, 112k bandwidth. Also known as the Preconfigured Single Conference MCU.

T

TDM
See “Time-division multiplexing.”

T1
A digital transmission standard primarily used in North America (and sometimes in Japan and Middle Eastern countries) that carries traffic at the digital signal level-1 (DS1) rate of 1.544 Mbps. The T1 facility is divided into 24 channels (DS0s) of 64k information. The channels are numbered

from 1 to 24. These 24 channels, with an overall digital rate of 1.536 Mbps, along with an 8k framing and synchronization channel, comprise the 1.544 Mbps transmission. Whenever a D-channel is present, it occupies channel 24.

Time-division multiplex (TDM) bus
A bus that is time-shared regularly by preallocating short time slots to each transmitter. In a PBX, all port circuits are connected to the TDM bus. This allows any port to send a signal to any other port.

Time-division multiplexing (TDM)
Multiplexing that divides a transmission channel into successive time slots.

Tones
See Notification Tones.

V

Video Endpoint
A video codes, with camera, speakers, screen and other equipment required for multimedia conference.

Voice-Activated Switching
A speaker’s endpoint becomes the broadcasting endpoint. The broadcasting endpoint sees the last broadcaster. AT&T MCU software and hardware provides for suppression of “ping-pong” effect to avoid excessive switching during a rapid conversation.

W

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

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