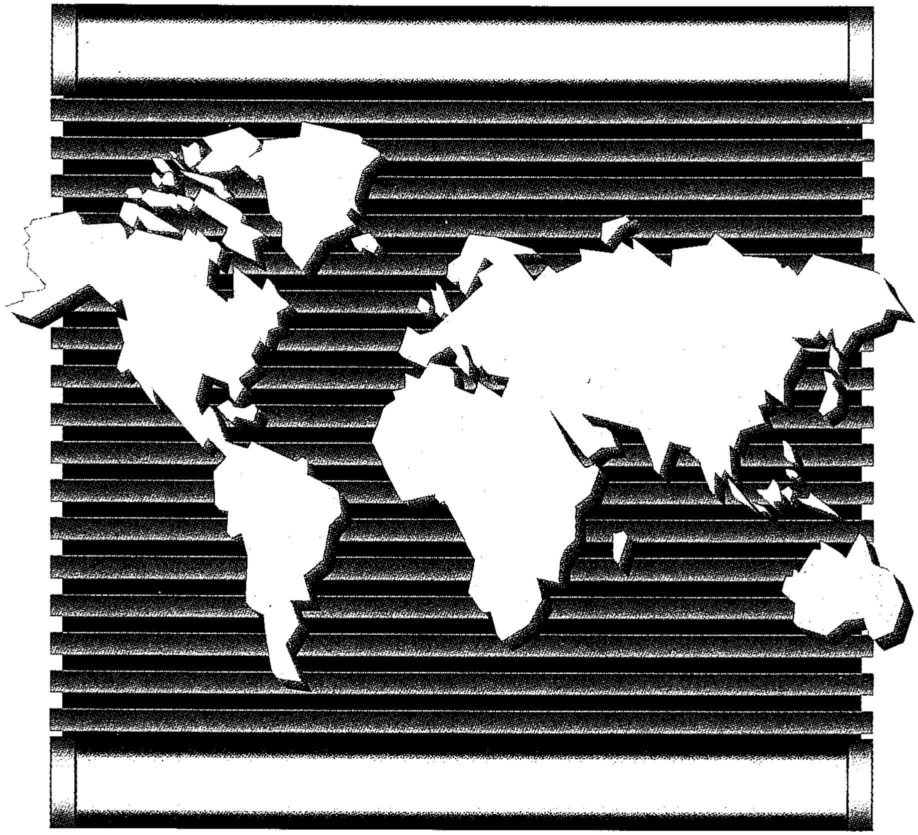


MITEL®

NOTICE TO U.S. CUSTOMERS

This SX-200 ML PABX equipment complies with Part 68 of the FCC rules. On the rear of this equipment is a label that contains, among other information, the FCC registration number and ringer equivalence number (REN) for this equipment. If requested, this information must be provided to the telephone company.

The FCC Registration Number for the PABX equipment is: BN285B-64724-MF-E or BN259C-16891-MF-E depending upon the product's country of manufacture (refer to the registration label of the product).

Port	FIC	SOC	REN	Jack
Loop Start	02LS2	N/A	1.0B	RJ21X
Ground Start	02GS2	N/A	1.0B	RJ21X
OPS	0L13B	9.0F	N/A	RJ21X
DID (Reverse Battery)	02RV2-T	AS.2	N/A	RJ21X
E&M, 2 Wire, Type I	TL11M	9.0F	N/A	RJ2EX
E&M, 4 Wire, Type I	TL31M	9.0F	N/A	RJ2GX
DS-1	04DU9-B	6.0N	N/A	N/A

The REN is useful to determine the quantity of devices you may connect to the telephone line. Excessive REN's on the telephone line may result in the devices not ringing in response to an incoming call. In most, but not all areas, the sum of the REN's of all devices connected to one line should not exceed five (5.0). To be certain of the number of devices you may connect to your line, as determined by the total REN's, contact the local telephone company.

If the PABX equipment causes harm to the telephone network, the telephone company will notify you in advance that temporary discontinuance of service may be required. But if advance notice isn't practical, the telephone company will notify the customer as soon as possible. Also, you will be advised of your right to file a complaint with the FCC if you believe it is necessary.

The telephone company may make changes in its facilities, equipment, operations or procedures that could affect the operation of the equipment. If this happens, the telephone company will provide advance notice in order for you to make necessary modifications in order to maintain uninterrupted service.

If the equipment is causing harm to the telephone network, the telephone company may request that you disconnect the equipment until the problem is resolved.

For information concerning repairs that can be made by the customer, refer to MITEL *Troubleshooting Practice* and the *General Maintenance Information Practice.* These documents are part of the MITEL Standard Practices (technical documentation) shipped with the equipment.

This equipment may not be used on public coin phone service provided by the telephone company. Connections to party lines service is subject to state tariffs. (Contact the state public utility commission, public service commission or corporation commission for information).

WHEN PROGRAMMING EMERGENCY NUMBERS AND (OR) MAKING TEST CALLS TO EMERGENCY NUMBERS:

1. Remain on the line and briefly explain to the dispatcher the reason for the call.
2. Perform such activities in the off-peak hours, such as early mornings or late evenings.

ALLOWING THIS EQUIPMENT TO BE OPERATED IN SUCH A MANNER AS TO NOT PROVIDE FOR PROPER ANSWER SUPERVISION IS A VIOLATION OF PART 68 OF THE FCC's RULES.

Proper Answer Supervision is when:

a) This equipment returns answer supervision signals to the PSTN when DID calls are:

- Answered by the called station
- Answered by the attendant
- Routed to a recorded announcement that can be administered by the CPE user
- Routed to a dial prompt.

b) This equipment returns answer supervision on all DID calls forwarded to the PSTN. Permissible exceptions are:

- A call is unanswered
- A busy tone is received
- A reorder tone is received

This equipment is capable of providing users access to interstate providers of operator services through the use of access codes. Modification of this equipment by call aggregators to block access dialing codes is a violation of the Telephone Operators Consumers Act of 1990.

NOTICE TO CANADIAN CUSTOMERS

NOTICE: The Industry Canada label identifies certified equipment. This certification means that the equipment meets certain telecommunications network protective, operational, and safety requirementsas prescribed in the appropriate Terminal Equipment Technical Requirements document(s). The Department does not guarantee the equipment will operate to the user's satisfaction.

Before installing this equipment, users should ensure that it is permissible to be connected to the facilities of the local telecommunications company. The equipment must also be installed using an approved method of connection. The customer should be aware that compliance with the above conditions may not prevent degradation of service in some situations.

Repairs to certified equipment should be coordinated by a representative designated by the supplier. Any repairs or alterations made by the user to this equipment, or equipment malfunctions, may give the telecommunications company cause to request the user to disconnect the equipment.

Users should ensure for their own protection that the electrical ground connections of the power utility, telephone lines, and internal metallic water pipe system, if present, are connected together. This precaution may be particularly important in rural areas.

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

NOTICE: The Ringer Equivalence Number (REN) assigned to each terminal device provides an indication of the maximum number of terminals allowed to be connected to a telephone interface. The termination on an interface may consist of any combination of devices subject only to the requirement that the sum of the Ringer Equivalence Numbers of all the devices does not exceed 5.

The Ringer Equivalence Number for the SX-200® ML PABX is 1.0B.

Printed in Canada

Industry Canada Notice

This Class A digital apparatus meets all requirements of the Canadian Interference-causing Equipment Regulations.

Federal Communications Commission (FCC) Notice

Note: This equipment has been tested and found to comply with the limits for a Class A digital device, pursuant to Part 15 of the FCC Rules. These limits are designed to provide reasonable protection against harmful interference when the equipment is operated in a commercial environment. This equipment generates, uses, and can radiate radio frequency energy and, if not installed and used in accordance with the instruction manual, may cause harmful interference to radio communications. Operation of this equipment in a residential area is likely to cause harmful interference in which case the user will be required to correct the interference at his own expense.

9109-098-501-NA Issue 1
March 1997

SX-200® ML PABX

LIGHTWARE™ 16 ML Practices Index

MITEL®

NOTICE

1 General

This section contains a list of practices for the SX-200® ML Private Automatic Branch Exchange with *SX-200* ML LIGHTWARE™ 16 software.

Documentation Index

1.1 The *SX-200* ML PABX documentation is contained in three volumes as follows:

Volume 1 - 9109-098-001-NA, contains system description, feature operation information, peripheral equipment descriptions, and engineering information pertaining to the system and its components.

Volume 2 - 9109-098-002-NA, provides installation and administration information which includes testing, data entry, troubleshooting, and maintenance information.

Volume 3 - 9109-098-003-NA, contains ARS and SMDR documentation as well as various feature and application package details.

Table 1-1 Practices Index

VOLUME 1		**9109-098-001-NA**
Practice Number	**Title**	**Issue**
9109-098-501-NA	Practices Index	Issue 1
9109-098-100-NA	General Description	Issue 1
9109-098-105-NA	Features Description	Issue 1
9109-098-126-NA	Peripheral Devices	Issue 1
9109-098-180-NA	Engineering Information	Issue 1

9109-098-100-NA Issue 1
March 1997

SX-200® ML PABX

General Description

MITEL®

NOTICE

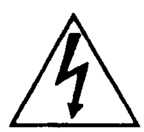
IMPORTANT SAFETY INSTRUCTIONS

WARNING: Failure to follow all instructions may result in improper equipment operation and/or the risk of electric shock.

- This product is to be installed and serviced only by qualified personnel.
- Read all instructions before attempting to install or use this product.
- Install all assemblies using the procedures described in this Practice.
- Configure this product with only the assemblies specified in this Practice.
- Grounding circuit continuity is vital for safe operation of telecommunication equipment. Never operate telecommunication equipment with grounding conductor disconnected. Ensure grounding conductor is installed before connecting telecommunication cabling to any system (see Note).

Note: It is recommended that all cabinets be unplugged from the ac mains during servicing. To reduce static susceptibility on a cabinet always attach the wrist strap from the cabinet being serviced, and immediately place any removed item into an antistatic bag.

EXPLANATION OF USE FOR SYMBOLS AND NOTICES



DANGEROUS VOLTAGE

The lightning flash with arrowhead symbol, within an equilateral triangle, is intended to alert the user to the presence of uninsulated "dangerous voltage" within the product's enclosure that may be of sufficient magnitude to constitute a significant risk of electric shock to persons.

INSTRUCTIONS

The exclamation point within an equilateral triangle is intended to alert the user to the presence of important operating and maintenance (servicing) instructions in the literature accompanying the product.

PROTECTIVE GROUNDING TERMINAL

The ground symbol within a circle identifies the terminal which is intended for connection to an external protective conductor. This connector must be connected to earth ground prior to making any other connections to the equipment.

DANGER	Danger indicates an imminently hazardous situation which, if not avoided, will result in death or serious injury.
WARNING	Warning indicates a potentially hazardous situation which, if not avoided, could result in death or serious injury.
CAUTION	Caution indicates a potentially hazardous situation which, if not avoided, may result in minor or moderate injury and/or damage to the equipment or property.

Table of Contents

List of Figures

List of Tables

1 Introduction

General

This practice provides an overview of hardware features, as well as a summary of environmental, electrical, and maintenance considerations for the SX-200® ML PABX. Refer to the following sections for more detailed descriptions of related information:

- *Features Description* Practice
- *Engineering Information* Practice
- *Installation Information* Practice

Reason for Issue

This issue describes the *SX-200* ML LIGHTWARE™ 16 software variant.

Disclaimer

The following products have been manufacture discontinued by Mitel. These products are supported but not described in *SX-200* ML Practices:

- SUPERSET 3™ and SUPERSET 4® telephone sets
- SUPERSET 3DN™ and SUPERSET 4DN™ telephone sets
- DATASET 1101 data cartridge
- SUPERSET™ DSS module.

The following products and peripheral devices are not supported on the *SX-200* ML PABX and are not described in *SX-200* ML Practices:

- Modem Interconnect Panel
- DATASET 1102 Rack-mounted Dataset
- DATASET 2102 Rack-mounted Dataset
- DATACABINET 9000 data cabinet
- DATASHELF 9100 datashelf
- ISDN Node
- Fiber Interface Module (and associated products)
- Peripheral Node
- LCD Console (and Console module for Universal Card).

2 System Overview

General

The *SX-200* ML PABX is a microprocessor-controlled telephone system that handles both voice and data switching. The system hardware is electrically compatible with most:

- single line telephones,
- MITEL®proprietary sets,
- key telephone systems,
- PABX telephone systems, and
- central office exchanges.

The following subsections outline the hardware configuration and the software version. Refer to Section 3 of this practice for additional information about the system hardware, and to Section 4 for an overview of the software structure. Table 2-1 describes physical characteristics of the PABX. Figure 2-1 shows the *SX-200* ML PABX.

Physical Description

The system cabinet has a steel frame and is approximately 22.9 cm wide x 48 cm high x 42.0 cm deep (9 inches x 19 inches x 16.5 inches). The internal structure of the cabinet is designed to hold the system cards and components. The front panel is plastic and can be removed with a screwdriver, allowing access to the cards.

The cabinet holds one Main Control Card II (MCC II), one Bay Control Card, one Bay Power Supply, and up to eight Peripheral Interface Cards. Located on the rear of the cabinet is a connector panel for printer and maintenance ports, and the SFT (system fail transfer) control port.

The system consists of one cabinet providing up to 96 ports.

System Fail Transfer

System Fail Transfer (SFT) (or power fail transfer) is provided by the SFT control port, allowing preselected DTMF or rotary telephones to be connected directly to CO trunks in the event of system failure in the PABX. To provide system fail transfer, a SFT unit must be connected to the system.

Maintenance Connector

2.1 A connector for a remote maintenance terminal is provided on the rear panel of the cabinet.

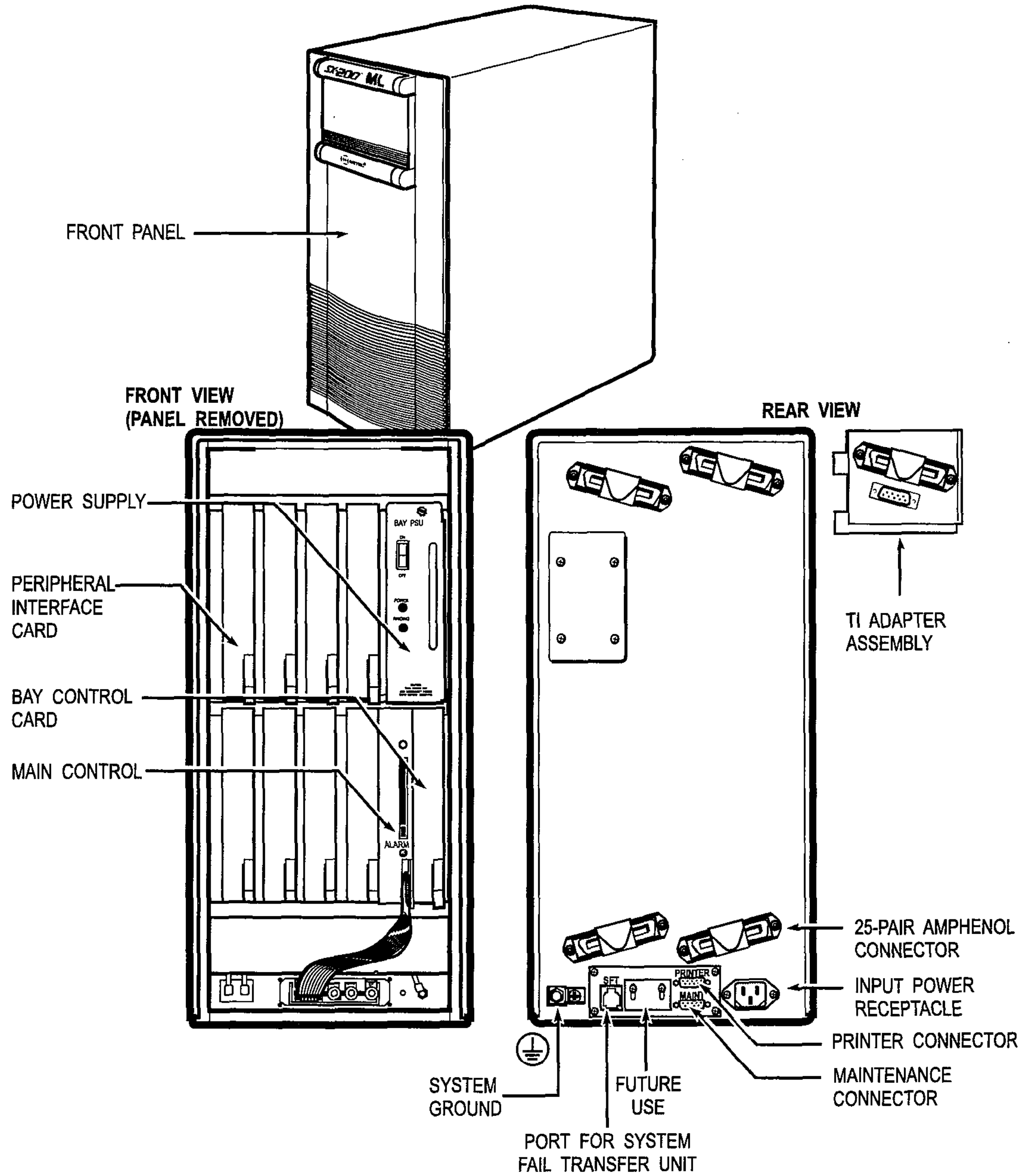


Figure 2-1 ***SX-200*** **ML Cabinet**

General Description

Physical Characteristics

Table 2-1 Physical Characteristics		
Cabinet Height	48.0 cm	19.0 inches
Cabinet Width	22.9 cm	9.0 inches
Cabinet Depth	42.0 cm	16.5 inches
Cabinet Weight (cards installed)	19 kg	42 pounds

System Requirements

Environmental and electrical requirements for the PABX are shown in Table 2-2 and Table 2-3 respectively.

Table 2-2 Environmental Requirements		
Operating Temperature	0 to 40° C	32 to 104° F
Storage Temperature	-20 to 50° C	-4 to 122° F
Relative Humidity (Operating) (non-condensing)	20 to 80%	20 to 80%
Relative Humidity (Storage) (non-condensing)	10 to 90%	10 to 90%

Table 2-3 Electrical Requirements	
AC Input Voltage	102 - 135 Vac (115 v - 9109-008-000-SA)
AC Input Frequency	47-63 Hz
AC Input Power	125 Watts RMS for each cabinet
DC Output Power	100 Watts for each bay
Reserve Power	Uninterruptable Power Supply

Software Configurations and Supported Peripheral Devices

SX-200 ML *LIGHTWARE* 16 Software

Each *SX-200* ML *LIGHTWARE* 16 software package includes all features and is accompanied with a System ID module, which is plugged into the Main Control Card II to enable operation. The *SX-200* ML *LIGHTWARE* 16 optional features are enabled by a password and include:

- MITEL Telephony Applications Interface Package (includes MiTAI™)
- Data communication features

- Hotel/Motel
- Property Management System
- Automated Attendant, and
- Automatic Call Distribution.

Voice Capabilities

The *SX-200 ML* PABX interfaces to analog and digital peripheral devices using standard twisted-pair office wiring. Peripheral devices include, but are not limited to, the following:

Standard Telephones: Industry-standard rotary dial and DTMF telephones are supported; ONS and OPS lines provide an interface for these sets.

SUPERSET 401+™ Telephone: The *SUPERSET 401+* telephone is a single-line digital telephone. It has a **Trans/Conf** key, a **Message** key, a **Hold/Retrieve** key, and six **Speed Dial** keys, a hold lamp and a message indicator lamp, as well as keys for adjusting the ringer and handset receiver volume.

SUPERSET 410™ Telephone: The *SUPERSET 410* telephone is a multi-line digital telephone. In addition to its10 fixed function keys, the *SUPERSET 410* has 6 keys that can be programmed as Speed Call keys, Feature Access keys or Line Appearances. The set also has a **Message** key and **Microphone** key with indicator lamps. Up to three MILINK™ Data Modules can be connected to this set to provide an interface for a data device.

SUPERSET 420™ Telephone: The *SUPERSET 420* is a multi-line digital telephone. It has 12 programmable keys with associated LCD indicators. These keys can be programmed as Speed Call keys, Feature Access keys or Line Appearances. Directly below the 12 programmable keys is a 2x16 alphanumeric bitmap graphics display and three softkeys. The three softkeys allow set users to select command prompts that appear in the display. The set also has a **Message** key and **Microphone** key with indicator lamps, and eight fixed function keys without indicator lamps. Up to three *MILINK* Data Modules can be connected to this set to provide an interface for a data device.

SUPERSET 430™ Telephone: The *SUPERSET 430* telephone is a multi-line digital telephone that provides advanced telephony features. The *SUPERSET 430* has12 keys that can be programmed as Speed Call keys or Line Appearances. An alphanumeric bitmap graphics display and six softkeys allow users to select call handling features easily. Up to three *MILINK* Data Modules can be connected to this set to provide an interface for a data device.

Attendant Console: The SUPERCONSOLE 1000™ and SUPERSET 7000™ Attendant consoles interface by one pair wiring to a DNIC circuit on a Digital Line Card.

Programmable Key Module: The Programmable Key Module (PKM) provides 30 additional keys and line status displays to a *SUPERSET 410, SUPERSET 420*, or *SUPERSET 430* telephone. Flash rates for the PKM indicators are the same as on its associated telephone.

DNIC Music-on-Hold/Paging Unit: The DNIC Music-on-Hold/Paging Unit (DMP) can be wall-mounted next to the *SX-200* ML PABX. It is powered by the *SX-200 ML* PABX, and does not require a separate power source. A single 25 pair amphenol connects to the *SX-200* ML PABX via the main distribution frame. A single LED indicator provides basic status information. The DMP interfaces a standard *SX-200 ML* DNIC port to the following external equipment:

- External music source for Music on Hold
- External paging amplifier (with or without answerback capability)
- Up to two night bells, and
- An external alarm.

Data Capabilities

MITEL datasets provide data facilities for terminals, digital *SUPERSET* telephones, and other types of data circuits. The DATASET 1100 series support asynchronous data communications at rates up to 19.2 kbps and interface with the *SX-200* DIGITAL PABX through a Digital Line Card. The DATASET 2100 series support asynchronous and synchronous data communications at rates up to19.2 kbps and interface with the PABX through a Digital Line Card.

***MILINK* Data Module:** The *MILINK* Data Module is connected to a modular jack located on the base of *SUPERSET 410*, *SUPERSET 420*, or *SUPERSET 430* telephones, and is used to interface a terminal, personal computer, or other peripheral device to a host computer. The *MILINK* Data Module interfaces to the PABX Digital Line Card through the same pair of wires that the telephone set voice circuit uses.

DATASET 1103 Standalone: The DATASET 1103 Standalone is packaged in a flat case which can be placed under a standard desk telephone set. It is functionally the same as one-half of the DATASET 1102 Dual Rack Mounted Card.

DATASET 2103 Standalone: The DATASET 2103 Standalone is a Synchronous/Asynchronous data set which is used with Mitel digital private automatic branch exchanges (PABX) to interface peripheral devices to the PABX. It is packaged in a flat case which can be placed under a standard desk telephone set. The DATASET 2103 Standalone can be connected to a Digital Line Card within the PABX by a single twisted pair (the telephone set is connected independently).

Terminals and Printers: A VT100™ compatible terminal or personal computer with terminal emulation software can be connected to the system as a maintenance terminal. Printers are used to create hard-copy records such as traffic reports, maintenance information, etc.

Customer Data Entry (CDE)

Customer Data Entry is accomplished from the maintenance terminal or from the attendant console. The console LCD guides the attendant through the data entry procedure by displaying a series of prompts and listing the required steps to be performed. The console display has four lines of 80 characters each. The two top lines display the steps to be taken, and the two bottom lines display the prompts which define the 10 function keys on the attendant console. For additional information, refer to the Customer Data Entry Practice. The maintenance terminal displays a full screen of each programming form.

3 Hardware Overview

General

The heart of the system is the Main Control Card II (MCC II), a 16-bit microprocessor operating at a clock frequency of 8 megaHertz. The main controller, through its address, data, and control buses, interfaces to memory, Direct Memory Access Controller (DMAC), a Digital Signal Processor (DSP), a Message Subsystem, and a DX Matrix.

In the *SX-200* ML system configuration, the MCC II controls the digital bay through the Bay Control Card to which it is attached.

SX-200 ML System Configuration

The main hardware components of the cabinet include:

Main Control Card II - performs call processing and maintains overall control through communication with the Bay Control Card. Four megabytes of RAM, one megabyte of non-volatile RAM (NVRAM), a System ID Module, a DX module, and a Stratum 3 or Stratum 4 clock are part of the Main Control Card II. The MCC II provides seven tone receivers, conferencing, and DTMF tone generation

System Software Storage - system software is stored on a two-megabyte PCMCIA memory card. When the system is powered up, call processing and maintenance software is loaded from PCMCIA memory to the MCC II RAM.

Customer Data Entry Storage - CDE software is stored in NVRAM, and can also be stored on an off-board personal computer. When the system is powered up, CDE software is loaded from NVRAM to the MCC II RAM.

Bay Control Card - interfaces the peripheral cards with the Main Control Card.

Peripheral Interface Cards - interface trunks and peripheral devices, such as telephones, *SUPERSET* telephones, and datasets into the system. Up to eight Peripheral Interface Cards (PICs) can be installed in Slots 1 to 8. The following section provides functional descriptions of the Peripheral Interface Cards.

Bay Power Supply - provides required voltages to peripheral cards, control cards, and system peripheral devices.

Backplane - the Bay Control Card, the Bay Power Supply, and the eight Peripheral Interface Cards plug into connectors on the backplane.

Peripheral Interface Cards

Digital Peripheral Cards measure 35.8 cm (14.1 inches) long and 15.8 cm (6.2 inches) high. High-power cards can only plug into upper card slots; low-power cards can plug into upper or lower card slots.

Universal Card

The Universal Card holds up to four modules. Each module is assigned a power rating (x). The cumulative ratings of the modules on the Universal Card cannot exceed a value of 10. The Universal Card plugs into an upper (high-power) slot. If more than seven receivers are required, a Universal Card and one or more Receiver Modules must be installed. The modules are as follows:

- Receiver/Relay Module (four DTMF Receivers and two Relays per Module) (power rating = 2)
- Music On Hold/Pager Module (one music input, one paging output) (power rating = 1)
- E&M Trunk Module (one trunk) (power rating = 3).

ONS Line Card. There are 12 DTMF/Rotary line circuits per card. The card accepts up to three industry-standard DTMF/Rotary telephone sets per line circuit. The ONS Line Card interfaces the telephone analog input with the system's digital crosspoint network. It converts the analog telephone signals into the digital format used by the system, and converts the digital information back into the analog signals required by the telephone sets. This is a low-power card that may be installed in any digital peripheral slot.

Note: Each *SX-200* ML PABX includes one ML only ONS Line Card, part number 9109-010-003-NA.

Digital Line Card. The Digital Line Card (DLC) interfaces *SUPERSET 401+* telephones, *SUPERSET 410* telephones, *SUPERSET 420* telephones, *SUPERSET 430* telephones, asynchronous DATASETs, synchronous DATASETs, a DMP Unit, the *SUPERCONSOLE 1000* Attendant Console, and the *SUPERSET 7000* Attendant Console to the *SX-200* ML PABX through its Digital Network Interface Circuits (DNIC); the DNIC is a proprietary integrated circuit. The card has 12 circuits, and is a low-power card that can plug into any digital peripheral slot. If a *SUPERCONSOLE 1000* Attendant console is connected to a DLC, that DLC must be installed in a high-power slot.

Note: Each *SX-200* ML PABX includes two ML only Digital Line Cards, part number 9109-012-002-NA.

LS/GS Trunk Card. The LS/GS Trunk Card contains six loop start or ground start trunks (jumper-selectable) and six message registration inputs. This is a low-power card that may be installed in any digital peripheral slot.

Direct Inward Dial (DID) Trunk Card. The DID trunk card contains six 1-way Direct Inward Dial circuits. The DID trunk allows incoming trunk calls to dial directly to an extension within the PABX without attendant intervention. It plugs into a high-power slot.

Off-Premise (OPS) Line Card. The OPS line card interfaces the PABX to extensions which are part of the system, but are located in a different building from the PABX. It contains additional protection circuitry to protect the PABX from extraneous high voltages or induced currents that may appear on the line. Each OPS card has six circuits which each connect to an extension. The OPS Line Card plugs into a high-power slot.

T1 Trunk Card. The T1 Trunk Card provides an interface to one 24-channel (D4 format) T1 trunk. It is a high-power card; because of signal cable restrictions it must be positioned in slot 6. With a dual T1 adapter, two T1 trunk cards (in slots 5 and 6) are allowed.

4 Software Overview

General

The software is divided into two separate sections, one to run the main control processor and one to run the peripheral control processors.

Upon power-up, call processing and maintenance routines are downloaded to the main controller memory. The main controller executes the maintenance routines and logs all major occurrences to non-volatile random access memory (NVRAM), Main Control Card II 7-segment displays, and the maintenance terminal (if equipped).

When the system is functional, the software relevant to the peripheral systems is downloaded from the main controller memory to the Bay Control Card's memory. The transfer is accomplished via the message subsystem link, using a single 64 kilobit per second bi-directional channel.

Each software section has its own layers to carry out system functions. The layers defined below apply to both controllers unless otherwise specified.

Physical Layer

The physical layer consists of the operating hardware such as the microprocessor, its associated memory, and input/output devices.

Scheduling Layer

This layer provides for scheduling of the different events to be handled.

Communications Layer

This layer takes care of the message subsystem software. The messages between the processors are sent in HDLC formatted packets. The formatting, sending, receiving, and unpacking of these messages is handled by the message subsystem software. Messages are transmitted using a single 64 kb/s channel in each direction.

Message Subsystem

The message subsystem is used as a communication link between the main controller and the peripheral controllers. Initially the message subsystem is used to download software from the main controller memory to the peripheral controllers' memory using one or two channels per peripheral bay. During system operation, a single channel is used as a communication link between the Main and Peripheral Processors. Messages are sent in HDLC formatted frames, ensuring error-free transmission of data.

Device Input/Output Layer

This layer handles low level details of interfacing to input/output devices such as telephony devices and RS-232 ports.

Utilities Layer

This software layer provides general utilities needed for resource management, error handling, and command interpretation.

Main Control Processor Applications Software

The main control processor applications software is responsible for controlling all activities in the PABX. It communicates with the peripheral processors via the communications software layer for such things as "origination", "digits received", etc. It also interprets Classes of Service, System Abbreviated Dial, Personal Speed Call, Automatic Route Selection, etc. The main control processor software controls all call processing, customer data entry, and maintenance management applications.

Peripheral Control Processor Software

The peripheral processor handles tasks that are real-time intensive, such as debouncing switchhooks, collecting rotary dial pulses, tone cadencing, and signaling on trunks. The peripheral processor is also responsible for monitoring hardware activity, such as cards being added and removed, and reporting all events to the main control processor via the message subsystem.The main control processor acts as the manager of the tasks performed by the peripheral control processor.

Database

The customer data entry database is stored on RAM and backed up onto a non-volatile RAM (for up to 68 hours) on the MCC II. Upon system power-up, the database is transferred from NVRAM to the MCC II RAM. Customer data entry information can also be kept on a remote PC for retrieval in case of major system failures. Other system information such as switchhook flash timing, trunk timings, and rotary digit translation for different countries, is routed to the peripheral control systems for processing.

5 Maintenance

General

This section briefly describes the maintenance diagnostics for the PABX. These diagnostics test the operation of the system hardware. The main control system controls and schedules the diagnostics.

Upon power-up or reset conditions, the diagnostics software (if enabled) has temporary control of the entire system. Once the system has been verified and the PABX is operational, the diagnostics run as low priority background tasks.

Modular design and functional packaging of the equipment permits rapid location and replacement of defective components. Circuit malfunctions are detected by diagnostic routines automatically initiated by the MCC II. Diagnostic routines, detailed in the *General Maintenance Information* Practice, and the *Troubleshooting* Practice, direct service personnel to the defective circuit card or assembly, and identify the required field-replaceable unit. Diagnostic routines and maintenance procedures do not interfere with users unaffected by the malfunction.

Maintenance Objectives

The objectives of the maintenance routines are to isolate a fault to a replaceable card or module. Maintenance functions can be performed from either the attendant console or an RS-232 terminal.

RS-232 Maintenance Terminal

The main control system interfaces to an RS-232 VT-100 maintenance terminal and to the Attendant Console in order to enable the user to access the diagnostic menu. The diagnostic menu will direct the user through the required procedures in order to interrogate the status of the diagnostic subsystem. The user can initiate maintenance routines with specific parameters via the terminal. Refer to the *RS-232 Maintenance Terminal Practice*. Results can be routed to the appropriate device (printer or terminal) according to the user's instructions.

Diagnostic Log Files

A file of the major occurrences in the diagnostic system is maintained in NVRAM. This file can be directed to the RS-232 maintenance terminal, the attendant console, or a printer.

Types of Diagnostics

There are three types of diagnostic routines:

- power-up
- background, and
- directed.

Power-up diagnostic routines consist of the complete set of diagnostics for the system. When enabled, they are executed upon system power-up and may last several minutes. These diagnostics will perform a rigorous check on the response and performance of the hardware and firmware. Any failures will be displayed on the LED display and the RS-232 terminal (when requested), and logged in a file on the NVRAM.

The background diagnostic routines consist of the complete routines which are run during system operation. When enabled they are executed as low priority background routines during system operation.

Directed diagnostics are diagnostic routines that are selected by the maintenance person and then run on specified circuits or devices.

Database Installation and Updates

The database can be installed or updated from a maintenance terminal or PC with terminal emulation connected to the *SX-200* ML PABX Maintenance connector. Back-up copies of the database can be stored on a PC or on one of the PC's floppy disks. There are no storage devices on an *SX-200* ML PABX.

General Description

NOTES

NOTES

9109-098-105-NA Issue 1
March 1997

SX-200® ML PABX

Features Description

MITEL®

NOTICE

Table of Contents

List of Figures

List of Tables

1 Introduction

General

1.1 This practice describes the features provided by the MITEL® SX-200® ML Private Automatic Branch Exchange (PABX).

Reason for Issue

1.2 This issue includes information for LIGHTWARE™ 16 ML software.

About this Practice

1.3 The feature descriptions in this practice are arranged alphabetically. An index contains detailed cross references for all PABX features, Class of Service (COS) options, and system options.

Each feature is described under four subheadings as follows:

Description:	Gives a brief description of the feature.
Conditions:	Lists the conditions and limitations of the feature.
Programming:	Summarizes the required system programming steps to activate the feature.
Operation:	Describes the operations which the end user must perform to use this feature.

Most feature descriptions in this practice make reference to one or more of the following:

- Class of Service (COS) Options
- System Options
- Feature Access Codes

These terms are explained briefly following.

Features in the PABX are controlled by enabling COS options that are accessed through customer data programming (CDE). COS options include attendant feature options and station feature options. Refer to the index at the end of this practice for detailed cross references to each of the COS options. Indexing to COS options is by both COS number and feature name.

Features Description

System Options and Timers - includes the selectable system options and variable timers. Refer to the index at the end of this practice for detailed cross references to each of the system options. Indexing is by both system option number and name.

Feature Access Codes - are the set of access codes that the user is required to enter to access features. Refer to the index at the end of this practice for detailed cross references to each of the features. Indexing is by both feature number and name.

Disclaimer

The following products have been manufacture discontinued by Mitel. These products are supported but not described in *SX-200* ML Practices:

- SUPERSET 3™ and SUPERSET 4® telephone sets
- SUPERSET 3DN™ and SUPERSET 4DN™ telephone sets
- DATASET 1101 data cartridge
- SUPERSET™ DSS module.

The following products and peripheral devices are not supported on the *SX-200* ML PABX and are not described in *SX-200* ML Practices:

- Modem Interconnect Panel
- DATASET 1102 Rack-mounted Dataset
- DATASET 2102 Rack-mounted Dataset
- DATACABINET 9000 data cabinet
- DATASHELF 9100 datashelf
- ISDN Node
- Fiber Interface Module (and associated products)
- Peripheral Node
- LCD Console (and Console module for Universal Card).

2 Features

This section of the practice describes the features which apply to the PABX.

Some features are available to all devices, while others are available to specific devices such as extensions, datasets, consoles, trunks, or *SUPERSET* telephones; certain features are available only with specific software releases. Throughout the text, specific descriptions are provided where required.

Abbreviated Dial

Description

The Abbreviated Dial feature allows trunks and extensions to be accessed by dialing a 2 to 8-digit number which is then translated by the system into the actual number. The actual number can contain up to 26 digits.

The Abbreviated Dial feature can also give system-wide access to a defined set of long distance numbers, while denying general access to long-distance dialing.

Abbreviated dial numbers can also be used as dial-in trunk prefixes, as routing points for automatic call distribution (ACD) interflow, automated attendant, and as call-forwarding points.

The attendant or customer data entry (CDE) programmer can program or display system abbreviated dial numbers. These numbers can be marked as confidential to prevent them from appearing on display telephones or in SMDR reports.

Special codes can be embedded within the abbreviated dial number for different features. The following codes can be inserted into a stored number:

*3 = Wait for user to manually insert digits (2 digits)
*5 = Call to Call Announce Port. This code must be at the end of the number; it cannot have digits following it.
** = DTMF digit *
# = DTMF digit #

Example: For example, a typical number for external directory assistance is 9+1+(area code)+5551212; the area code is to be dialed manually. The number to be stored would be 91*3035551212.

Conditions

The following conditions apply to this feature:

- Tie, DID, and DISA trunks, extensions, *SUPERSET* telephones, data devices, and consoles can access abbreviated dial numbers directly.
- Calls can be forwarded to abbreviated dial numbers.
- No toll control is done for external calls using abbreviated dial.

- Through the Automated Attendant feature, all devices can directly dial abbreviated dial numbers.
- Confidential numbers can only be viewed from CDE or by consoles with COS Option 110 (Attendant Abbreviated Dial Confidential Number Display) enabled.
- Access to abbreviated dial numbers is available to: interflow points in ACD paths, and automated attendant default destinations.
- Abbreviated dial numbers dialed after automatic route selection (ARS) leading digits will be interpreted as ARS digits to be outpulsed rather than as an abbreviated dial number.
- If an abbreviated dial number is to appear in another speed dial number (Abbreviated Dial or Speedcall), then the abbreviated dial number index must be three digits in length (leading zeros required).

Programming

Enable COS Option 245 (Abbreviated Dialing Access) for non-console devices to access abbreviated dial directly. Consoles do not need this option enabled to access abbreviated dial directly.

Enter the desired abbreviated dial index numbers, and the digit strings to be dialed, into CDE Form 31 (System Abbreviated Dial Entry). These can also be entered from the attendant console. See *Attendant Abbreviated Dial Number Entry*.

Assign an access code to Feature 24 (Abbreviated Dial Access) in CDE Form 02 (Feature Access Codes).

Operation

To dial an abbreviated dial number, when dial tone is heard:

- Dial the Abbreviated Dial access code.
- Dial the desired Abbreviated Dial index number (one to three digits).

Access Codes - Global Find

Description

The Access Codes feature allows the user to view all access codes in the system. The system reports the type of device associated with the access code and its location (Bay/Slot/Circuit). The user can also query the system about a particular access code. Data in Customer Data Entry Form 35 (Global Find Access Code), is generated by the system and cannot be modified.

Conditions

None.

Programming

None.

Operation

Refer to the *Customer Data Entry Practice*.

Account Codes

Description

Account codes are typically used to charge the cost of outgoing trunk calls to departmental cost centers or project accounts. The account code can be optional or mandatory, and appears on all station message detail recording (SMDR) records. An account code can apply to both incoming and outgoing trunk calls.

Account codes can range from 4 to 12 digits.

For verified account codes, see *Account Codes - Verified* in this section.

Account codes can also apply to data calls. See *Data Account Codes* in the Data Features section of this guide.

Conditions

The following conditions apply to this feature:

- If verified account codes are used, account codes can be four, six, eight or 12 digits in length.
- If the account code is of the variable length type, the account code digits must be followed by a "#", except where the entered code length is 12 digits.
- Rotary-dial type telephones always have a digit length of six, as they can not dial the # character.
- The user of a SUPERSET 420™ or SUPERSET 430™ telephone can also enter account code(s) during a call.
- A "*" or a "#" is not allowed in an account code.
- A "#" is only allowed to be dialed if the account codes are of the variable length type.
- If no SMDR buffers are available, the extension user receives busy tone when the account code entry is attempted.
- If you enable forced account codes for industry-standard telephones that also have the Direct To ARS feature enabled, users must dial an account code before they can originate a call.
- The Direct To ARS feature applies to *SUPERSET* telephones when a valid account code has been entered.
- Extension users who have Forced Account Codes enabled in their COS are not forced to enter an account code for an external call unless the SMDR feature is programmed to record the call and the account code.
- For *SUPERSET 420* and *SUPERSET 430* telephones, an account code can only be entered during a call if the call is a 2-party call, the other party being a trunk (of any type). There is one exception. An account code can also be entered by an ACD agent on such a call, who is also being silent monitored by a supervisor.
- For a *SUPERSET 420* and *SUPERSET 430* telephones, the **Account Code** softkey is present, providing that a consultation hold is not in progress, the set has completed dialing, and SMDR is enabled.

- In addition to station and *SUPERSET* telephones and consoles, DISA, Tie and DID trunks can also access account codes. If an error is made when entering a verified account code, the trunk is dropped.
- Account Code access for DID trunks is available.

Programming

To force an extension user, Tie, DID or DISA trunk to use account codes for long distance calls, enable COS Option 201 (Account Code, Forced Entry - Long Distance Calls) for the device. Refer to the *Automatic Route Selection and Toll Control Practice*, for the designation of Long Distance calls.

To force an extension user, Tie, DID or DISA trunk to use account codes for all external calls, enable COS Option 200 (Account Code, Forced Entry - External Calls) for the device's class of service.

Set Account Code Length (System Option 55) in CDE Form 04 (System Options/System Timers).

Assign a feature access code to Feature 01, Account Code Access in CDE Form 02 (Feature Access Codes).

COS Option 700 (SMDR - Does Not Apply) must be disabled in the extension, or Tie, or DISA trunk COS before the extension may use account codes.

COS Option 806 (SMDR - Record Incoming Calls) must be enabled in an incoming trunk COS to permit account codes to be used on incoming calls on the trunk to a *SUPERSET 420* or *SUPERSET 430* telephone.

Enable all of required SMDR options as outlined in the *Station Message Detail Recording Practice*.

Enable System Option 36 (End Of Dial Character) for use with variable length account codes.

To provide feature key activation for account codes at SUPERSET 410™ telephone program an ACCOUNT CODE feature key. (See *Feature Keys*.)

Operation

Operation varies depending upon the type of device as described below.

Industry-standard and SUPERSET 401+™ Telephones:

To access a trunk via account code entry:

- Dial the access code for account code entry.
- Dial the account code digits (if System Option 55 - Account Code Length is selected, account code digits must be followed by a "#" to indicate end of account code).
- Dial tone is returned. The account code entered now applies to the trunk call immediately made while still listening to dial tone.
- Dial the trunk call.

At the completion of the call, an SMDR record is printed. This printout includes the time of call, trunk used, duration of call, and the account code.

***SUPERSET 410* Telephones:**

To access a trunk via account code entry:

- Press the ACCOUNT CODE feature key.
- Dial the account code digits (if System Option 55 - Account Code Length is selected, account code digits must be followed by a "#" to indicate end of account code).
- Dial tone is returned. The account code entered now applies to the trunk call immediately made while still listening to dial tone.
- Dial the trunk call.

At the completion of the call, an SMDR record is printed. This printout includes the time of call, trunk used, duration of call, and the account code.

***SUPERSET 420* Telephones:**

To access a trunk via account code entry:

- Dial the access code for account code entry.
- Dial the account code digits (if System Option 55 - Account Code Length is selected, account code digits must be followed by a "#" to indicate end of account code).
- Dial tone is returned. The account code entered now applies to the trunk call immediately made while still listening to dial tone.
- Dial the trunk call.

OR

- During a call, press SUPERKEY key.
- Press the NO softkey until ACCOUNT CODE? appears in the display.
- Press the YES softkey.
- Enter the account code. Press the ← softkey to erase incorrect digits.
- Press the SAVE softkey.

***SUPERSET 430* Telephones:**

To access a trunk via account code entry:

- Dial the access code for account code entry.
- Dial the account code digits (if System Option 55 - Account Code Length is selected, account code digits must be followed by a "#" to indicate end of account code).
- Dial tone is returned. The account code entered now applies to the trunk call immediately made while still listening to dial tone.
- Dial the trunk call.

OR

- During a call, press the ACCOUNT CODE softkey. The LCD second line shows ENTER ACCOUNT CODE:
- Enter the account code. Press the ← softkey to erase incorrect digits.
- Press SAVE softkey.

Account Codes - Verified

Description

The Account Codes - Verified feature helps to ensure accuracy for accounting purposes, and helps to prevent fraudulent use of Direct Inward System Access (DISA) lines and outgoing trunks. Verified account codes control access to trunks and external (DISA) access to the system by checking the dialed account code against a list of preprogrammed codes.

The caller's COS and COR can be changed (traveling class of service) when a valid account code is entered. This can give the caller access to different features and external call privileges. If the caller's COS is changed and the direct to ARS feature is enabled in the COS, the direct to ARS feature operates immediately.

Each verified account code has an active/inactive status. This allows accounts to be denied access when problems are encountered such as nonpayment of billings.

See *Account Codes Verified (Special DISA)*, *Resale Package*, *Trunk Operation (DISA)*. See also *Analog Networking* for the use of Verified Account Codes in networking.

Conditions

The following conditions apply to this feature:

- When there is a COR and COS associated with an account code, the COS and COR apply to the device for the duration of the call. Once the device has completed the call (goes on–hook), the original COS and COR are restored.
- The match for account code is attempted when the account code is completely entered. For fixed length account codes this is the account code length and for variable length account codes this is when the "#" is entered.
- The caller is given reorder tone if there is no exact match for the account code entered.
- By default, all account codes are active.

Programming

Enable System Option 05 (Verified Account Codes).

Select the number of account code digits (VARIABLE or 4 - 12 digits) via (System Option 55) (Account Code Length).

Enter the Verified Account Codes, COS, and COR into CDE Form 33 (Account Code Entry) as required.

Activate or deactivate Verified Account Codes in CDE Form 33 (Account Code Entry) as required.

Operation

See *Account Codes.*

Account Codes - Verified (Special DISA)

Description

Verified Account Codes can be used to replace the Direct Inward System Access (DISA) code. A caller who accesses a Special DISA trunk must dial an account code rather than the DISA code. By using a Verified Account Code, each DISA trunk can have access to its own COS options through the COS and COR associated with the account code. SMDR records each of these calls.

Conditions

The following conditions apply to this feature:

- There is no minimum digits dialed before the trunk is given re-order tone as is the case with regular DISA. The rules for dialing invalid verified account codes apply.
- The use of a verified account code rather than the DISA code for entry in to the system applies on an individual trunk basis. DISA trunks not selected to use the verified account code still use the DISA code.

Programming

See programming under *Trunk Operation - Direct Inward System Access (DISA)* and *Account Codes - Verified.*

Enable COS Option 808 (Special DISA) for the DISA trunk(s).

Operation

To access the feature from an external line:

- Access the PABX on a specified DISA trunk - ringback is heard, followed by dial tone.
- Dial the DISA Account Code - if the system verifies the account code, dial tone is returned; if not, the trunk is dropped.
- Dial the required number.

Add Held

Description

Add Held allows a user engaged in an active call on a *SUPERSET 420*, or a *SUPERSET 430* telephone to add a call that is on hold on another line to the current line.

Conditions

The following conditions apply to this feature:

- This feature is applicable to *SUPERSET 420* and *SUPERSET 430* telephones only.
- The set must have, in addition to its prime line, an appearance of another line.
- A private trunk line cannot be added into an established call.
- The feature can be used to add parties into a conference call or to transfer a call in progress to another line appearance.
- A conference on hold on a line cannot be added.

- The party doing the add held must not have a consultation hold in progress and must not be talking to a third party.
- The console cannot be involved in the affected call.
- The line being added to can be a private trunk line only if dialing on the line (there can be no established call on the line).
- There must be less than five parties in the current call.
- The current call must not be on consultation hold.
- There must be no other parties in the current call performing an add held at the same time.

Programming Enable COS Option 302 (Flash-in Conference) in the set's COS to allow it to add a held line while talking to another party.

Operation ***SUPERSET 420*** **Telephones:**

- Establish a call, with a call on hold on another line.
- Press the **Addheld** softkey. SELECT HELD LINE appears in the display.
- Press the flashing line key of the call on hold. The call on hold is added to your current call.

SUPERSET 430 **Telephones:**

- Establish a call, with a call on hold on another line.
- Press the **Add Held** softkey to add a call on a previously held line to the current call. The LCD display prompts the user to press the line select key associated with the call on hold, and the call on the line is added to the current call.

Analog Networking

Description Analog Networking allows an *SX-200* PABX to send and receive caller information over a private network. The other nodes in the network can be any combination of MITEL *SX-200* DIGITAL PABXs, *SX-200* LIGHT PABXs and SX-2000® INTEGRATED COMMUNICATIONS™ systems. Analog Networking uses the ARS Modified Digit feature to insert feature access codes and other codes (called information elements) into the outgoing digit string (refer to information elements in Table 2-2). The information elements consist of special codes for inserting the caller's extension number, account code, and node identification. A glossary of analog networking terms can be found in Table 2-1.

An *SX-200* system with Analog Networking can function as an end node, an intermediate node, or a hub. As an end node, network elements are transmitted into the network via DTMF digits. As an intermediate node, all information elements received by the intermediate node are passed on to the next node. As a hub, an *SX-200* PABX receives information elements to provide Calling Party Identification.

The information elements are:

Caller's Extension Number: Consoles and *SUPERSET* Display telephones show the caller's extension number when a user answers a call. The caller's extension number replaces the trunk number or trunk name in the display, and the trunk number in SMDR records.

Caller's Dialed Account Code: The existing account code access code identifies the beginning of a caller's dialed account code. The account code is written into the SMDR record associated with the call. Verified account codes can be used to implement Traveling Class Marks, by providing a COS and COR with the account code, which, at the destination node, replaces the COS and COR associated with the trunk being used. The destination node processes the call using the COS and COR associated with the caller's verified account code that was passed on the trunk.

PABX Node ID: This information element is associated only with the originating node and must be imbedded with another information element because it has no access code of its own (usually it is the first digit of the extension number).

Conditions

The following conditions apply to this feature:

- SMDR for outgoing calls must be enabled for the account code to be sent to the destination. Refer to the *Station Message Detail Recording Practice.*
- This feature is compatible with that on the *SX-2000 INTEGRATED COMMUNICATIONS* System.
- The calling extension number and account code are prefixed with an appropriate feature access code.
- If data associated with an information element (such as an account code) is not available, no digits are outpulsed for that element. This could cause the call to fail at the far end because the feature access code is sent, but not the data associated with it.
- The Analog Networking feature transmits additional digits, increasing post dialing delay.
- SMDR can be instructed not to record the digit modification done for passing the network information.
- Dial pulse trunks cannot be used for analog networking because the # digit is not available on them.
- Analog networking does not work with System Option 36 (End of Dial Character #) enabled.
- The destination node trunk must be a dial-in type trunk (Tie or Special DISA).
- Only the last five digits of the calling extension number are displayed on *SUPERSET* display telephones and consoles.
- The # digit is required after the account code if the far end expects a variable length account code.
- The node identification provides a combined access code or calling extension number for the far end. If not associated with one of these elements, then it must match the same access code in the far end.

- If a device is forwarded to an analog network trunk, then the extension number of the forwarding telephone is sent as the calling extension number.
- On an intermediate node call (if incoming and caller's extension were received on the trunk), the caller's extension number received on the trunk is sent as the caller's extension number, even if the incoming trunk was forwarded to an outgoing trunk or routed to ARS.

Programming

In CDE Form 22 (ARS: Modified Digit Table), program the caller information elements that are to be transmitted on the trunk.

In CDE Form 02 (Feature Access Codes) assign a feature access code to Feature 39 (Analog Network Accept Caller's Extension) for PABXs that will be receiving analog network trunk calls.

To record SMDR information for incoming calls, enable COS Option 806 (SMDR - Record Incoming Calls) and COS Option 808 (Special DISA) in the COS of the destination node trunk; refer to the *Station Message Detail Recording Practice.*

Enable System Option 06 (Analog Networking SMDR) to overwrite the trunk ID information with the calling party's extension number.

Programming Example

This example shows how CO type trunks can be used with analog networking. The trunks are programmed as Special DISA trunks on the destination node so that the network information can be dialed on the far end and accepted. The Special DISA account code must be sent prior to the analog networking information. Since the originating node must receive dial tone from the destination node before dialing the Special DISA account code, precede the Special DISA account code with a pause (the number and type of pauses depend on the trunk and the destination node DISA answer time). For example, if the destination node has a DISA answer time of 4 seconds, a Special DISA account code of 3333, and an accept caller's extension access code 49, the following modified digit string is required:

*2 3333 49 *6 #   where:

- *2 is wait for dial tone (in originating node) - to wait for dial tone from the far end
- 3333 is a Special DISA account code (in destination node)
- 49 is accept callers extension (in destination node)
- *6 is insert callers extension number (in originating node)
- # is callers extension terminator (in destination node)

Operation

The *SX-200* PABX detects an incoming network information element on an incoming analog network call, and then displays the calling extension number on the called console or *SUPERSET* display telephone.

Table 2-1 Glossary of Analog Networking Terms

Term	Definition
COS	Class of Service
COR	Class of restriction
ARS	Automatic Route Select
ANI	Automatic Number Identification
Traveling Class Marks	A means of tagging the call with the caller's level of authorization (COS and COR); based on account codes, to be used by the private network in the handling of the call.
End Node	A network node that supplies network information elements to the network, but does not receive information elements from the network; (i.e., information flows from the end node out into the network.)
Intermediate Node	A network node linking at least two other network nodes. Information elements received by Node B from Node A, are supplied to Node C.
Hub	A node in the network that uses information elements from other nodes in the network in order to provide some network feature.
Information Elements	The following information elements are available: Caller's Extension Number, Caller's Dialed Account Code, and Primary Node-ID of the PABX.
Null Network Element	This is a network element's access code followed immediately by a terminator.
Marker	This is a special character to identify Information Elements that are programmed in CDE Form 22 (ARS Modified Digit Table); currently "*6", "*7", and "*8" are used.

Table 2-2 Information Elements for Analog Networking

Information Element	Feature Access Code	Marker	Comments
Caller's extension number	Accept Caller's Extension (destination node)	*6	The extension number passed in intermediate node or the extension number of the forwarding party or the extension number associated with the telephone is sent.
Caller's dialed account code	Verifiable Account Code (destination node)	*7	The account code the caller dialed is sent. If the trunk at the destination is a Special DISA trunk and this information element appears first in the digit string, the feature access code is not required.
PABX Node ID	Node ID (originating node)	*8	The Node ID of this PABX is sent. This marker usually precedes the caller's Extension marker, forming a compound extension number.

Features Description

Attendant Abbreviated Dial Number Entry

Description

This feature allows the attendant to program system abbreviated dial numbers from the attendant console. Selected attendants have the option of making abbreviated dial numbers confidential. This restricts the viewing and changing of the number to only those attendants permitted to do so.

See *Abbreviated Dial.*

Conditions

The following conditions apply to this feature:

- Only one device (console or terminal) can be programming system abbreviated dial numbers at one time.
- While the console is using the abbreviated dial number entry feature, CDE Form 31 (System Abbreviated Dial Entry) cannot be accessed. Similarly, the console cannot use the abbreviated dial number entry feature while CDE Form 31 (System Abbreviated Dial Entry) is being accessed.
- The number programmed is not limited to external numbers - it can be any access code in the system.

Programming

To allow abbreviated dial programming, enable COS Option 111 (Attendant Abbreviated Dial Programming) for the console.

To allow the display of confidential abbreviated dial numbers, enable COS Option 110 (Attendant Abbreviated Dial Confidential Number Display) for the console. This option applies to both programming and dialing of the confidential abbreviated dial numbers.

The console has unrestricted use of all abbreviated dial numbers. Program an access code in CDE Form 02 (Feature Access Codes) for Feature 24 (Abbreviated Dial Access).

See *Abbreviated Dial* for further information.

Operation

To program an abbreviated dial number, perform the following procedure:

- Press the FUNCTION key.
- Press the ATT FUNCTION softkey.
- Press the ABBR DIAL softkey.
- Enter the desired index number.
- Press the ENTER softkey.
- Press the PRIVATE softkey (only if this is to be a confidential number).
- Enter the desired index number.
- Press the SET softkey.
- Press the EXIT softkey.

To view abbreviated dial numbers, enter the following:

- Press the FUNCTION key.

- Press the ATT FUNCTION softkey.
- Press the ABBR DIAL softkey.
- Enter the desired index number.
- Press the EXIT softkey.

Attendant Access (Dial 0)

Description

A feature access code (usually 0) is provided for reaching the attendant. The destination can change based on night/day service. The destination can be a device type other than a console or LDN. There is also a second class of dial 0. See *Priority Dial 0* in this practice.

The attendant can be reached by dialing:

- attendant access code (usually 0), which is to a console directory number,
- console directory number, or
- attendant access code which is routed to an LDN key.

Conditions

The following conditions apply to this feature:

- The attendant access code can generally be used in the system wherever an attendant individual directory number can be programmed except in the call rerouting table; this can include call forwarding.

Programming

Assign an access code to Feature 11 (Extension General Attendant Access) in Form 02 (Feature Access Codes).

Assign a console directory number to Station Dial 0 Routing in CDE Form 19 (Call Rerouting Table).

Operation

Dial Attendant Access Code.

Attendant Advisory Message Setup

Description

There are eight default and seven programmable messages for use on *SUPERSET 420*, and *SUPERSET 430* LCD displays. The attendant can read a set's currently displayed message, or read through the available messages and choose one for display on the set; see *Messaging - Advisory*.

Conditions

None.

Programming

None.

Operation

To read all of the available messages, perform the following procedure:

- Press the FUNCTION key.

- Press the ATT FUNCTION softkey.
- Press the STATIONS softkey.
- Enter a *SUPERSET 420* or a *SUPERSET 430* extension number.
- Press the SET UP MSG softkey.

The messages will appear at the start of the second line on the console's LCD display. All of the available messages can be read by pressing the NEXT softkey. An OFF softkey is presented for the message currently displayed (if any) on the specified *SUPERSET* telephone. All other messages will result in an ON softkey being presented.

To set up or remove a message from a *SUPERSET* telephone, enter the same keys as above in the same order, and press the ON or OFF softkey as required.

Attendant Alarm Readout

Description

The attendant console can display the alarm logs presently active in the system. The attendant can cause a readout of the alarm messages one by one using the softkeys. The message indicates the fault and its location.

Conditions

For the attendant to access the alarms, the following conditions must apply:

- There must be no other console accessing alarms.
- There must be no console or maintenance terminal accessing maintenance or CDE function.
- There must be no current test line access.
- Logs can be reviewed only once from the attendant console.
- The ALARM softkey indicator appears only if an alarm state currently exists.
- The alarm icon is not presented for minor alarms conditions.

Programming

Enable COS Option 102 (Attendant Display of System Alarms) for the console.

Operation

To obtain an alarm readout, press the following keys in sequence:

- FUNCTION key.
- ALARM softkey.
- MORE softkey.
- CANCEL softkey.

Attendant Automatic Overflow

Description

Attendant automatic overflow provides a recorded announcement to incoming calls that are not answered by the attendant within a pre-defined time. This feature operates primarily during peak periods of incoming traffic. For more information on the operation and programming see the feature *RAD Support.*

Conditions

An answering machine should not be used. Because the connection to the recorded announcement device is listen only, the caller cannot leave a message.

Programming

In the console's class of service, set COS Option 118 (Attendant Call Forward - No Answer Timer) for the time that the answering machine should wait before playing the recorded announcement. Enable COS Option 705 (Automatic Overflow from Attendant) in the caller's COS (trunk, extension, or *SUPERSET* telephone). Enter the extension of a recording device hunt group into CDE Form 19 (Call Rerouting Table) under "UCD/Attendant Recording Routing for This Tenant", under the console's tenant.

Operation

When an incoming call is not answered within the programmed time, a recording is connected to the caller to advise the caller that there are many incoming calls and the call will be answered shortly. The actual message given is a pre-recorded message, recorded by the customer on a customer-provided recorded announcement device. The position of an incoming call being held in the calls waiting to be answered queue, is maintained while receiving the recorded message. The recorded message will be terminated as soon as an attendant becomes available. If the incoming call is still waiting to be answered after the complete message has been delivered, the incoming call is connected to system Music-on-Hold. The Final Ring Time-out (System Option 51) in CDE Form 4 continues to run until the console answers the call. It does not stop when the RAD answers the call.

Attendant Bell Off

Description

This feature allows the attendant to mute the console ringer. Incoming calls are indicated by a flashing Answer Key LED and LDN softkeys displayed on the console. When the console ringer is disabled, "BELL OFF" appears on the second line of the console LCD display.

Conditions

The following conditions apply to this feature:

- The system default state for this feature at power-up is "BELL ON".
- The status of the bell is ignored by lockout alarms ringing at the console.
- There is no bell on the console if Attendant Tone Signaling has been enabled on the console.

Programming Enable COS Option 100 (Attendant Bell Off) for the console.

Operation To disable the console ringer, press the following keys:

- FUNCTION key.
- BELL OFF softkey.

To enable the console ringer, press the following keys:

- FUNCTION key.
- BELL ON softkey.

Attendant Busy Override

Description This option allows the attendant who encounters a busy connection, to override the connection and enter the call; see *Override (Intrude).*

Conditions The following conditions apply to this feature:

- This applies to override of trunk calls through the Attendant Direct Trunk Select feature. See *Override (Intrude)* for more conditions.
- You cannot override a call if a trunk or extension in the call has COS Option 238 (Override Security) enabled in its class of service.
- An Override warning tone is heard by both parties before voice contact is established.

Programming Enable COS Option 500 (Override) for the console.

Operation Having reached a busy number:

- Press and hold down the console OVERRIDE softkey
  - all parties in the connection hear a warning tone and the attendant is connected to the call.
- Release the OVERRIDE softkey
  - the attendant is released from the call.

Note: If the call cannot be overridden, re-order tone is returned, and the console LCD displays "CANT".

Attendant Callbacks - Busy/No Answer

Description The attendant can set up a callback if the called destination is busy or does not answer. The attendant can also cancel all callbacks in the system.

See *Expensive Route Warning* for callbacks to ARS for less expensive routes.

See *Callbacks* for details on callbacks.

Conditions

The following conditions apply to this feature:

- See *Callbacks*.
- The attendant can only cancel callbacks from a system device if the Device Interconnection Control Table allows the device and the console to be connected (check CDE Form 30 - Device Interconnection Table).
- See *Attendant Default Call Position* for information on the call position where the CALLBACK softkey appears.

Programming

None.

Operation

Having reached a busy or non-answering number, press the CALLBACK softkey. The console rings as soon as the destination becomes available. When the attendant answers the ring, CALLBACK appears on the display. As soon as the attendant answers the callback, the called party rings.

Attendant Call Forward Setup and Cancel

Description

The Attendant Call Forward Setup and Cancel feature allows the attendant to set up, review and cancel call forwarding for any extension. The extension for which the attendant sets up forwarding need not have any of the call forwarding features in its COS. The attendant may also set up call forwarding from the extension to the attendant. The attendant can also cancel call forwarding for all extensions at once.

Conditions

The following conditions apply to this feature:

- The extension or console to which the calls are forwarded must not have Option 234 (Never a Forwardee) in its COS.
- Valid forward destinations are: hunt groups (not data or modem pool), consoles, telephones, abbreviated dial numbers, Dial 0 access code, LDN, night bells, and ACD paths.
- The cancel of all forwarding by the console only applies to extensions for which device interconnection control checks pass between the console and the extension.

Programming

Enable COS Option 123 - Attendant Call Forward Setup and Cancel in the attendant COS, and if split call forwarding is required, enable COS Option 260 (Internal/External Split Call Forwarding).

Operation

To set up call forwarding to an internal number, perform the following procedure:

- Press the FUNCTION key.
- Press the ATT FUNCTION softkey.
- Press the STATIONS softkey.
- Dial the extension number.
- Press the CALL FWD softkey.

- Press the EXTERNAL or INTERNAL softkey (only when split forwarding enabled).
- Enter the desired call forward destination extension number.
- Press the ALWAYS **or** NO ANSWER **or** ON BUSY **or** BUSY/NO ANS softkey.

To set up call forwarding to an external number, press the following keys in this sequence:

- Press the FUNCTION key.
- Press the ATT FUNCTION softkey.
- Press the STATIONS softkey.
- Dial the extension number.
- Press the CALL FWD softkey.
- Press the EXTERNAL or INTERNAL softkey (only when split forwarding enabled).
- Enter the desired call forward destination speed call number.
- Press the ALWAYS or NO ANSWER or ON BUSY or BUSY/NO ANS softkey.

To review call forwarding for an extension, press the following keys, in this sequence:

- Press the FUNCTION key.
- Press the ATT FUNCTION softkey.
- Press the STATIONS softkey.
- Dial the extension number.
- Press the CALL FWD softkey.
- Press the EXTERNAL or INTERNAL softkey (only when split forwarding enabled).

To cancel call forwarding for a single extension, press the following keys, in this sequence:

- Press the FUNCTION key.
- Press the ATT FUNCTION softkey.
- Press the STATIONS softkey.
- Dial the extension number.
- Press the CALL FWD softkey.
- Press the EXTERNAL or INTERNAL softkey (only when split forwarding enabled).
- Press the CANCEL softkey.

To cancel call forwarding at all stations:

- Press the FUNCTION key.
- Press the ATT FUNCTION softkey.
- Press the MORE . . . softkey.
- Press the CAN. ALL FWD softkey.

Attendant Call Selection

Description

The attendant console has up to ten call selection positions that appear as softkeys when the console is receiving an incoming call. The system sets up some positions by default for certain call types. Other positions are user defined LDN keys.

Calls arriving at the console are queued on a first come first served basis and the answer LED indicator flashes. The LCD display also indicates the number of calls waiting.

The attendant call selection feature allows the attendant to answer calls in sequence or by call type. Pressing the answer key answers calls in the order that they arrived at the console regardless of call type. Using a selection position softkey answers calls by call type regardless of the time they arrived at the console.

Conditions

None.

Programming

For setting up answering positions other than the defaults listed in ATTENDANT DEFAULT CALL POSITIONS, see *Console LDN Keys*.

Operation

To answer the first call in the attendant queue, press the ANSWER key - the tone ringer stops, the prompt associated with the call type is removed if there are no more calls of that type calling, the display shows the number of the calling trunk or extension and the attendant is connected to the calling party.

To answer a specific call type, press the softkey associated with the desired call type - the tone ringer stops, the ANSWER LED indicator indicates steadily, the display shows the number of the originating party, and the attendant is connected to the calling party.

Attendant Call Splitting and Swapping

Description

During the setting up of a call between two parties, the attendant may be required to speak to both parties, or to speak privately with either party. The attendant can do this by using the CONF, SOURCE, and DEST softkeys.

Conditions

None.

Programming

None.

Operation

Establish a 3-party conference via the CONF softkey - the attendant may now speak to both parties.

Press either the SOURCE or DEST softkey to split the call and talk to the selected party privately. The attendant may alternate between the parties by pressing one of the two softkeys, as required to select the desired party.

Press the RELEASE key to disconnect the attendant from the conference, leaving the two parties connected.

Press the CANCEL key to drop the conference, disconnecting both parties from the attendant, and leaving the console in idle mode.

Attendant Calls Forwarded On No Answer

Description

Calls directed to the Listed Directory Number (LDN) of a console which are not answered within a predetermined time-out period are rerouted to a NIGHT 1 destination (if there is one). For CO trunks, the reroute is to the trunk's NIGHT 1 answer point. For DID trunks, the reroute is to the Attendant DID Access NIGHT 1 answer point for the trunk's tenant. For dial-in Tie trunks, the reroute is to the Attendant Dial-in Tie Access NIGHT 1 answer point for the trunk's tenant. For sets, consoles, and DISA trunks, the reroute is to the Dial 0 NIGHT 1 Answer Point for the set's or console's or trunk's tenant.

See *Call Rerouting* for how this fits in with recalls to the console.

Conditions

The following conditions apply to this feature:

- Rerouting does not occur unless the NIGHT 1 and day service answer points are different.
- No further rerouting is done for DID/TIE trunks after a DID/TIE BUSY or ALWAYS reroute, or after a DID/TIE trunk is routed to the Night Access Point for DID or TIE trunks.
- No reroute is done for calls directed to the LDN from a DND, Vacant or Illegal number intercept, or from a DID/TIE intercept routing.
- No reroute is done for a call recalling to the console (calling the default RECALL softkey).
- The feature applies after forwarding has occurred to an LDN.
- This feature is mutually exclusive with Attendant Automatic Overflow; also see *Attendant Automatic Overflow.*

Programming

Enable COS Option 107 (Attendant Automatic Call Forward - No Answer) for the console where the LDN is programmed. If the LDN is programmed on more than one console, enable COS Option 107 in the Class of Service of the console that has the lowest bay/slot/circuit location.

Set the time-out period via COS Option 118 (Attendant Call Forward - No Answer Timer) for the console where the LDN is programmed; the default time is 30 seconds.

Enter a NIGHT 1 routing point for the appropriate call types via CDE Form 19 (Call Rerouting Table).

Operation

When the call arrives at the console, the timer for the Forward On No Answer is started. When the timeout occurs, the caller is routed to the appropriate answer point.

Attendant Conference

Description

This feature allows the attendant to enter into a conference with the destination party and the source party of a call. The attendant may also initiate a 3-party conference call. When the attendant is in a conference, a periodic warning beep is given to all internal parties if System Option 10 (Attendant Conference Beeps) is enabled.

Conditions

The following conditions apply to this feature:

- The console may be involved in a conference with a maximum of two other parties.
- A conference cannot be created involving another console or a call announce port.
- Device interconnection checks do not apply to the conference.

Programming

To enable attendant conference warning beeps, enable System Option 10 (Attendant Conference Beeps).

COS Option 120 (Attendant Conference Disable) must be DISABLED in the console's COS.

Operation

To enter into a conference with the source and destination parties, press the CONF softkey.

To initiate a conference, perform the following procedure:

- Dial the first party and establish a connection.
- Dial the second party and establish a connection.
- Press the CONF softkey. The three parties are now in a conference.

Attendant Console Display Language

Description This feature allows the attendant to display the attendant console softkeys in English, French or Spanish.

Conditions None.

Programming None.

Operation To change the softkey language:

- Press the FUNCTION key.
- Press the LANGUAGE softkey.
- Press the softkey of the desired language.

Attendant Console LCD Display

Description The PABX continually displays the time of day on the right-hand portion of the status line of the attendant console LCD display. When the console is idle, the date (month, day, year) is also displayed. The displayed time is used by Message Waiting, Traffic Measurement, SMDR, *SUPERSET 420* and *SUPERSET 430* telephones, and other features. The console attendant can change the date and/or time (see *Attendant Date and Time Setup*).

The attendant console may have calls from outside trunks and extensions queued that are waiting to be answered. The total number of calls in the queue is displayed in the attendant console queue (Calls Waiting) area of the display located in the top right corner of the LCD display.

The attendant can put a party on softhold or hardhold that also has a call (one party or Conference) on softhold. This is called stacked hold.

When the attendant console establishes or answers a call, the display provides information about the call. The available items of call information are:

- Extension number and set name
- Tenant name
- Trunk name
- Trunk group name
- Trunk number
- ANI information
- COS and Class of Restriction (COR)
- COS name
- Call forward recall information

- On hold information (Extension Number or Conference) of the calling party is provided if that party has a call on softhold.

The system programmer can assign names to sets, classes of service, tenants, trunks and trunk groups; see *Names*.

Conditions

The following conditions apply to this feature:

- A name must be programmed for the set. Names are programmed in CDE Form 09 (Station/*SUPERSET* Telephones). Users of *SUPERSET 420* and *SUPERSET 430* telephones can program their name at their set.
- The COS number and COR number appear only if there is no COS name programmed.
- Names associated with trunks appear only after the console answers a trunk call.
- If the attendant answers a call and that caller has a call (one party or Conference) on softhold, the attendant does not take over the caller's softhold.
- When a HOLD key is pressed to put someone on hold, only the party that is talking to the attendant will be put on hardhold. If there is a softhold party (either the SOURCE or DEST), this party will be connected to the attendant. The attendant must press the CANCEL or RELEASE key to hang up on this call.
- The tenant name appears only on rerouted TIE, DID, or Dial-In calls where the tenant name is programmed in the CDE Call Rerouting Table. If the tenant name is displayed, no other trunk information appears.
- The trunk group name appears only if there is no trunk name programmed.
- The trunk number appears only if there is no trunk name or trunk group name programmed.
- If there are no calls in the calls waiting queue, there is nothing in the CW area of the LCD display.
- The maximum number of calls waiting that can be displayed at the console is 99.

Note: The actual queue maximum is 200 calls waiting.

Programming

None.

Operation

When the attendant console establishes or answers a call, the display provides information described above about the call. For Attendant Calls Waiting operation refer to *Attendant Call Selection and/or Attendant Hold Positions* for operational details.

Attendant Console LDN Keys

Description

Each console has nine programmable listed directory number (LDN) positions. Each LDN position can be programmed as the answer point for a particular type of call. Each LDN key can be given a descriptive label, allowing the attendant to answer the call with an appropriate response.

LDNs can appear at more than one console in a system to allow calls to be presented to specified consoles simultaneously.

LDN's may also appear at subattendant positions. See feature description for *Subattendant Console LDN Keys.*

The attendant console can answer calls from an LDN by either using the ANSWER key or by selecting the LDN key directly. See *Attendant Call Selection* in this document.

Conditions

The following conditions apply to this feature:

- Any or all LDN keys and labels may be programmed to appear.
- Each console must have its own LDN numbers explicitly programmed in CDE Form 08 (Attendant LDN Assignments).
- There are several default call positions provided by the system for various call types; see *Attendant Default Call Positions.*
- The RECALL key (key #1) cannot be changed.
- Station and *SUPERSET* telephones and consoles cannot directly dial an LDN.
- Each LDN key is provided with a default label. For key F2, the label is 'INTERNAL'. For key F3 to key F0 it is 'LDN n', where n is from 1 to 8. An LDN programmed at the key can have a label that replaces the default label.
- It is recommended that key F2 not be assigned to an LDN key since some calls are directed to that key by the system; see *Attendant Default Call Positions.*
- If the Trunk Answer From Any Station (TAFAS) feature is used then is it recommended that key F0 not be programmed as an LDN or that calls not be routed to the LDN when TAFAS calls are also to be answered; see *Attendant Default Call Positions.*
- For multi-appearance LDN keys, the console (or subattendant position) with the lowest physical location number (bay #/ slot #/ circuit #) is always the owner of an LDN.

Programming

Assign access codes to the console's LDN positions via CDE Form 08 (Attendant LDN Assignments). The feature descriptions in this practice identify which features can use console LDN keys as answer points.

Operation

The attendant may selectively answer any incoming call type by pressing the appropriate LDN softkey.

Attendant Console Lockout

Description The attendant can enter an access code to restrict the capabilities of the attendant console. This prevents system tampering via the console during breaks, etc. When the console is locked out, the following restrictions take effect:

- no outgoing trunk calls can be made.
- there is no attendant function access.

The attendant console can still be used to initiate internal calls, and to answer incoming trunk calls.

Conditions The attendant can lock out the console at any time as long as there is no source party connected.

Programming Assign an access code to Feature 17 (Console Lockout Access Code).

Operation To lock out the console:

- Enter the console Lockout access code.
  - The display changes to "Console in Restricted Service".

To return the console to normal operation:

- Re-enter the code.

Attendant Date and Time Setup

Description The system continually displays the time of day on the right-hand portion of the status line of the attendant console LCD display. When the console is idle, the date (month, day, year) is also displayed. The displayed time is used by Message Waiting, Traffic Measurement, SMDR, *SUPERSET 420* and *SUPERSET 430* telephones, and other features. The time may be displayed in 12- or 24-hour format. The console can change the date and/or time.

A subattendant may also change the time and date. See *Subattendant Date and Time Setup*.

Conditions A date/time change may cause some traffic measurements to be lost, and can also affect ACD reports. Care must be given when setting this COS option.

Programming Enable COS Option 122 (Attendant Setup Time/Date). If 12-hour time display is required, no clock options are required. If 24-hour time display is required, enable System Option 01 (24-Hour Clock).

Operation

To set Time-of-Day, perform the following procedure:

- Press the FUNCTION key.
- Press the ATT FUNCTION softkey.
- Press the SET TIME softkey.
- Enter the desired time.
- Press the SET or PM softkey.

To set date, enter the following in order:

- Press the FUNCTION key.
- Press the ATT FUNCTION softkey.
- Press the SET DATE softkey.
- Enter the desired date.
- Press the SET softkey.

Attendant Default Call Positions

Description

Three incoming call indicators are provided by the system to identify calls to the console directory number. In addition to these call indicators are LDN keys on the console; see *Console LDN Keys*.

The three default positions are:

- F0 (NIGHT BELL) - calls ringing any night bell in the console's tenant group (see *TAFAS*).
- F1 (RECALL) - recalls of calls handled by the console, or for multiple console operation, by any console in the system (see *Recall*).
- F2 (INTERNAL) - calls directed to the console's internal directory number.

Conditions

The following conditions apply to this feature:

- The INTERNAL position uses whatever label is programmed at key F2. The label is by default INTERNAL but an LDN key with a different label may be programmed at the same key. This does not affect the direction of internal calls to this position - only the label used changes.
- Callbacks ringing the console appear at the INTERNAL position.
- If key F0 is programmed with an LDN, calls to the LDN take precedence over night bell calls. In this case, the key label shows the name programmed for the LDN key. Pressing the key answers calls to the LDN. Calls to the night bell appear at key F0 only after all LDN calls have been handled; the label changes to NIGHT BELL.
- Serial calls, being recalls, appear at the RECALL position.
- The RECALL position (key F1) cannot be changed and cannot have an LDN key programmed at the same key.

Programming

None.

Operation

None.

Attendant Destination (DEST) Key

Description This feature allows the attendant to press a softkey (DEST) to speak to the destination party of a call, to SWAP between the destination and source parties or to SPLIT a conference call.

The destination party's extension number, COS, and COR are displayed on the second line of the console's LCD display and the source party is put on consultation hold.

See *Attendant Call Splitting and Swapping.*

Conditions This softkey only appears when the attendant console is connected to a multi-party call and the source party can be put on consultation hold.

Programming None.

Operation Press the DEST softkey - the console is connected to the destination party and the source party is put on consultation hold.

Attendant Directed Call Pickup

Description The attendant can perform a directed call pickup from the console. This will permit calls to be retrieved before the recall timer expires or if calls have been transferred to the wrong extension.

Conditions None.

Programming Assign an access code to Feature 09 (Directed Call Pickup).

Enable COS Option 218 (Directed Call Pickup).

Operation

- Dial the Directed Call Pickup code.
- Dial the extension number of the ringing telephone - the call is connected.

Attendant Directed Page

Description

This allows the attendant to page a specific key system telephone set via its telephone speaker. The connection is one-way audio, and is terminated when the paging party hangs up. Another party attempting to call a set that is being paged in this manner will receive busy tone. The paged party can answer the page as if it were a normal incoming call to the **Intercom** key.

Conditions

The following conditions apply to this feature:

- A directed page cannot be made to a set that has DND enabled.
- A directed page can only be made to *SUPERSET 410*, *SUPERSET 420*, and *SUPERSET 430* telephones that have been programmed as key system telephones.
- *SUPERSET 410*, *SUPERSET 420*, or *SUPERSET 430* telephone users can respond handsfree to a directed page. Refer to *Key Sets - Handsfree Answerback to a Directed Page* for details.
- COS Option 600 (Auto Answer) does not enable *SUPERSET 410, SUPERSET 420,* or *SUPERSET 430* telephone users to respond handsfree to a directed page.
- The attendant cannot make directed page calls to *SUPERSET 401+* telephones.

Programming

Program the Direct Paging feature access code (48) in Form 02.

Operation

To Initiate a Directed Page:

- Dial the Direct Paging feature access code.
- Dial the extension number of the party to be paged.
- Broadcast the page message.

Attendant Direct Trunk Select

Description

The console can be used to directly access (seize) a trunk for maintenance or operational procedures.

Conditions

The following conditions apply to this feature:

- For viewing the status of and accessing a trunk, the attendant must be allowed to connect to the trunk. See *Device Interconnection Control.*
- For accessing the trunk, the trunk must not be in the process of being seized by another device in the system.
- SMDR applies to the call when a trunk is accessed. If there are no SMDR records available, the call continues without an SMDR record; see *Station Message Detail Recording.*

- Trunks cannot be accessed by "Attendant Function" if the console's COS has COS Option 200 (Account Code, Forced Entry - External Calls) or COS Option 201 (Account Code, Forced Entry - Long Distance Calls) enabled.
- When an LS/GS trunk is used as a dictation trunk, the M/MM leads are used to make the trunk busy while the tape is removed or the unit is powered down. If a console tries to access this type of busy trunk, only the EXIT softkey appears (the OVERRIDE softkey and FORCE RLS softkey are not available) and the message "Trunk busy due to M/MM leads" is displayed.
- The trunk must be a member of a trunk group.

Programming

None.

Operation

To check the current status of a trunk, perform the following procedure:

- Press the FUNCTION key.
- Press the ATT FUNCTION softkey.
- Press the TRUNKS softkey.
- Enter the desired trunk number.
- Press the STATUS softkey.
- Press the EXIT softkey.

To select a trunk for attendant access only, enter the following softkeys:

- Press the FUNCTION key.
- Press the ATT FUNCTION softkey.
- Press the TRUNKS softkey.
- Enter the desired trunk number.
- Press the ATT ACCESS softkey.

To override a call on a busy trunk:

- Press the OVERRIDE softkey (see Conditions).

To force-release a non-idle trunk (a call in progress is dropped), enter the following softkeys:

- Press the FUNCTION key.
- Press the ATT FUNCTION softkey.
- Press the TRUNKS softkey.
- Enter the desired trunk number.
- Press the ATT ACCESS softkey.
- Press the FORCE RLS softkey (see Conditions).

To release a trunk from attendant access, press the RELEASE key.

Attendant DISA Code Setup

Description

This option allows the attendant to change the Direct Inward System Access (DISA) security code (Feature 19) in CDE Form 02 (Feature Access Codes) that a DISA caller must dial to access the system.

Conditions

The following conditions apply to this feature:

- The DISA code cannot conflict with the numbering plan.
- The DISA code is limited to a maximum of five digits.
- The DISA code cannot be displayed.
- An attendant cannot delete the DISA code; it may only be deleted via CDE Form 02 (Feature Access Codes). Refer to the *Customer Data Entry Practice*, for further details.
- While the DISA code (Feature 19) in CDE Form 02 (Feature Access Codes) is being setup from the Console, CDE Form 02 cannot be accessed from customer data entry. Similarly, while Feature 19 in CDE Form 02 (Feature Access Codes) is being accessed from customer data entry, the DISA code cannot be changed from the console.

Programming

Enable COS Option 103 (Attendant DISA Code Setup) for the console.

Operation

To change the DISA access code, perform the following procedure:

- Press the FUNCTION key.
- Press the ATT FUNCTION softkey.
- Press the MORE... softkey.
- Press the DISA CODE softkey.
- Enter a new DISA access code.
- Press the SET softkey.

Attendant Do Not Disturb (DND) Setup, Cancel or Override

Description

The attendant can set up or cancel Do Not Disturb (DND) for an extension.

When calling an extension with DND enabled, the attendant can override DND.

See *Do Not Disturb.*

Conditions

If the extension does not have COS Option 121 - Attendant Station Do Not Disturb enabled in its COS, the extension is not able to alter the DND setting; it can only be done through the console.

Programming

Enable COS Option 121 - Attendant Station Do Not Disturb in the console COS.

Operation

To set up Do Not Disturb on an extension, perform the following procedure:

- Press the FUNCTION key.
- Press the ATT FUNCTION softkey.
- Press the STATIONS softkey.
- Enter the extension number.
- Press the NO DISTB softkey.
- Press the EXIT softkey.

or

- Press the FUNCTION key.
- Press the GUEST ROOM softkey.
- Enter the extension/room number.
- Press the NO DISTB softkey.
- Press the EXIT softkey.

To cancel Do Not Disturb on an extension, press the same keys in the same order.

To set up or cancel Do Not Disturb while talking to an extension, toggle the NO DISTB softkey.

Attendant Extension Busy-Out

Description

This feature allows the attendant to busy out any extension (the extension is removed from service and cannot originate or receive any calls), and to remove the busy-out condition. The same operation is provided in maintenance. Refer to the *RS-232 Maintenance Terminal Practice*. If the attendant dials the number of a busied-out extension, the console displays the extension number and "BSY OUT" in the destination display and the attendant will receive reorder tone.

Conditions

The following conditions apply to this feature:

- If the extension is idle when the attendant sets the busy-out condition, the extension is busied-out immediately.
- If the extension is not idle when the attendant sets the busy-out condition, the extension is busied-out as soon as the extension becomes idle.
- The extension is treated as being busy with regard to forwarding. If the extension has "Call Forwarding - Busy" or "Call Forwarding - Follow Me" set up, the forwarding occurs.
- If the extension is a member of a hunt group, all calls to the hunt group bypass the busied-out extension.
- A locked out extension cannot be busied out.

Programming

Enable COS Option 112 (Attendant Station Busy-Out).

Operation

To busy out an extension, perform the following procedure:

- Press the FUNCTION key.
- Press the ATT FUNCTION softkey.
- Press the STATIONS softkey.
- Enter the extension number.
- Press the BUSY OUT softkey.
- Press the SET softkey.

To remove the Busy-Out condition from an extension, perform the following procedure:

- Press the FUNCTION key.
- Press the ATT FUNCTION softkey.
- Press the STATIONS softkey.
- Enter the extension number.
- Press the BUSY OUT softkey.
- Press the CLEAR softkey.

Attendant Flash Over Trunk

Description

The attendant can flash on a trunk by pressing the FLASH softkey. A flash is sent out on the trunk, and dialing is restarted on the trunk.

Conditions

The following conditions apply to this feature:

- There must be only one party at the console.
- The attendant may flash on incoming and outgoing trunks.
- The trunk must be in a trunk group.
- The feature is not available when the console is locked out.
- The flash duration is based on the trunk hardware used. For trunks in digital bays, the duration is programmed using the Flash Duration Trunk Circuit Descriptor option. For all other trunks, the time is approximately 200 ms.
- The flash type is programmable for certain trunks. See *Trunk Circuit Descriptor Options*. The Flash Over Trunk option is ignored when an attendant does a flash on a trunk.
- If the trunk is the source party then the trunk becomes the destination party.
- Dialing continues until an inter-digit timeout or the maximum number of digits (26) have been dialed. Digits dialed on the console keypad are sent out on the trunk by the system.
- When dialing is restarted:
    - If COS Option 802 (Limited Wait For Dial Tone) is not enabled for the trunk then the system will wait indefinitely for dial tone to be detected. COS Option 805 (Trunk No Dial Tone Alarm) does not apply.
    - Tones are turned off at the console and must be turned on again if needed later; see *Attendant Tone Signaling.*
    - The trunk's SMDR is completed before dialing is restarted. SMDR applies to the dialing that is started on the trunk.

Programming

None.

Operation

While the attendant is connected to an outgoing trunk:

- Press the FLASH softkey.
- Dial as required.

Attendant Function Access

Description

By pressing the console FUNCTION key and the ATT FUNCTION softkey, the attendant can access the following attendant functions:

- ABBR DIAL
- ALARM (read alarms)
- APPLICATION (To access CDE or maintenance)
- BELL ON/OFF (if named)
- BUSY OUT
- CALL FORWARD
- CANCEL ALL CALLBACKS
- CANCEL ALL CALL FORWARDING
- DAY/NIGHT 1/NIGHT 2 switching
- DO NOT DISTURB SETUP AND CANCEL
- ESPAÑOL (Spanish language prompts and messages on console)
- FLEXIBLE NIGHT SERVICE
- FORCED TRUNK RELEASE
- FRANÇAIS (French language prompts and messages on console)
- GUEST ROOM functions (if enabled)
- MESSAGE WAITING SETUP AND CANCEL
- SEND MESSAGE
- SET DATE
- SET TIME
- SET UP MESSAGE
- STATIONS
- SYSTEM IDENTIFIER
- TRUNK STATUS/ACCESS/BUSY OUT.

For more information about attendant functions, refer to the individual feature descriptions in this practice.

Conditions

Attendant functions are not available if the attendant console Lockout option (Feature 17) is invoked.

Programming

None.

Operation

Press the FUNCTION key - the attendant function softkeys appear on the console LCD display. One of the softkeys is MORE. Press this key for access to more functions. Press MORE again to return to the first set of functions.

Attendant Hold Positions

Description

The attendant may place an extension or trunk on hold in one of eight HOLD positions. There are four keys; HOLD1, HOLD2, and HOLD3 are for hold positions 1 through 3, HOLD4 is for hold positions 4 through 8. A call hold recall time of 10 to 240 seconds may be programmed (default is 30).

Conditions

The following conditions apply to this feature:

- A HOLD key LED is on when it has a call on hold.
- If music is available on the system the held party receives Music-on-Hold while on hold, even after the party starts alerting the console after a hold time-out. If no music is provided, the caller hears silence rather than ringback.
- When the call hold timeout occurs, the call alerts the console at that HOLD key - the call does not appear at any LDN position nor does it recall. The calls waiting display is updated.
- The console can selectively answer the held call by selecting individual HOLD keys.
- Non-busy extensions cannot be placed on hold.
- Conference calls cannot be placed on hold.
- If the current party is a call announce port at the console, the call on hold cannot be retrieved. If the source has a call on consultation hold, the call on hold cannot be retrieved.
- If the attendant with a call on hold is talking to a destination, and presses the HOLD key to retrieve the call, the destination party is placed on consultation hold and becomes the source. The held call is connected to the attendant and becomes the destination. The attendant can switch between parties by pressing the SRC and DEST keys on the console, or can connect the parties by pressing the RELEASE key.
- If the attendant is visually impaired and unable to see the HOLD key LED, enable COS 124 - Attendant Hold Position Security. When this class of service is enabled, an error beep sounds if the attendant attempts to put a call on hold by pressing a HOLD key that already has a party on hold.

Programming

Program COS Option 116 (Attendant-Timed Recall - Hold) in the console's COS to set the time-out period (default time is 30 seconds).

Enable COS Option 124 - Attendant Hold Position Security for consoles that require an error beep to sound when the attendant presses a HOLD key that already has a party on hold.

Operation

To put a call on hold at the console:

- Press the ANSWER key when call rings console.
- Press an idle HOLD key (1-4); call is put on hold at this HOLD key.

To retrieve a call on hold at the console:

- Press the HOLD key to speak with the call on hold.

Note: If the call has been recalled by a call hold time-out, the HOLD key indicator flashes.

If the call is to be retrieved before a time-out, the attendant may press the HOLD key where the call is being held. By pressing the HOLD key, the call is transferred to the SOURCE, or to the DEST if there is a SOURCE already.

If HOLD key 4 is used, the user must next press one of the softkey hold slots (HOLD slots 4 through 8) for both holding and retrieving.

Attendant Implicit New Call

Description

When the attendant presses a key on the console dial pad, by default a new call is initiated. When the first key is pressed, an existing party is automatically placed on hold. At the completion of dialing, the attendant can transfer the call to the dialed destination by releasing from the call.

By pressing the TONES ON softkey (if available), this feature is temporarily disabled. See *Attendant Tone Signaling.*

Conditions

None.

Programming

None.

Operation

Press any dial pad key.

- Immediately, the current call (if one exists) is placed on hold, and a new call is initiated.

Attendant Individual Directory Number

Description

Each attendant console has a unique directory number identifying that console.

The directory number is in addition to the general attendant access number (usually 0) used to obtain the attendant or any LDN keys programmed at that console. A calling party has the choice of either dialing the general attendant access number, or dialing the directory number which is dedicated to a particular attendant position (useful when there is more than one attendant position).

Conditions

The following conditions apply to this feature:

- Calls to an individual directory number are not presented to other attendants (if any).
- Calls directed to the directory number appear at a default softkey position provided by the system; see *Attendant Default Call Positions.*

Programming

Program the directory number of each attendant position in CDE Form 07 (Console Assignments).

Operation

None.

Attendant Interposition Calling and Transfer

Description

In a multiple console environment, an attendant can call or transfer a call to any other attendant using the individual attendant directory number. The call is transferred in the same method as a call to an extension.

Conditions

The following condition applies to this feature:

- Since consoles cannot be put on hold, normal attendant operations (hold, swap, etc.) are not available when talking to another console.
- One console can call and talk to another by dialing the DN number but not by dialing 0.

Programming

Program the directory number of each attendant position in CDE Form 07 (Console Assignments).

Operation

When the call has been answered, dial the directory number of the attendant console to which the call is to be transferred.

While listening to ring back or when the attendant answers, press the RELEASE key to transfer the SOURCE caller to the called console.

Attendant Message Waiting Setup and Cancel

Description

The Attendant Message Waiting Setup and Cancel feature allows the attendant to inform extension users that there is a message waiting. The message waiting indication may take the form of:

- a message on the display of a *SUPERSET 420* or *SUPERSET 430* telephone,
- a continuously flashing lamp on the extension (if equipped), or
- a distinctive ringing pattern repeated every 20 minutes. The pattern is three cycles of 3.5 ips ringing.

When the user returns and calls the attendant, the "MSW" indicator appears on the console display to indicate that there is a message waiting for that extension.

When Transparent Multi-Console Operation is used, a console may review or cancel a message waiting indication for any station; without this feature, only the console that set Message Waiting for a specific station can review or cancel it. See *Attendant Transparent Multi-console Operation.*

A message may also be setup and canceled from the front desk or PMS. See the *Hotel/Motel Feature Package Description Package*.

Use the *Attendant Message Waiting Setup and Cancel* feature if you know that the person you are trying to contact is out of the office. Use the *Attendant Callbacks- Busy/No Answer* feature if you know that the person you are trying to contact is somewhere in the office.

Conditions

The following conditions apply to this feature:

- If either message waiting COS Option is enabled, the extension rings every 20 minutes when idle, or until the message waiting is canceled. If the message waiting indication is given by a lamp, the lamp flashes (at 60 ipm).
- If the message waiting indication is given by ringing the extension, the first ring starts 10 seconds after the extension becomes idle.
- The message waiting lamp or display indicator on *SUPERSET* telephones is always lit for any message irrespective of COS options.
- If the extension has Do Not Disturb enabled, the ringing indication is not given.
- Any console within this tenant with COS Option 320 (Transparent Multi-Console Operation) enabled may cancel Message Waiting, instead of just the console that set it.

Programming

Enable COS option 232 (Message Waiting Setup - Lamp) and/or COS Option 231 (Message Waiting Setup - Bell) for each extension on which the console is to place a message.

Enable COS Option 320 (Transparent Multi-Console Operation) for each console that is to operate in Transparent Multi-Console Operation mode.

Operation

To set up Message Waiting on an extension, perform the following procedure:

- Press the FUNCTION key.
- Press the ATT FUNCTION softkey.
- Press the STATIONS softkey.
- Enter the extension number.
- Press the SEND MSG softkey.
- Press the EXIT softkey.

or

- Press the FUNCTION key.
- Press the GUEST ROOM softkey.
- Enter the extension/room number softkey.
- Press the SEND MSG softkey.
- Press the EXIT softkey.

To cancel Message Waiting on an extension:

- Press the CANCEL MSG softkey.

Attendant Multi-New Call Tone

Description

If an attendant is actively engaged with an incoming call, the first call placed in the attendant call waiting queue signals the attendant with a single burst of tone. As long as there are one or more calls waiting in the queue, the attendant will continue to hear the single burst of tone at the programmed time interval. The presence of any calls waiting is also shown by the call waiting indication on the top line of the display; see *Attendant Calls Waiting Display.*

Conditions

The following conditions apply to this feature:

- This feature is disabled if the attendant bell is turned off from the console. Refer to Attendant Bell Off.
- The frequency of the new call tone is controlled by COS Option 404 (Recording Failure to Hangup Timer).

Programming

Enable COS Option 106 (Attendant New Call Tone) for the console.

Enable COS Option 125 (Attendant Multi-New Call Tone).

Set the desired time on COS Option 404 (Recording Failure to Hangup Timer). The default for COS Option 404 is 30 seconds.

Operation

None.

Attendant New Call Ring

Description If an attendant is already actively engaged with an incoming call, the first call placed in the attendant call waiting queue signals the attendant with a single burst of ringing. Subsequent calls do not alert the attendant when they are added to the queue. Their presence is shown by the call waiting indication on the top line of the display; see *Attendant Calls Waiting Display.*

Conditions The following conditions apply to this feature:

- This feature is disabled if the attendant bell is turned off from the console. Refer to Attendant Bell Off.
- This feature is disabled if TONES ON is on.

Programming Enable COS Option 106 (Attendant New Call Tone) for the console.

Operation None.

Attendant Night/Day Switching

Description The attendant may select NIGHT1 service, NIGHT2 service, or DAY service via softkeys; also see *Night Services.*

Conditions The following conditions apply to this feature:

- When NIGHT1 or NIGHT2 has been selected by an attendant console, the status is displayed on the right-hand side of the destination line of the display for all affected consoles.
- The console switches service for tenants controlled by that console; refer to CDE Form 06 (Tenant Night Switching Control) and to the *Tenanting Practice.*

Programming Refer to *Night Services* and to *Tenanting.*

Operation To switch to NIGHT1, NIGHT2, or DAY service, select the following keys:

- FUNCTION
- NIGHT 1 or NIGHT 2 or DAY SERVICE

Attendant Paging Access

Description

The attendant can access a paging zone or zones by using the PAGE key on the attendant console. Pressing the PAGE key connects the console handset directly to the zones of the paging equipment programmed for default access for the console. This overrides any extension announcement in progress. The attendant can alternatively access the paging circuit by dialing the associated access code followed by a digit 0 - 9 for the zone required (0 accesses all zones).

See also *Attendant Paged Hold Access And Paging.*

Conditions

The following conditions apply to this feature:

- If the attendant is to connect to specific zones, program an access code in Feature 13 (Paging Access to Specific Zones) in CDE Form 02 (Feature Access Codes).
- Any extension(s) using the paging zone(s) that the console is attempting to access is overridden and removed from the pager and given busy tone.
- The console bell is turned off while the PAGE key is held down.
- If System Option 03 (Single Paging Amplifier) is enabled then the attendant cannot override the current pager user(s).
- A console cannot override another console in a paging zone.
- For paging on all zones simultaneously, all zones must be either not in use or the console must be able to override the user of the zone(s).
- While the PAGE key is held down, the rest of the console keys are non-operational.
- When on a pager that has been accessed by a feature access code, tone signaling can be turned on using the TONES ON softkey. See *Attendant Tone Signaling.*

Programming

Enable one or more of the following COS Options for the console in CDE Form 03 (COS Define) as shown in the following table.

COS Option Number	Description
303	Paging Zone 1 Access
304	Paging Zone 2 Access
305	Paging Zone 3 Access
306	Paging Zone 4 Access
307	Paging Zone 5 Access
308	Paging Zone 6 Access
309	Paging Zone 7 Access
310	Paging Zone 8 Access
311	Paging Zone 9 Access
312	Paging Default (0 to 9) (0 Gives All Enabled Zones)

For access to the default zone, assign an access code to Feature12 (Paging Access to Default Zone) in CDE Form 02 (Feature Access Codes).

Assign an access code to Feature 13 (Paging Access to Specific Zones) in CDE Form 02 (Feature Access Codes) for access to zones other than the default zones.

Operation

To connect to the default paging zone, hold down the PAGE key. The connection remains until the PAGE key is released.

To connect to a paging zone other than the default zone, dial the 'Paging Access To Specific Zones' access code, followed by the desired paging zone number (1-9).

While the console's default page zone is in use, the PAGE key LED is lit on all consoles for which the same default zone applies.

If the paging zone(s) cannot be accessed, busy tone is returned.

Attendant Paged Hold Access

Description

The attendant can put a party on hold and page for someone to pick up the call from the attendant hold position. When paging the called party, the attendant announces the digit string that must be dialed to pick up the call.

Also see *Attendant Paging Access.*

Conditions

The following conditions apply to this feature:

- Consoles, industry-standard telephones, *SUPERSET* telephones, DISA trunks, and TIE trunks can pickup the held calls.
- The party picking up the call must be able to connect to the held party; see *Device Interconnection Control.*
- The attendant cannot pick up a held call if it has a source party.
- An extension cannot pick up the held party if the extension has a consultation hold in progress and the held party has COS Option 233 (Never A Consultee) enabled.
- A station or *SUPERSET* telephone (with a consultation hold in progress) cannot pick up the held trunk if the station or *SUPERSET* telephone has COS Option 214, (Cannot dial a trunk after flashing) enabled.
- A station or *SUPERSET* telephone on a conference with a trunk on consultation hold cannot pick up the held trunk if the station or *SUPERSET* telephone has COS Option 215, (Cannot Dial a Trunk if Holding or in Conf With a Trunk) enabled.

Programming

Assign an access code to Feature 16 (Hold Pickup Access - Attendant Hold Slots).

Enable COS Option 225 (Hold Pickup Access - Attendant Hold Slots) in the COS of the device from which the pickup call is made.

Operation

If paging the default paging zone:

- Put the calling party on hold using one of the console HOLD slots.
  - When the attendant presses the PAGE key, the console displays the access code assigned to Feature 16 followed by two digits that identify the console, followed by n. The "n" digit represents the hold slot number. The attendant should learn the Hold Pickup Access code; it is not displayed when specific zone paging is used.
- Page the second party, specifying the displayed number (the last number being the number of the hold slot)
  - When the second party dials the displayed number, the second and held parties are connected.
  - If the paged party does not call, the held party recalls to the attendant automatically; see *Attendant Hold Positions.*

If paging a zone other than the default zone:

- Put the calling party on hold using one of the console HOLD slots.
- Dial the 'Paging Access To Specific Zones' access code, followed by the desired zone.
- Page the second party, specifying the console's Hold Pickup Access Code, followed by the number of the hold slot, e.g. 677002 (where 677 is the Access Code, 00 is the Console, and 2 is the Hold Slot number).
  - The paged party dials the announced code. If the paged party does not call, the held party recalls the attendant automatically.

Attendant Serial Call

Description

The attendant serial call feature allows an incoming trunk call to be set as a serial call before being transferred by the attendant. After the call is finished, the serial call recalls the attendant. This allows a caller to speak to several individuals in the PABX without the need for transfers by the called extensions.

Conditions

The following conditions apply to this feature:

- Attendant Serial Call is available on all trunk calls for all trunk types.
- Serial call returns a trunk to the console that established the call under the following conditions:
  - The party, except a console, that is talking to a serial trunk hangs up.
  - ACD interflows a serial trunk to a DROP CALL interflow point.
  - A final ringback time out occurs for the serial trunk and it is not ringing a console, LDN or night bell.
- Transparent Multi-Console Operation has no effect on serial calls.

- The RING AGAIN softkey does not appear for a serial trunk call that recalls back to the console (however it recalls).
- A serial call that is released to a subattendant will not have its recall point changed to the subattendant telephone; see *Subattendant.*
- Enabling serial call clears any previous serial call setting by another console and any recall point set up to any other device; see *Recall.*
- Serial calls appear at the RECALL call position at the console; see *Attendant Default Call Positions.*

Programming

Enable COS Option 109 (Attendant Serial Call) for the console.

Operation

To establish a serial call:

- Answer an incoming trunk call.
- Press the SERIAL CALL key.
- Dial the required destination extension number.
- Press the RELEASE key.

To answer a serial recall:

- ANSWER lamp flashes and RECALL softkey appears.
- Press the ANSWER key - ANSWER lamp is lit; SER, SRC, and SERIAL CALL is displayed on console.
- The attendant is connected to the recalling trunk.

To cancel a serial recall:

- ANSWER lamp flashes and RECALL softkey appears.
- Press the ANSWER key - ANSWER lamp is lit; SER, SRC, and SERIAL CALL is displayed on the console.
- Press the SERIAL CALL key and the RELEASE key; SERIAL CALL goes out and the call is cleared.

Attendant Source Key

Description

This feature allows the attendant to press the SOURCE softkey to speak to the source party of a call, to swap between the source and destination parties or to split up a conference call.

The source party's extension number, COS, and COR are displayed on the first line of the console's LCD display and the destination party is put on consultation hold.

A party that is on consultation hold at the console does not hear system music.

See *Attendant Call Splitting And Swapping.*

Conditions

This softkey only appears when the attendant console is connected to a multi-party call and the destination party can be put on Consultation Hold.

Programming

None.

Operation

Press the SOURCE softkey - the console is connected to the source party and the destination party is put on consultation hold.

Attendant Timed Recall

Description

This feature automatically alerts the attendant when a call extended through the console or a call held at the console has not been answered within the preselected time.

Selectable recall times include:

COS Option	Name	Timer Range
115	Attendant-Timed Recall - No Answer	5 to 240 s (0 = disable)
116	Attendant-Timed Recall - Hold	10 to 240 s
117	Attendant-Timed Recall - Campon	5 to 240 s (0 = disable)

For full details of Recall, see *Recall.* Also see *Attendant Transparent Multi-console Operation.*

Conditions

The following conditions apply to this feature:

- A value of 0 for COS Option 115 and COS Option 117 disables the Recall; however, the final ring timeout applies.
- Recalls to the console are inoperative during Night Service unless the console is the night answer point.

Programming

Select the desired recall times for COS Options 115 (Attendant-Timed Recall - NO ANSWER), 116 (Attendant-Timed Recall - HOLD), and 117 (Attendant-Timed Recall - CAMPON), in the attendant console's COS.

COS Options 115 (Attendant-Timed Recall - NO ANSWER), 116 (Attendant-Timed Recall - HOLD), and 117 (Attendant-Timed Recall - CAMPON) apply to attendant consoles only. They do not apply to sets.

Operation

See *Recall.*

Attendant Tone Signaling

Description

The attendant console usually does not transmit DTMF tones. Applications such as voice mail, however, may require the attendant to transmit tones. The attendant tone signaling feature allows the console to transmit DTMF tones during a call.

Conditions

The following conditions apply to this feature:

- The TONES ON/OFF softkey does not appear if the Attendant is receiving an Audible Lockout Alarm (if enabled).
- The tones remain on unless turned off or the attendant places a party on hold, or retrieves a party from hold, or goes idle.
- The key appears if the console is talking to a trunk, *SUPERSET* telephone or industry-standard telephone or the console is on the pager through dialing a pager access code (not through the PAGE key).
- Dialing on the keypad by the attendant usually starts a new call implicitly, and puts the current party on consultation hold as the source party. With TONES ON enabled, this feature is disabled and either tones must be turned off or the call must be put on hold at one of the attendant hold positions before another call can be started.
- New Call Tone is disabled while TONES ON is enabled.

Programming

Enable COS Option 119 (Attendant Tone Signaling) for the attendant console.

Operation

Do the following:

- During a call, press the TONES ON key.
- Send DTMF tones (by pressing dial pad keys).
- Press TONES OFF to terminate DTMF signaling.

Attendant Training Jacks

Description

Training jacks are provided on the attendant console for use by a supervisor or trainer who is training a new attendant. Each console is equipped with two attendant jacks; either jack may be used by the attendant, while the other provides a monitoring, supervisor, or training function.

Conditions

Removal of both headsets and/or handsets does not automatically switch the console into Night Service.

Programming

None.

Operation

Console or system operation does not change in any way.

Attendant Transfer To Campon

Description

This feature allows the attendant to connect calls to a busy extension, hunt group or trunk group for automatic completion when the called busy party becomes free. The attendant itself cannot camp on but can transfer calls into campon; see *Campon*. For details of recall from campon, see *Recall.*

Conditions

The following conditions apply to this feature:

- Calls that are not completed within the campon time-out will recall; see *Recall.*
- The transferred party must be able to connect to the busy party; see *Device Interconnection Control.*
- A transfer cannot be made to a locked-out extension.

Programming

Specify a time-out period in COS Option 117 (Attendant-Timed Recall - CAMPON); (default time is 30 seconds). A value of 0 disables recall from campon to the attendant.

Enable COS Option 301 (Campon) for the console to transfer to internal devices; enable COS Option 237 (Outgoing Trunk Campon) for transfers to busy trunk groups.

Operation

To camp a call onto a busy number:

- Attempt a call to a busy extension.
- Press the RELEASE key.
  - this automatically camps on the calling party to the busy number. If the transfer is not allowed then a beep tone is heard, CANT is displayed on the console, and no transfer is done.

Attendant Transparent Multi-Console Operation

Description

The Attendant Transparent Multi-Console Operation feature allows some features to apply to a group of consoles within a tenant. Messages set by any console in this group can be read or canceled by any consoles within this group.

Conditions

The following conditions apply to this feature:

- Transparent Multi-Console Operation must be enabled for each participating console.
- When a *SUPERSET 420* or *SUPERSET 430* telephone user presses the CALL softkey to call the console in response to a message received, if the message was left by a console with the Transparent Multi-Console Operation feature enabled, the call is turned into a normal Dial 0 call and routes to the Dial 0 point of the *SUPERSET* telephone.

Features Description

- Recalls to a console can be answered from any other console that the console is allowed to connect with. Form 05 (Tenant Interconnection Table) specifies which tenant groups may be connected together.

Programming Enable COS Option 320 (Transparent Multi-Console Operation) for each console within this tenant that is to be a member of the group.

Assign access codes to the Dial 0 entry in CDE Form 19 (Call Rerouting Table) for the tenant of the caller. To have the calls ring the attendant in the group, the access code must be an LDN on the consoles in the console group.

Operation One attendant can read or cancel another attendant's message waiting indications.

Calls that have been extended by one attendant will recall at multiple attendant consoles.

Attendant Trunk Busy-Out

Description The attendant may busy-out a trunk to prevent access to the trunk, and may remove the busy condition as required. If the Trunk Busy-Out Enable option is not selected, the attendant may still access individual trunks, but is unable to force them into a busy condition.

Conditions The following conditions apply to this feature:

- As with station and *SUPERSET* telephones, if the trunk is not idle when the busy is attempted, the busy out will be pending and will be processed when the trunk becomes idle.
- The console must be able to connect to the trunk; see *Device Interconnection Control.*

Programming Enable COS Option 114 (Attendant Trunk Busy-Out) for the console.

Operation To busy out a trunk, perform the following procedure:

- Press the FUNCTION key.
- Press the ATT FUNCTION softkey.
- Press the TRUNKS softkey.
- Enter trunk number.
- Press the BUSY OUT softkey.
- Press the SET softkey.

To return a trunk to service, perform the following procedure:

- Press the FUNCTION key.
- Press the ATT FUNCTION softkey.
- Press the TRUNKS softkey.
- Enter trunk number.
- Press the BUSY OUT softkey.
- Press the CLEAR softkey.

Attendant Trunk Group Status Display

Description

This feature allows the attendant to check the status of the PABX trunk groups. An attendant can display the status of all 50 trunk groups, although a maximum of 20 trunk groups are shown in each screen display. A black square, under a trunk group number, indicates that all the trunks in that group are busy.

The attendant can display the trunk status whether the console is idle or busy. If this feature is activated while the console is idle, the display is refreshed approximately every 5 seconds.

Conditions

None.

Programming

None.

Operation

To display the trunk group status, press the following keys:

- Press the TRUNK GROUP key. The display shows the status of the first 19 trunk groups.
- Press the MORE softkey to display the status of the next 20 trunk groups.
- Press the MORE softkey to display the status of the remaining 11 trunk groups.
- Press the EXIT softkey.

Auto-Answer

Description

When the Auto-Answer feature is active, incoming calls ring briefly, then the set answers the call in Handsfree mode; see *Handsfree Operation.* When the caller hangs up, a short burst of tone is heard over the *SUPERSET* telephone's speaker and the set goes idle. Call origination is not affected.

This feature is available on *SUPERSET 410, SUPERSET 420,* and *SUPERSET 430* telephones.

Conditions

The following conditions apply to this feature:

- A directed page to a *SUPERSET 410, SUPERSET 420,* or *SUPERSET 430* in auto-answer mode is handled like a normal directed page: the set rings briefly, then the set speaker turns on, but the set microphone remains off.
- There are no restrictions on the connection of Auto-Answer *SUPERSET* telephones and loop start CO and DISA trunks.
- Auto Answer is not available on industry-standard telephones or *SUPERSET 401+* telephones.
- When a call arrives at the set, the set does not warble. Instead the user hears a short tone through the speaker. The user hears another short burst of tone when the call is completed.
- If Auto-Answer is used with the headset feature then the tones are heard in the headset and not through the speaker.
- The feature prevents calls directed at the prime line of the *SUPERSET* telephone from ringing other appearances of the prime line because the call is automatically answered. If the call is not automatically answered (the prime appearance or the *SUPERSET* telephone is busy) then the other appearances ring if possible.
- The feature only operates for calls directed at the prime line of a *SUPERSET* telephone.
- The feature prevents any other line appearances on the set from causing the set to ring. Instead, the new call ring is used to notify the set user of new calls on other lines.
- Auto-Answer is ignored when callbacks, recalls and hold timeout recalls ring the *SUPERSET* telephone.
- In ACD, agents can be forced into Auto-Answer mode upon login; refer to the ACD TELEMARKETER® *Application Package.*
- If Auto Answer is activated at a set and if the prime line of the set is busy, calls to any other line appearances on the set will not be indicated with standard ringing. Instead, the calls will be indicated by New Call Ring (a single burst of ringing).

Programming

Enable COS Option 600 *SUPERSET* (Auto-Answer) in the telephone's COS.

To provide access code activation of Auto-Answer, program a Feature 02 (Auto-Answer) access code in CDE Form 02 (Feature Access Codes).

To provide feature key activation of Auto-Answer at *SUPERSET 410*, *SUPERSET 420*, or *SUPERSET 430* telephones, program an AUTO-ANSWER feature key. (See Feature Keys.)

To enable Auto Answer for a *SUPERSET* telephone that is equipped with a headset, enable COS Option 600 (*SUPERSET* Telephone - Auto-Answer) and COS Option 612 (*SUPERSET* Telephone - Headset Operation) in the telephone's COS.

Operation

Operation varies depending on the type of telephone as described below.

***SUPERSET 420* Telephones:**

To activate Auto-Answer:

- Press SUPERKEY.
- Press the NO softkey until AUTO ANSWER? appears in the display.
- Press the TURNON softkey. AUTO ANSWER ON appears briefly in the display. The display returns to showing the date and time.

To deactivate Auto-Answer:

- Press SUPERKEY.
- Press the NO softkey until AUTO ANSWER? appears in the display.
- Press the TURNOFF softkey. AUTO ANSWER OFF appears briefly in the display. The display returns to showing the date and time.

Note: You can also use the AUTO-ANSWER feature key to turn Auto Answer on and off.

***SUPERSET 410* and *SUPERSET 430* Telephones:**

To activate Auto-Answer:

- Press the AUTO-ANSWER feature key. The adjacent LCD indicator darkens, or,
- Dial the Auto-Answer feature access code, followed by the digit "1". If a feature key has been programmed, the adjacent LCD indicator darkens.

To deactivate Auto-Answer:

- Press the AUTO-ANSWER feature key. The adjacent LCD indicator clears, or,
- Dial the Auto-Answer feature access code, followed by the digit "2". If a feature key has been programmed, the adjacent LCD indicator clears.

Auto-Hold

Description A *SUPERSET* telephone user automatically places a call on hold when another Line Select or Speedcall key on the telephone is pressed. When this is not desirable, a COS option can be programmed which allows a call to be placed on hold only by pressing the HOLD key.

Conditions The following conditions apply to this feature:

- Auto-Hold is not available on *SUPERSET 401+* telephones or on industry-standard telephones.
- When this feature is disabled, calls can only be placed on hold by pressing the HOLD key.
- See *Hold* for conditions on putting a call on hold. If the hold is not allowed then the Line Select or Speedcall key is ignored.
- When this feature is disabled, and another line is selected, the call on the current line is dropped.
- The auto-hold only occurs when a Line Select key is pressed while dialing or talking, or when a Speedcall key is pressed while talking.
- If a Line Select key is pressed when dialing, if there is a consultation hold in progress then that party is automatically placed on hold.

Programming Enable COS Option 601 (*SUPERSET* - Auto-Hold Disable) to disable the Auto-Hold feature.

Operation Establish a call.

Press any Speedcall key or Line Select key; the call is placed on hold. If a Speedcall key is pressed, the system prompts for a line to use to start the Speedcall call.

Automated Attendant

Description The automated attendant feature connects incoming calls from a DTMF telephone to a recording and a receiver. The recording directs the caller to dial one or more digits to be routed to a specific answering point, such as sales, service, parts, or general office. Once a digit has been dialed, the system can add prefix digits in front of the dialed digit to provide a valid extension number, a hunt group number, a system abbreviated dial number, or a feature access code, to route the call to a specific answering point. If a digit is not dialed (or cannot be dialed because the incoming telephone is not DTMF) the incoming caller is routed to a default answering point at the completion of the recording.

For details refer to the *Automated Attendant Application Package*.

Conditions Refer to the *Automated Attendant Application Package.*

Programming The following System Options apply:

- 106 - Automated Attendant
- 59 - Receivers Reserved for Non-Auto-Attendant Use.

Refer to the *Automated Attendant Application Package.*

Operation None.

Automatic Call Distribution (ACD)

Description Automatic Call Distribution (ACD) is a method of distributing calls evenly among trained operators (Agents). Refer to the *ACD TELEMARKETER Application Package,* for more information.

Conditions Refer to the *ACD TELEMARKETER Application Package.*

Programming The following System Options apply:

- 104 - Maximum ACD Agents
- 42 - ACD Silent Monitoring
- 43 - ACD Silent Monitoring Beeps
- 44 - ACD Reports.

The following COS Options apply:

- 650 - ACD Agent Template
- 651 - ACD Supervisor Template
- 652 - ACD Senior Supervisor Template
- 653 - ACD Agent Always Auto-Answer
- 654 - ACD Display Path Always
- 655 - ACD Allow Continuous Monitor of Agent
- 812 - Loop Start Trunk to ACD Path Connect.

The following Feature Access Codes apply:

- 44 - ACD Login/Logout
- 45 - ACD Silent Monitoring.

Refer to the *ACD TELEMARKETER Application Package* for further information.

Operation Refer to the *ACD TELEMARKETER Application Package.*

Automatic Number Identification (ANI) on Outgoing Trunks

Description

This feature is a mechanism that allows the PABX to identify a calling party on an outgoing trunk. The identifying information consists of the calling party's extension number which is transmitted (tones or pulses) on the trunk, after the PABX has successfully dialed an external number on that trunk.

Conditions

The following conditions apply to this feature:

- The far-end must provide answer supervision and be capable of handling the ANI protocol.
- The extension numbers of station and *SUPERSET* telephones and consoles are sent.
- If the caller has forwarded to the ANI trunk then the forwarding extension's number is sent.

Programming

Enable COS Option 800 (ANI Applies) for the trunk.

Enable Circuit Descriptor option "Far End Gives Supervision" for the trunk.

The Circuit Descriptor option "DTMF" for the trunk must match the far end.

Operation

When a call is made on an ANI trunk, an answer supervision received on the ANI trunk is treated as a signal to send the calling party's extension number. During transmission of the calling extension number, the calling party is not connected to the speech path. This inhibits putting any noise (that could be mistaken for DTMF signaling) onto the line until all dialing is finished. Once the extension number has been sent, the calling party's audio will be cut through and no further signaling will occur.

Automatic Number Identification (ANI)/ Dialed Number Identification Service (DNIS) on Incoming Trunks

Description

This feature allows the PABX to identify Automatic Number Identification (ANI) numbers and Dialed Number Identification Service (DNIS) numbers that are transmitted to the PABX on an incoming trunk. ANI provides the telephone number of the calling party; DNIS provides the telephone number dialed by the calling party.

The PABX can receive ANI and DNIS numbers on specific types of incoming non-DISA trunks. After receiving ANI and DNIS numbers for an incoming call, the PABX can then make the numbers available to other functions within the system. You can program the PABX to provide ANI and DNIS numbers to *SUPERSET* display telephones, to *SUPERSET* consoles, to SMDR printers, and to the MITEL Application Interface (MAI) platform package.

On a *SUPERSET* console, the ANI and DNIS digits for an incoming call are displayed after the call is answered. The ANI digits appear where the trunk information is normally displayed; the DNIS digits overwrite the COS and COR fields on the console display. When ANI digits are not received, the trunk information is shown instead; when DNIS digits are not received, the COS and COR fields are visible. By default, ANI and DNIS numbers are not displayed.

On *SUPERSET* display telephones, the display shows DNIS digits while the call is unanswered. After the user answers the call, the ANI digits are displayed. Whenever ANI digits are not available for a call, the display shows the DNIS digits during both the unanswered and answered states. Whenever DNIS digits are not available, the display shows the ANI digits during both the unanswered and answered states. If neither are available, the display shows the normal trunk information. By default, ANI and DNIS numbers are not displayed.

If COS Option 613 (Display ANI Information Only) is enabled, the *SUPERSET* telephone displays the ANI digits during both the unanswered and answered states. In this case, if ANI digits are not provided, the display shows the normal trunk information.

You can program the system to record ANI and DNIS digits in the SMDR reports. The digits are recorded at the end of the trunk SMDR record. Both ANI and DNIS numbers are recorded up to a maximum of 10 digits in length. By default, ANI and DNIS numbers are not recorded in SMDR reports.

Conditions

The following conditions apply to this feature:

- Only non-DISA trunks that are programmed with a T1 E&M, T1 E&M DISA, T1 TIE DISA, or T1 DID/TIE trunk circuit descriptor can receive ANI and DNIS digits. Ensure that the trunk's circuit descriptor has the Debounce Timer is set to 100 ms.
- Incoming trunks must use DTMF signaling, and must be set to Wink Start.
- Only MCI and US Sprint standards are supported.
- If US Sprint standard is used, calls cannot be routed to extensions that have an * in their access number.
- The PABX waits a maximum of 300 ms for each ANI/DNIS digit. If it doesn't receive a digit within the 300 ms timeout period, the PABX routes the call through to the extension.
- The PABX uses the DNIS digits that are collected for an incoming call to route the trunk call through the system.
- The ANI and DNIS digit displays take priority over analog networking digits, trunk group names, trunk names and trunk numbers.
- If the PABX doesn't receive any ANI or DNIS digits for a call, then the console or set display shows the trunk information of the call instead.
- ANI and DNIS digit displays follow standard call forwarding rules (See Call Forwarding).

- On *SUPERSET* Consoles, tenant names and calling line identification displays take priority over ANI digits, and the following displays take priority over DNIS digits:
  - Intercept reasoning (FROM xxxx DND)
  - Forwarding information (FWD FROM xxxx).

Programming

Enable COS Option 811 (ANI/DNIS Trunk) and disable COS Option 801 (Incoming Trunk Call Rotary) for the trunk in Form 03.

In Form 13, assign the ANI/DNIS trunk with one of the following trunk circuit descriptors:

- T1 E&M
- T1 DID/TIE

In the circuit descriptor options subform of Form 13, program the following options for the ANI/DNIS trunk:

- Set the "Incoming Start Type" option to "Wink".
- Set the "Wink Timer" to the required standard (100-350 ms for MCI; 140-290 ms for US Sprint).
- Set the "Debounce Timer" to 100 ms or greater.

Define the ANI/DNIS trunk as a (non-dial-in) trunk in Form 14, or as a TIE or DID trunk (non-DISA) in Form 15.

To program the PABX to display ANI and DNIS digits on a *SUPERSET* display set or console, enable COS Option 502 (Display ANI/DNIS/CLASS Information) in the class of service for the set/console.

When COS Option 502 (Display ANI/DNIS/CLASS Information) is enabled, *SUPERSET* display telephones show the DNIS digits during the ringing state, and ANI digits during the answered state. To program a *SUPERSET* display telephone to show the ANI digits during both the unanswered and answered states, enable COS Option 613 (Display ANI Information Only) in the Class of Service for the set. By default, COS Option 613 (Display ANI Information Only) is disabled.

If you want ANI and DNIS digits recorded in the SMDR records, enable COS Option 806 (SMDR-Record Incoming Calls) and COS Option 814 (SMDR-Record ANI/DNIS/CLASS) in the class of service for the trunk.

Operation

None.

Automatic Route Selection (ARS)

Description The Automatic Route Selection (ARS) feature package combines the concepts of standard ARS (selecting optimum call routes, inserting/deleting routing digits) and Toll Control (allowing/disallowing specific extensions the ability to make specific types of external/long distance calls).

Also provided are:

- a universal numbering plan capability,
- six time-of-day zones,
- three day zones for week days, and
- overlap outpulsing.

Conditions Refer to the *Automatic Route Selection and Toll Control Practice.*

Programming System Option 47 - ARS Unknown Digit Length Timeout applies. Refer to the *Automatic Route Selection and Toll Control Practice*, for further information.

Operation Refer to the *Automatic Route Selection and Toll Control Practice.*

Background Music

Description This feature permits the user of a *SUPERSET 410, SUPERSET 420*, or *SUPERSET 430* telephone to have background music played through the speaker of the telephone when idle. The Music-on-Hold source provides the music; see *Music-on-Hold.*

Conditions The following conditions apply to this feature:

- This feature applies only to *SUPERSET 410*, *SUPERSET 420*, and *SUPERSET 430* telephones.
- Background music is automatically temporarily suspended when a call is placed or received; it is restored after the *SUPERSET* telephone is returned to idle.

Programming Enable COS Option 602, Background Music in the set's COS.

To activate and deactivate music on a *SUPERSET 410* or *SUPERSET 420* telephone, program a MUSIC feature key. (See Feature Keys.)

Operation Operation varies depending on the type of *SUPERSET* telephone:

Features Description

***SUPERSET 410* and *SUPERSET 420* Telephones:**

To turn on background music:

- Press the MUSIC feature key. The adjacent LCD indicator darkens. Music is heard through the set's speaker. Adjust the set's speaker volume as desired. This adjustment does not affect the handsfree speaker volume.

To turn off background music:

- Press the MUSIC feature key. The adjacent LCD indicator, which was dark, clears.

***SUPERSET 430* Telephones:**

To receive background music:

- Press the MUSIC ON softkey. Music is heard through the set's speaker. Adjust the set's speaker volume as desired. This adjustment does not affect the handsfree speaker volume.

To turn off background music:

- Press the MUSIC OFF softkey.

Broker's Call (Station Swap)

Description

Broker's Call allows the user to speak privately with two separate parties. When the user hangs up, the two parties are not connected. This feature applies to *SUPERSET 401+* telephones and industry-standard telephones.

Users that have this feature enabled can't transfer calls. However, a variant on Broker's Call is the Broker's Call with Transfer feature. The Broker's Call with Transfer feature enables the user to speak privately with two separate parties while allowing transfers; see *Broker's Call with Transfer.*

Conditions

The following conditions apply to this feature:

- Enable COS Option 203 (Broker's Call) in the COS of industry-standard telephones, and *SUPERSET 401+* telephones that require the Broker's Call feature. Users of *SUPERSET 410, SUPERSET 420*, and *SUPERSET 430* telephones perform Broker's Calls using a SWAP feature key or SWAP softkey (see *Swap-Trade Calls*).
- Industry-standard telephones, *SUPERSET 401+* telephones that have COS Option 203 (Broker's Call) enabled cannot set up conference calls or make call transfers.
- This feature prevents a *SUPERSET 401+* telephone or an industry-standard telephone from putting a conference on consultation hold; reorder tone is returned if it is attempted.

- If the extension originating the Broker's Call hangs up with a party on hold, the held party recalls to the extension instead of being transferred to a third party (see *Station Transfer Security*). COS Option 403 (Trunk Recall Partial Inhibit) is an exception to this.
- An extension with the Broker's Call feature may access the Call Hold, Call Hold and Retrieve, and Paging features after flashing on a call.
- COS Option 203 (Broker's Call) and the following COS Options are mutually exclusive:

  302 (Flash-In Conference)

  252 (Broker's Call With Transfer)

  223 (Flash Disable)

  224 (Flash for Attendant).
- The following COS Options do not apply to an extension with Broker's Call enabled:

  214 (Cannot Dial a Trunk after Flashing)

  215 (Cannot Dial a Trunk if Holding or Conf With One).

Programming Enable COS Option 203 (Broker's Call) for the extension.

Operation Operation varies depending on the type of telephone as described below.

Industry-standard Telephones:

A normal 2-party call is established, and the extension user wishes to consult a third party:

- Flash the switchhook - dial tone is heard, and the second party is placed in consultation hold.
- Dial the number of the third party and establish a private 2-party conversation.
- To alternate between calls, flash the switchhook.

***SUPERSET 401+* Telephones:**

A normal 2-party call is established, and the extension user wishes to consult a third party:

- Press the FLASH key - dial tone is heard, and the second party is placed in consultation hold.
- Dial the number of the third party and establish a private 2-party conversation.
- To alternate between calls, press the FLASH key.

Broker's Call With Transfer (Transfer With Privacy)

Description

The Broker's Call With Transfer (Transfer With Privacy) is similar to the Broker's Call feature but users are also able to transfer calls. A flash is interpreted as a Swap instead of an attempt to conference (see Broker's Call). A conference cannot be formed by an extension that has Broker's Call With Transfer enabled. However, unlike Broker's Call, when the extension goes on–hook to transfer an extension, the transfer is not prevented. As well, some consultation COS option checks are not done with the Broker's Call feature but they are done for Broker's Call With Transfer. Also see *Broker's Call (Trade Calls)* for general information about Broker's Calls.

This feature applies to *SUPERSET 401+* telephones and industry-standard telephones.

Conditions

The following conditions apply to this feature:

- COS Option 252 (Broker's Call With Transfer) (previously Transfer With Privacy) is mutually exclusive with the following options:
  - 302 (Flash-in Conference)
  - 224 (Flash for Attendant)
  - 223 (Flash Disable)
  - 203 (Broker's Call).
- An extension with COS Option 233 (Never a Consultee) may not be consulted.
- COS Options 214 (Cannot Dial a Trunk After Flashing) and 215 (Cannot Dial a Trunk After Flashing If Holding or in Conference With One), do not apply to an extension with COS Option 252 (Broker's Call With Transfer) (previously Transfer With Privacy) enabled.

Programming

If desired for extensions enable COS Option 252 (Broker's Call With Transfer).

Operation

Industry-standard Telephones:

A normal 2-party call is established, and the extension user requires to consult a third party:

- Flash the switchhook - dial tone is heard, and the second party is placed in consultation hold.
- Dial the number of the third party and establish a private 2-party conversation.
- To alternate between calls, flash the switchhook.
- Hang up to connect both parties.

***SUPERSET 401+* Telephones:**

A normal 2-party call is established, and the extension user requires to consult a third party:

- Press the FLASH key - dial tone is heard, and the second party is placed in consultation hold.
- Dial the number of the third party and establish a private 2-party conversation.
- To alternate between calls, press the FLASH key.
- Hang up to connect both parties.

Busy Lamp Field

Description

A Busy Lamp Field (BLF) on the Programmable Key Module (PKM), indicates the status (idle, busy, DND) of a line appearance for a device, such as, normal stations, *SUPERSET* prime lines, key system telephone intercom lines, logical lines and trunks. Any *SUPERSET* telephone with line keys may be programmed to use BLF indicators. If you add a PKM to a *SUPERSET* telephone you can have additional BLF indicators. A PKM provides up to 30 BLF indicators arranged in two rows of 15.

You can combine the device types with a set, to a maximum of three devices. The type limits are three PKMs with a set. PKM modules connect directly to the set. Refer to Programmable Key Module in this practice for details.

Note: If you program an indicator on a PKM as a BLF, the key beside the BLF indicator is not assigned any function. You must program the key as a Direct Station Select key - see *Direct Station Select* in this document.

The BLF indicator of a trunk line appearance, indicates the state of the trunk (idle or not idle).

The BLF indicator of logical lines, indicates whether or not a call to the line will find the line busy. If the line is a multi-call line, the BLF indicator indicates idle until there are no free line appearances to ring. If the line is a key line, the BLF indicator indicates whether or not the line is occupied.

Conditions

None.

Programming

To program BLF indications of line appearances on a PKM, perform the following steps:

- Display CDE Form 09 (Stations/*SUPERSET* Telephones) or CDE Form 45 (Key System Telephones).
- Move the cursor to the bay/slot/circuit number of the *SUPERSET 410*, *SUPERSET 420*, or *SUPERSET 430* telephone. An asterisk (*) appears to the left of type if PKMs are programmed for the set.
- Select the EXPAND PKM softkey.
- "ENTER PKM NUMBER (1-3):" appears in the command line.

- Type the address (1, 2, or 3) of the desired PKM in the command line.
- Select the ENTER softkey to display the Expand PKM Set Subform for the PKM. The Expand PKM Set Subform allows you to program the functions of the 30 available PKM keys. Keys 31 and 32 in the form are listed as RESERVED since the PKM only has 30 keys.
- Move the cursor to the desired key. (See Figure 2-2 for numbering.)
- Move the cursor to the TYPE column (use the TAB key if necessary).
- Select the BLF/DSS softkey (Type becomes "Busy Lamp").
- Press TAB key 3 times to get to EXT NUM column (or 4 times to get to the TRK NUM column).
- Enter the number of the device to be monitored by the Line Appearance.
- Select the ENTER softkey.

Note: The keys on a PKM are preceded by a 1, 2, or 3. The 1, 2, or 3 indicates the order in which you associated the PKM with the set (e.g., the keys belonging to the first PKM that you associated with the set will be preceded by a "1").

Operation

When answering an incoming call for one of the monitored stations, the attendant or *SUPERSET* user can scan the Busy Lamp Field on the PKM to determine the station's current status.

Calculator

Description

SUPERSET 430 telephones provide a basic four function calculator using the telephone keypad, display, and softkeys. The calculator has 2 modes of operation - General and Programmer. Once the calculator feature is accessed, the # key may be used to switch between the two modes. The default mode is the General mode.

General Mode: In this mode of operation, the calculator allows for arithmetic operations. The telephone keypad is used as the numeric keypad; the * key is used as the decimal point key. Arithmetic operators (x, ÷, -, +/=), clear entry/clear (CE/CLR) and another decimal point key appear on the softkeys. The display is on the top line of the LCD display.

Programmer Mode: In this mode of operation, the calculator allows for integer arithmetic / logical operations in either decimal or hexadecimal. In this mode, each softkey has 2 functions - the active function is designated by the ▶ pointer. To change to the second function, press the 2ND FUNCTION softkey. The display is on the top line of the LCD display. The second line of the display is the status register (nzvc - negative, zero, overflow, carry). When a bit in the status register is set, the corresponding letter is capitalized. The RADIX softkey switches between hexadecimal and decimal. The CLR softkey clears the displayed value (if there is no pending operator. Pressing the CLR softkey twice will always clear the status register. All operations are 32-bit. To enter hexadecimal characters A through F, enter the following: A=*1, B=*2, C=*3, D=*4, E=*5 and F=*6.

Conditions This feature is available only on *SUPERSET 430* telephones.

Programming None.

Operation The calculator feature operates as follows:

- Press SUPERKEY.
- Press CALCULATOR softkey.
- Press SUPERKEY when completed.

Call Forwarding

Description Call Forwarding allows a user to forward calls from the normal extension to an alternate destination. The user may establish call forwarding for many reasons. Call Forwarding is set up by the extension, or by an attendant console, subattendant, or another extension for the extension (for example - I'm here). Call Forwarding destinations are internal to the PABX, except for Call Forwarding - External which forwards calls to an external destination, by system abbreviated dial number, personal speedcall key, or personal speedcall. Table 2-3 lists the types of call forwarding available. Each type of call forwarding is described separately in this section.

Table 2-3 Call Forwarding Types
Call Forwarding - Busy
Call Forwarding - Don't Answer
Call Forwarding - Follow Me
Call Forwarding - I'm Here
Call Forwarding - External (destination)
Call Forwarding - Internal / External Split (according to source)
Call Forwarding - Busy / No Answer

Feature Access Codes Which Activate Call Forwarding Types

To activate call forwarding, the extension must dial the digits programmed for one of the following feature access codes.

- 03 - Call Forwarding - All Calls
  (split forwarding must be disabled)
- 04 - Call Forwarding - Internal Only
  (split forwarding must be enabled)
- 05 - Call Forwarding - External Only
  (split forwarding must be enabled)
- 06 - Call Forwarding - I'm Here
- 07 - Call Forwarding - Cancel I'm Here.

Condition Codes To Be Dialed Following Feature Access Codes 03 - 05

After dialing a feature access code, the user must dial the condition code to identify the condition under which the call is to be forwarded. This does not apply for Call Forward - I'm Here and Cancel I'm Here.

- 1 - Always (Unconditionally)
- 2 - Busy
- 3 - No Answer
- 4 - Busy or No Answer

Two examples of call forwarding from an extension are:

Internal calls are to be forwarded to 1701 when not answered	External calls are to be forwarded to 1702 when busy or not answered
Go off-hook and receive dial tone	Go off-hook and receive dial tone
Dial Call Forward Internal access code - 65	Dial Call Forward External access code - 62
Dial Condition code for No Answer - 3	Dial Condition code for Busy/No Answer - 4
Dial destination extension number - 1701	Dial destination extension number - 1702
Receive dial tone and hang up	Receive dial tone and hang up

Conditions

The following conditions apply to the call forwarding feature:

- Valid call forwarding destinations are: Dial 0 access code, night bell, LDN's, hunt groups (not data or modem pooling type), industry-standard telephones, *SUPERSET* telephones, consoles, ACD Paths, personal speedcall keys, system abbreviated dial numbers, or personal speedcalls.
- An extension is considered BUSY for Call Forwarding - Busy if the extension is not idle, has Do Not Disturb activated, or has its prime line appearance busy. See programming COS Option 262 for exceptions to Call Forward Busy.
- Forwarding is operational for when the called party is a logical line if the line appears on only a single telephone in the system. The forwarding setting of the *SUPERSET* telephone where the line is programmed is used to forward calls.
- Forwarding can be set up to the Dial 0 access code. The destination is translated from the access code to the routing point based upon the current NIGHT/DAY service at the time of forwarding (giving NIGHT/DAY based forwarding). Priority Dial 0 can be used as well, with the feature being checked for on the forwarding extension.
- Forwarding does not apply if the calling extension is the party to which the call would be forwarded.
- Forwarding does not apply if the forwardee is an industry-standard telephone, *SUPERSET* telephone, or console and has COS Option 234 (Never a Forwardee) in its COS.
- An extension can set up forwarding to an LDN.
- Forwarding cannot be programmed from an extension to itself.

- Forwarding is ignored when callbacks are honored, except when Call Forwarding - No Answer is programmed and enabled.
- No forwarding is done if the caller is the attendant, and the forwarding destination is the Dial 0 access code, and the forwarding destination is the calling attendant itself or an LDN. This restriction on the Dial 0 access code applies for both the first and second hop of call forwarding.
- If an extension is a member of a hunt group and has forwarding enabled, the extension will still be rung if called via the hunt group (its immediate forwarding is ignored, except for Call Forward - No Answer). If it is called directly, its forwarding is honored.
- With Split Forwarding enabled, an extension can have two forwarding destinations; one for internal calls, and one for external calls. Forwarding must be enabled in the set's COS.
- For Call Forwarding, all devices are treated as internal calls unless COS Option 709 (Follow External Call Forward) is enabled for the calling device.
- Calls will not be forwarded from an extension that has COS 200 (Account Code Forced Entry External Calls) activated, unless the forwarding destination is an ONS voice mail port.
- With multihop forwarding, calls may be forwarded twice (maximum) on any combination of conditions.
- When forwarding to a speedcall or abbreviated dial number, the forwarding always occurs regardless of the outcome of the call, unless to an illegal forwarding destination, in which case forwarding is ignored. For example, if toll control denies access to an external number or the forwarding is to an invalid number, no forwarding occurs. For forwarding to a personal speedcall key, the contents of the key are examined as per the validity of forwarding; if unsuitable, the forwarding is de-programmed and no forwarding occurs.
- *SUPERSET* telephone key or multicall line appearances of extension numbers do not ring or provide visual indication of incoming calls if the extension has activated Forwarding - Follow Me or Forwarding - Busy, if the caller is actually forwarded. See programming COS Option 262 for exceptions to Call Forward Busy.
- Forwarding is not done if the forward destination is busy (it is busy if: Do Not Disturb is activated, it is not idle, or its prime line appearance is not idle). An exception is for Subattendant *SUPERSET* telephones and enhanced answering positions. In these cases, the *SUPERSET* telephone is considered busy if there are no idle appearances of its prime line or multicall appearances of itself; see *Subattendant - Basic Function.*
- When Call Forwarding - Follow Me is activated, calls are always forwarded, even if the forward destination is busy.
- If the *SUPERSET* telephone being forwarded to has the Auto-Answer feature activated then the caller is answered with that feature; see *Auto-Answer.*
- Connection checking is done between the calling party and the party being forwarded to; see *Device Interconnection Control.*

- If forwarding is activated but for some reason the forwarding is not done, the call continues as if forwarding was not active. If the call forward -don't answer timer expires and no forwarding is performed, the recall no answer timer is started; see *Recall.*
- Calls forwarded to a speedcall or abbreviated dial number are treated as an external call by the system even if the entry is an internal destination.
- For *SUPERSET* display telephones and the console, the forwarding party's identity is displayed when the forwarded party rings the *SUPERSET* telephone or is answered at the console. In a multi-hop forwarding situation, the first hop forwarder is displayed.
- If a caller is a trunk or a party with a single trunk on soft hold, System Option 21 (Incoming to Outgoing Call Forward) is checked; otherwise, check the COS Option 208 (Call Forward - External) of the forwarder, console or extension is externally call forwarded, the forwarding continues only if System Option 21 (Incoming to Outgoing Call Forward) is enabled.
- *SUPERSET* telephone keys with a ring type of DELAY, or multicall DELAY RING line appearances of extension numbers, will not ring if the extension has a DELAY RING timer active (controlled by COS option 263 Delay Ring Timer) which has a longer duration than the COS option 253 (Forwarding - Don't Answer timer). When both options are active, both timers begin at the same time. If the delay ring timer expires first, the line will ring. If the forwarding timer expires first, the delay timer is canceled and ringing will not occur.
- When an incoming DID call is made to a set or station with COS 264 (Half Fwd NA timer for DID call when VM msg on), and when that set or station has Voice Mail messages waiting, then the Call Forward No Answer timer is shortened by half. This is a toll saving option, allowing users who are calling in to check their messages to hang up if the cfna timer is not shortened, knowing they have no messages to retrieve.
- Forwarding - I'm here can only be canceled from the extension that set it up originally by using the Cancel I'm Here feature access code; it can also be canceled by the attendant or by the forwarded party by dialing the CLEAR ALL FEATURES access code. The forwarded party can re-program the forwarding via normal forwarding programming or via the CLEAR ALL FEATURES access code.
- When a *SUPERSET* display telephone receives a forwarded call, the display shows the number or name of the set that forwarded the call.
- If COS Option 258 (Display Prime Line as Forwarder) is disabled, the logical line will appear as the forwarder for all types of forwarding.
- Trunk calls forwarded by a set, subattendant, or console with COS Option 321 (Ignore Call Forward After Transfer) enabled will not follow the destination station's forwarding programming and will instead recall the forwarding station.
- If call forwarding is enabled at a set, you hear broken dial tone followed by regular dial tone when you go off-hook.

Programming

Assign access codes to the desired types of forwarding in CDE Form 02 (Feature Access Codes):

- Feature Access Code 03 (Call Forwarding - All Calls) *
- Feature Access Code 04 (Call Forwarding - Internal Only) **
- Feature Access Code 05 (Call Forwarding - External Only) **
- Feature Access Code 06 (Call Forwarding - I'm Here)
- Feature Access Code 07 (Call Forwarding - Cancel I'm Here)

  * Available only if COS Option 260 (Internal / External Split Call Forward) is disabled.

  ** Available only if COS Option 260 (Internal / External Split Call Forward) is enabled.

Programming is done in two steps:

- Select one of five types of Call Forwarding - internal, external, both (int + ext), I'm Here, and Cancel I'm Here.
- Dial the feature access code.
- Dial one of four Condition codes under which a call is forwarded (does not apply for Call Forward - I'm Here and Cancel I'm Here):
  1 - always, 2 - busy, 3 - no answer, 4 - busy/no answer.

Enable System Option 21 (Incoming to Outgoing Call Forward) to allow trunks to be call forwarded externally.

Enable one or more of the following COS options as required:

- 206 Call Forwarding - Busy
- 207 Call Forwarding - Don't Answer
- 208 Call Forwarding - External
- 209 Call Forwarding - Follow Me
- 210 Call Forwarding - Inhibit on Dial-in Trunks
- 222 Call Forwarding - Inhibit on Hold Timeout
- 253 Call Forwarding - Don't Answer Timer
- 258 Display Prime As Forwarder
- 260 Internal / External Split Call Forward
- 262 Ignore Forward Busy with Free Appearance; (see *Call Forwarding Busy / Don't Answer*)
- 263 Delay Ring Timer (2 - 6 Rings)
- 264 Half Fwd NA timer for DID call when VM msg on

Operation

Operation varies depending upon the device type as described in the following sections.

Call Forwarding - Busy

Description This feature forwards all calls when the extension is busy. While the extension is idle, calls may be made and received normally.

Conditions The following conditions apply to this feature:

- Refer to conditions listed under the heading, *Call Forwarding.*
- A telephone in Do Not Disturb is considered busy for forwarding.

Programming Enable COS Option 206 (Call Forwarding - Busy) in the extension's COS.

Ensure Feature Access Code 03 (Call Forwarding - All Calls) has a feature access code programmed.

For feature key activation of call forwarding on a *SUPERSET 410* or *SUPERSET 420* telephone, program a CALL FORWARD feature key (see Feature Keys).

Enable COS Option 262 (Ignore Forward Busy with Free Appearance) to have Call Forward Busy ignored when there are any free line appearances of the station or set. The next free appearance will be rung. COS option 262 will also cause call forward busy to be ignored on sets using a line other than prime and which do not have appearances of their prime elsewhere. This applies only to the busy portion of Call Forward Busy /Don't Answer.

Operation Operation varies depending upon the device type as described below.

Industry-standard and *SUPERSET 401+* Telephones:

To set up Call Forwarding - Busy:

- Lift the handset - wait for dial tone.
- Dial the access code for Call Forwarding - All Calls.
- Dial the Condition code for the desired type of forwarding.
- Dial the number to which calls are to be forwarded.
- Dial tone returns.
- Hang up - the extension is available for normal use.

To cancel Forwarding (all types):

- Lift the handset - wait for dial tone.
- Dial Call Forwarding - All Calls feature access code.
- Hang up - the forwarding is canceled.

or

- Lift the handset - wait for dial tone.
- Dial the Clear All Features access code (see *Clear All Features*). This also cancels all forwarding at the telephone.

***SUPERSET 410* Telephones:**

To program Call Forwarding - Busy:

- Follow the same procedure as for industry-standard telephones; it is also possible to dial the codes and extension destination directly on the keypad and then press the SPEAKER ON/OFF key to go on-hook.

To activate or deactivate forwarding already programmed:

- Press the CALL FORWARD feature key, if programmed. The LCD indicator is dark (activated) to indicate forwarding active, clear to indicate not active.

***SUPERSET 420* Telephones:**

To display the current forwarding type and destination (split forwarding disabled):

- Press SUPERKEY.
- Press the NO softkey until CALL FORWARDING? appears in the display.
- Press the YES softkey. The display shows the current forwarding type and destination, if programmed.
- Press SUPERKEY to return to normal display.

To activate forwarding already programmed:

- Press the CALL FORWARD Feature key. The indicator beside the key turns on to indicate that Call Forwarding is activated. (Refer to Feature Keys for instructions on programming this feature key).
  or, if a CALL FORWARD Feature key isn't programmed,
- Press SUPERKEY.
- Press the NO softkey until CALL FORWARDING? appears in the display.
- Press the YES softkey.
- Press the CHANGE softkey.
- Press the TURNON softkey. An asterisk appears next to the destination to indicate that call forwarding is activated.
- Press SUPERKEY to return to normal display.

To deactivate forwarding:

- Press the CALL FORWARD feature key. The indicator beside the key turns off to indicate that call forwarding is deactivated.
  OR, if a CALL FORWARD feature key isn't programmed,
- Press SUPERKEY.
- Press the NO softkey until CALL FORWARDING? appears in the display.
- Press the YES softkey.
- Press the CHANGE softkey.
- Press the TURNOFF softkey. The asterisk next to the destination disappears.
- Press SUPERKEY to return to normal display.

To program Call Forwarding - Busy (split forwarding disabled):

- Press SUPERKEY.
- Press the NO softkey until CALL FORWARDING? appears in the display.
- Press the YES softkey. The display shows the current forwarding type and destination, if programmed.
- Press the CHANGE softkey.
- Press the PROG softkey.
- Press the NO softkey until WHEN SET'S BUSY? appears in the display.
- Press the YES softkey.
- Enter the destination number. If an invalid number is entered, the display shows INVALID. Use the ← softkey to back space and erase the entry. Enter the correct number (can be a speedcall key, if programmed).
- Press the SAVE softkey. The type of forwarding, the forwarding destination and an * appear momentarily in the display. The display then returns to showing the date and time. Call Forwarding is now programmed and enabled.

Note: The asterisk indicates that call forwarding is enabled.

SUPERSET 430 Telephones:

To display the current forwarding type and destination (split forwarding disabled):

- Press SUPERKEY.
- Find the FORWARDING softkey. (Use the MORE softkey).
- Press the FORWARDING softkey. The display shows the current forwarding type and destination, if programmed.
- Press SUPERKEY to return to normal display.

To activate forwarding already programmed (split forwarding disabled):

- Press the TURN FWD ON softkey. <FWD ON> appears in the upper right corner of the LCD.

To deactivate forwarding (split forwarding disabled):

- Press the TURN FWD OFF softkey. <FWD ON> disappears from the display.

To set up Forwarding - Busy (split forwarding disabled):

- Press SUPERKEY.
- Find the FORWARDING softkey. (Use the MORE softkey).
- Press the FORWARDING softkey. The display shows the current forwarding type and destination, if programmed.
- Press the CHANGE softkey. Softkeys appear for the types of forwarding enabled in the set's COS.
- Select Call Forwarding - Busy. The display requests the destination number.

- Enter the destination number. If an invalid number is entered, the display shows INVALID NUMBER. Use the ← softkey to back space and erase the entry. Enter the correct number (can be a speedcall key, if programmed) or press the CURRENT NO. softkey to restore the previously programmed destination, if available (only available when the current forward destination is anything but a speedcall).
- Press the SAVE/ON softkey. The display returns to normal. <FWD ON> appears in the top right corner.

Call Forwarding - Busy/Don't Answer

Description

This feature forwards all calls received when the extension is busy, or calls that are not answered within a selected time-out period. While the extension is idle, calls may be made and received normally.

Conditions

The following conditions apply to this feature:

- COS Option 260 (Split Forwarding) must be disabled.
- Refer to conditions listed under the heading, *Call Forwarding.*
- For Call Forwarding - Don't Answer, a caller camps on to the forwarding destination if allowed to campon; see *Campon* for audio details.
- Call Forwarding - Don't Answer has no effect when a *SUPERSET* telephone answers the call using the Auto-Answer feature.
- *SUPERSET* telephone keys with a ring type of DELAY, or multicall DELAY RING line appearances of extension numbers, will not ring if the extension has a DELAY RING timer active (controlled by COS option 263 Delay Ring Timer) which has a longer duration than the Forwarding - Don't Answer timer. When both options are active, both timers begin at the same time. If the delay ring timer expires first, the line will ring. If the forwarding timer expires first, the delay timer is canceled and ringing will not occur.

Programming

Enable COS Options 206 (Call Forwarding - Busy), and 207 (Call Forwarding - Don't Answer) in the extension's COS.

Enable COS Option 262 (Ignore Forward Busy with Free Appearance) to have Call Forward Busy ignored when there are any free line appearances of the station or set. The next free appearance will be rung. COS option 262 will also cause call forward busy to be ignored on sets using a line other than prime and which do not have appearances of their prime elsewhere. This applies only to the busy portion of Call Forward Busy /Don't Answer.

Ensure Feature Access Code 03 (Call Forwarding - All Calls) has a feature access code programmed.

For feature key activation of call forwarding on a *SUPERSET 410* or *SUPERSET 420* telephone, program a CALL FORWARD feature key. (See *Feature Keys.*)

Operation

Operation varies depending upon the device type as described below:

Industry-standard and *SUPERSET 401+* Telephones:

To set up Call Forwarding - Busy/Don't Answer:

- Lift the handset - wait for dial tone.
- Dial the access code for Call Forwarding - All Calls.
- Dial the Condition code for the desired type of forwarding Busy/Don't Answer.
- Dial the number to which calls are to be forwarded.
- Dial tone returns.
- Hang up - the extension is available for normal use.

To cancel forwarding (all types):

- Lift the handset - wait for dial tone.
- Dial the Call Forwarding - All Calls access code.
- Hang up - the forwarding is canceled.

or

- Lift the handset - wait for dial tone.
- Dial the Clear All Features access code (see *Clear All Features)*. This also cancels all forwarding at the telephone.

***SUPERSET 410* Telephones:**

To set up Call Forwarding - Busy/Don't Answer:

- Follow the same procedure as for industry-standard telephones; it is also possible to dial the codes and extension destination directly on the keypad and then press the SPEAKER ON/OFF key to go on-hook.

To activate or deactivate forwarding already programmed:

- Press the CALL FORWARD feature key, if programmed. The LCD indicator is dark (activated) to indicate forwarding active, clear to indicate not active.

***SUPERSET 420* Telephones:**

To display the current forwarding type and destination:

- Press SUPERKEY.
- Press the NO softkey until CALL FORWARDING? appears in the display.
- Press the YES softkey. The display shows the current forwarding type and destination, if programmed.
- Press SUPERKEY to return to normal display.

To activate forwarding already programmed (split forwarding disabled):

- Press the CALL FORWARD feature key. The indicator beside the key turns on to indicate that Call Forwarding is activated. (Refer to Feature Keys for instructions on programming this feature key.)
  or, if a CALL FORWARD feature key isn't programmed,
- Press SUPERKEY.
- Press the NO softkey until CALL FORWARDING? appears in the display.
- Press the YES softkey.
- Press the CHANGE softkey.
- Press the TURNON softkey. An asterisk appears next to the destination to indicate that call forwarding is activated.
- Press SUPERKEY to return to normal display.

To deactivate forwarding (split forwarding disabled):

- Press the CALL FORWARD feature key. The indicator beside the key turns off to indicate that Call Forwarding is deactivated.

  or, if a CALL FORWARD feature key isn't programmed,
- Press SUPERKEY.
- Press the NO softkey until CALL FORWARDING? appears in the display.
- Press the YES softkey.
- Press the CHANGE softkey.
- Press the TURNOFF softkey. The asterisk next to the destination disappears.
- Press SUPERKEY to return to normal display.

To program Call Forwarding - Busy/Don't Answer:

- Press SUPERKEY.
- Press the NO softkey until CALL FORWARDING? appears in the display.
- Press the YES softkey. The display shows the current forwarding type and destination, if programmed.
- Press the CHANGE softkey.
- Press the PROG softkey.
- Press the NO softkey until BUSY/NO ANSWER? appears in the display.
- Press the YES softkey.
- Enter the destination number. If an invalid number is entered, the display shows INVALID. Use the ← softkey to back space and erase the entry. Enter the correct number (can be a speedcall key, if programmed).
- Press the SAVE softkey. The type of forwarding, the forwarding destination and an * appear momentarily in the display. The display then returns to showing the date and time. Call Forwarding is now programmed and enabled.

Note: The asterisk indicates that call forwarding is enabled.

***SUPERSET 430* Telephones:**

To display the current forwarding type and destination:

- Press SUPERKEY.
- Find the FORWARDING softkey. (Use the MORE softkey.)
- Press the FORWARDING softkey. The display shows the current forwarding type and destination, if programmed.
- Press SUPERKEY to return to normal display.

To activate forwarding already programmed:

- Press the TURN FWD ON softkey. <FWD ON> appears in the upper right corner of the LCD.

To deactivate forwarding:

- Press the TURN FWD OFF softkey. <FWD ON> disappears.

To set up Call Forwarding - Busy/Don't Answer:

- Press SUPERKEY.
- Find the FORWARDING softkey. (Use the MORE softkey.)
- Press the FORWARDING softkey. The display shows the current forwarding type and destination, if programmed.
- Press the CHANGE softkey. Softkeys appear for the types of forwarding enabled in the set's COS.
- Select Call Forwarding - Busy/Don't Answer. The display requests the destination number.
- Enter the destination number. If an invalid number is entered, the display shows INVALID NUMBER. Use the ← softkey to back space and erase the entry. Enter the correct number (can be a speedcall key, if programmed) or press the CURRENT NO. softkey to restore the previously programmed destination, if available (only available when the current forward destination is anything but a speedcall).
- Press the SAVE/ON softkey. The display returns to normal. <FWD ON> appears in the top right corner.

Call Forwarding - Follow Me

Description

This type of forwarding is unconditional. All calls are forwarded to the programmed destination. The number to which the calls are forwarded is the only party that may call the forwarding extension while Forwarding - Follow Me is active. The extension may originate calls in the normal manner.

Conditions

The following conditions apply to this feature:

- COS Option 260 (Split Forwarding) must be disabled.
- Refer to conditions listed under the heading, *Call Forwarding*.

Programming

Enable COS Option 209 (Call Forwarding - Follow Me) in the extension's COS.

Ensure Feature Access Code 03 (Call Forwarding - All Calls) has a feature access code programmed.

For feature key activation of call forwarding on a *SUPERSET 410* or *SUPERSET 420* telephone, program a CALL FORWARD feature key. (See *Feature Keys*.)

Operation

Operation varies depending upon the device type as described below.

Industry-standard and *SUPERSET 401+* Telephones:

To set up Call Forwarding - Follow Me:

- Lift the handset - wait for dial tone.
- Dial the access code for Call Forwarding - All Calls.
- Dial the condition code for the desired type of forwarding.
- Dial the number to which calls are to be forwarded.
- Dial tone returns.
- Hang up - the extension is available for normal use.

To cancel forwarding (all types):

- Lift the handset - wait for dial tone.
- Dial the Call Forwarding - All Calls access code.
- Hang up - the forwarding is canceled.

OR

- Lift the handset - wait for dial tone.
- Dial the Clear All Features access code (see *Clear All Features*). This also cancels all forwarding at the telephone.

***SUPERSET 410* Telephones:**

To set up Call Forwarding - Follow Me:

- Follow the same procedure as for industry-standard telephones; it is also possible to dial the codes and extension destination directly on the keypad and then press the **Speaker On/Off** key to go on-hook.

To activate or deactivate forwarding already programmed:

- Press the CALL FORWARD feature key, if programmed. The LCD indicator is dark (activated) to indicate forwarding active, clear to indicate not active.

***SUPERSET 420* Telephones:**

To display the current forwarding type and destination:

- Press SUPERKEY.

- Press the NO softkey until CALL FORWARDING? appears in the display.
- Press the YES softkey. The display shows the current forwarding type and destination, if programmed.
- Press SUPERKEY to return to normal display.

To activate forwarding already programmed (split forwarding disabled):

- Press the CALL FORWARD feature key. The indicator beside the key turns on to indicate that Call Forwarding is activated. (Refer to Feature Keys for instructions on programming this feature key.)

  or, if a CALL FORWARD feature key isn't programmed,
- Press SUPERKEY.
- Press the NO softkey until CALL FORWARDING? appears in the display.
- Press the YES softkey.
- Press the CHANGE softkey.
- Press the TURNON softkey. An asterisk appears next to the destination to indicate that call forwarding is activated.
- Press SUPERKEY to return to normal display.

To deactivate forwarding (split forwarding disabled):

- Press the CALL FORWARD feature key. The indicator beside the key turns off to indicate that Call Forwarding is deactivated.

  or, if a CALL FORWARD feature key isn't programmed,
- Press SUPERKEY.
- Press the NO softkey until CALL FORWARDING? appears in the display.
- Press the YES softkey.
- Press the CHANGE softkey.
- Press the TURNOFF softkey. The asterisk next to the destination disappears.
- Press SUPERKEY to return to normal display.

To program Call Forwarding - Follow Me:

- Press SUPERKEY.
- Press the NO softkey until CALL FORWARDING? appears in the display.
- Press the YES softkey. The display shows the current forwarding type and destination, if programmed.
- Press the CHANGE softkey.
- Press the PROG softkey.
- Press the NO softkey until ALWAYS FORWARD? appears in the display.
- Press the YES softkey.

- Enter the destination number. If an invalid number is entered, the display shows INVALID. Use the ← softkey to back space and erase the entry. Enter the correct number (can be a speedcall key, if programmed).
- Press the SAVE softkey. The type of forwarding, the forwarding destination and an * appear momentarily in the display. The display then returns to showing the date and time. Call forwarding is now programmed and enabled.

Note: The asterisk indicates that call forwarding is enabled.

SUPERSET 430 **Telephones:**

To display the current forwarding type and destination:

- Press SUPERKEY.
- Find the FORWARDING softkey. (Use the MORE softkey.)
- Press the FORWARDING softkey. The display shows the current forwarding type and destination, if programmed.
- Press SUPERKEY to return to normal display.

To activate forwarding already programmed:

- Press the TURN FWD ON softkey. <FWD ON> appears in the upper right corner of the LCD.

To deactivate forwarding:

- Press the TURN FWD OFF softkey. <FWD ON> disappears from the display.

To set up Forwarding - Follow Me:

- Press SUPERKEY.
- Find the FORWARDING softkey. (Use the MORE softkey.)
- Press the FORWARDING softkey. The display shows the current forwarding type and destination, if programmed.
- Press the CHANGE softkey. Softkeys appear for the types of forwarding enabled in the set's COS.
- Select ALWAYS. The display requests the destination number.
- Enter the destination number. If an invalid number is entered, the display shows INVALID NUMBER. Use the ← softkey to back space and erase the entry. Enter the correct number (can be a speedcall key, if programmed) or press the CURRENT NO. softkey to restore the previously programmed destination, if available (only available when the current forward destination is anything but a speedcall).
- Press the SAVE/ON softkey. The display returns to normal. <FWD ON> appears in the top right corner.

Call Forwarding - I'm Here

Description

This type of forwarding operates the same as Forwarding - Follow Me, but it is activated from another extension. All calls are forwarded to the new location. The forwarded extension can originate calls in the normal manner.

Conditions

The following conditions apply to this feature:

- COS Option 234 (Never a Forwardee) must be disabled for this extension.
- For a display telephone to receive the I'm Here message prompt, it must have COS Option 234 (Never a Forwardee) disabled and COS Option 209 (Call Forward - Follow Me) enabled.

Programming

Enable COS Option 209 (Call Forwarding - Follow Me) in the set's COS.

Program access codes for Feature Access Code 06 (Call Forwarding - I'm Here) and Feature Access Code 07 (Call Forwarding - Cancel I'm Here).

Operation

Operation varies depending upon the device types as described below.

Industry-standard and *SUPERSET 401+* Telephones:

To select Call Forwarding - I'm Here:

- At another extension, lift the handset - wait for dial tone.
- Dial the Call Forwarding - I'm Here access code.
- Dial your own extension number - dial tone returns.
- Hang up - the extension is available for normal use.

To cancel Call Forwarding - I'm Here from the telephone which set Call Forward - I'm Here:

- Lift the handset - wait for dial tone.
- Dial the Cancel I'm Here access code.
- Dial the extension whose forwarding is to be canceled; hang up - the forwarding is canceled for that extension.

OR

- Hang up - all I'm Here forwarding to this extension is canceled.

***SUPERSET 410* Telephones:**

To select Call Forwarding - I'm Here:

- At another extension, dial the Call Forwarding - I'm Here access code.
- Dial your own extension number - dial tone returns.
- Press the SPEAKER ON/OFF key - the extension is available for normal use.

To cancel Call Forwarding - I'm Here from the telephone which set Call Forward - I'm Here:

- Dial the Cancel I'm Here access code.

- Dial the extension whose forwarding is to be canceled and press the SPEAKER ON/OFF key. The forwarding is canceled for that extension.

OR

- Press the SPEAKER ON/OFF key to cancel all I'm Here forwarding to this extension.

***SUPERSET 420* Telephones:**

To set up and turn on Call Forwarding - I'm Here:

- At the destination *SUPERSET 420* telephone, press SUPERKEY.
- Press the NO softkey until CALL FORWARDING? appears in the display.
- Press the YES softkey.
- Press the NO softkey until I'M HERE? appears in the display.
- Press the YES softkey.
- Enter extension number from which calls are to be forwarded. If an invalid number is entered, the display shows INVALID. Use the ← softkey to back space and erase the entry.
- Press the SAVE softkey. The extension number and name from which your calls will be forwarded, appear momentarily in the display. The display then returns to showing the date and time. Call Forwarding is now programmed and enabled.

To cancel Call Forwarding I'm Here:

- Press the SPEAKER key.
- Dial the Cancel I'm Here access code.
- Dial the extension whose forwarding is to be canceled and press the SPEAKER key. The forwarding is canceled for that extension.

OR

- Press the SPEAKER key to cancel all I'm Here forwarding to this extension.

***SUPERSET 430* Telephones:**

To set up and activate Call Forwarding - I'm Here, using softkeys:

- At the destination *SUPERSET 430* telephone, find and press the FORWARDING softkey.
- Press CHANGE softkey (necessary only if Split Forwarding is disabled).
- Press TO ME softkey.
- Enter extension number from which calls are to be forwarded.
- Press SAVE/ON softkey.
- The set displays NOW FROM XXXXX ALWAYS.

To cancel Call Forwarding I'm Here:

- Press the SPEAKER ON/OFF key.

- Dial the Cancel I'm Here access code.
- Dial the extension whose forwarding is to be canceled and press the SPEAKER ON/OFF key. The forwarding is canceled for that extension.

OR

- Press the SPEAKER ON/OFF key to cancel all I'm Here forwarding to this extension.

Call Forwarding - Don't Answer

Description

This feature forwards all calls that are not answered within a selected time-out period. Calls may be made and received normally.

Conditions

The following conditions apply to this feature:

- For Call Forwarding - Don't Answer, a caller camps on to the forwarding destination if allowed to campon; see *Campon* for audio details.
- Call Forwarding - Don't Answer has no effect when a *SUPERSET* telephone answers the call using the Auto-Answer feature.
- *SUPERSET* telephone key or multicall DELAY RING line appearances of extension numbers do not ring if the extension has activated Call Forwarding - Don't Answer and the call is forwarded. The delay ring timer is the same as the call forward - don't answer timer. However, the *SUPERSET* telephone provides visual indication of the incoming call. The *SUPERSET* telephone user can answer the call before the forwarding occurs. The exception to this condition occurs when the DELAY RING is controlled by COS option 263 Delay Ring Timer in the called party's COS. In this instance, both timers begin at the same time. If the delay ring timer expires first, the line will ring. If the forwarding timer expires first, the delay timer is canceled and ringing will not occur.
- Refer to conditions listed under the heading, *Call Forwarding.*

Programming

Enable COS Option 207 (Call Forwarding - Don't Answer) in the extension's COS:

Ensure Feature Access Code 03 (Call Forwarding - All Calls) has a feature access code programmed.

For feature key activation of call forwarding on a *SUPERSET 410* or *SUPERSET 420* telephone, program a CALL FORWARD feature key. (See Feature Keys.)

Operation

Operation varies depending upon the device type as described below.

Industry-standard and *SUPERSET 401+* Telephones:

To set up Call Forwarding - Don't Answer:

- Lift the handset - wait for dial tone.

- Dial the access code for Call Forwarding - Don't Answer.
- Dial the Condition code for No Answer.
- Dial the number to which calls are to be forwarded.
- Dial tone returns.
- Hang up - the extension is available for normal use.

To cancel forwarding (all types):

- Lift the handset - wait for dial tone.
- Dial Call Forwarding All Calls access code.
- Hang up - the forwarding is canceled, or,
- Lift the handset - wait for dial tone.
- Dial the Clear All Features access code (see *Clear All Features*). This also cancels all forwarding at the telephone.

***SUPERSET 410* Telephones:**

To set up Call Forwarding - Don't Answer:

- Follow the same procedure as for industry-standard telephones; it is also possible to dial the codes and extension destination directly on the keypad and then press the **Speaker On/Off** key to go on-hook.

To activate or deactivate forwarding already programmed:

- Press the CALL FORWARD feature key, if programmed. The LCD indicator is dark (activated) to indicate forwarding active, clear to indicate not active.

***SUPERSET 420* Telephones:**

To display the current forwarding type and destination:

- Press SUPERKEY.
- Press the NO softkey until CALL FORWARDING? appears in the display.
- Press the YES softkey. The display shows the current forwarding type and destination, if programmed.
- Press SUPERKEY to return to normal display.

To activate forwarding already programmed:

- Press the CALL FORWARD Feature key. The indicator beside the key turns on to indicate that Call Forwarding is activated. (Refer to *Feature Keys* for instructions on programming this feature key).

OR, if a CALL FORWARD feature key isn't programmed,

- Press SUPERKEY.
- Press the NO softkey until CALL FORWARDING? appears in the display.
- Press the YES softkey.
- Press the CHANGE softkey.

- Press the TURNON softkey. An asterisk appears next to the destination to indicate that call forwarding is activated.
- Press SUPERKEY to return to normal display.

To deactivate forwarding:

- Press the CALL FORWARD feature key. The indicator beside the key turns off to indicate that call forwarding is deactivated.

OR, if a CALL FORWARD Feature key isn't programmed,

- Press SUPERKEY.
- Press the NO softkey until CALL FORWARDING? appears in the display.
- Press the YES softkey.
- Press the CHANGE softkey.
- Press the TURNOFF softkey. The asterisk next to the destination disappears.
- Press SUPERKEY to return to normal display.

To program Call Forwarding - Don't Answer:

- Press SUPERKEY.
- Press the NO softkey until CALL FORWARDING? appears in the display.
- Press the YES softkey. The display shows the current forwarding type and destination, if programmed.
- Press the CHANGE softkey.
- Press the PROGRAM softkey.
- Press the NO softkey until WHEN NO ANSWER? appears in the display.
- Press the YES softkey.
- Enter the destination number. If an invalid number is entered, the display shows INVALID. Use the ← softkey to back space and erase the entry. Enter the correct number (can be a speedcall key, if programmed).
- Press the SAVE softkey. The type of forwarding, the forwarding destination, and an * appear momentarily in the display. The display then returns to showing the date and time. Call forwarding is now programmed and enabled.

Note: The asterisk indicates that call forwarding is enabled.

SUPERSET 430 Telephones

To display the current forwarding type and destination:

- Press SUPERKEY.
- Find the FORWARDING softkey. (Use the MORE softkey).
- Press the FORWARDING softkey. The display shows the current forwarding type and destination, if programmed.
- Press SUPERKEY to return to normal display.

To activate forwarding already programmed:

- Press the TURN FWD ON softkey. <FWD ON> appears in the upper right corner of the LCD.

To deactivate forwarding:

- Press the TURN FWD OFF softkey. <FWD ON> disappears from the display.

To set up Call Forwarding - Don't Answer:

- Press SUPERKEY.
- Find the FORWARDING softkey. (Use the MORE softkey).
- Press the FORWARDING softkey. The display shows the current forwarding type and destination, if programmed.
- Press the CHANGE softkey. Softkeys appear for the types of forwarding enabled in the set's COS. This only applies if Split Forwarding is disabled.
- Select Call Forwarding - Don't Answer. The display requests the destination number.
- Enter the destination number. If an invalid number is entered, the display shows INVALID NUMBER. Use the ← softkey to back space and erase the entry. Enter the correct number (can be a speedcall key, if programmed) or press the CURRENT NO. softkey to restore the previously programmed destination, (if available).
- Press the SAVE/ON or SAVE/OFF softkey. The display returns to normal. <FWD ON> appears in the top right corner only if either internal or external forwarding is enabled.

Call Forwarding - External

Description

This feature forwards all calls received based on one of the conditions selected from above, to a personal speed call key, system abbreviated dial number, or a key system personal speedcall.

Conditions

The following conditions apply to this feature:

- Call Forwarding - External requires a receiver for dialing. If one is not available then forwarding is ignored in Call Forward - Don't Answer. If it is done during a reroute, the caller is dropped or given reorder tone.
- An extension with COS Option 200 (Account Code, Forced Entry - External Calls) in its COS cannot set up the Call Forwarding - External feature.
- An extension must have COS Option 245 (Abbreviated Dialing Access) enabled, COS Option 208 (Call Forwarding External) enabled, and COS Option 200 (Account Code, Forced Entry - External Calls) disabled in its COS to be able to program call forwarding to system abbreviated

dial numbers. The option is not needed for callers which are forwarded to abbreviated dial numbers.

- Toll Control applies to the calling party for Call Forwarding - External to personal speed call keys or to key system personal speedcalls.
- Toll Control is not done for forwarding to external numbers using the system abbreviated dial feature.
- No Toll Control checking is done for CO trunks which are externally call forwarded.
- A trunk or a party with a single party trunk on hold can only be forwarded externally if System Option 21 (Incoming to Outgoing Call Forward) is enabled.
- An extension cannot set up forwarding to an external number unless it has COS Option 208 (Call Forwarding - External) in its COS.
- Once a call is forwarded externally, no more forwarding hops occur.
- Campon to busy trunk groups after forwarding external is possible if the caller is allowed to campon to trunk groups; see *Campon*.
- When the attendant calls an extension that is forwarded to an external number, the display shows the external number and indicates the extension which forwarded the call. If the speed call or abbreviated dial number is private, text indicating a private number appears on the display instead of the external number. The console can display the number if it has COS Option 110 (Attendant Abbr. Dial Confidential Number Display) enabled in its COS.
- Consoles and extensions can transfer a party on consultation hold as soon as forwarding to an external number starts, at any point in dialing the external number.
- The reference is to the stored speedcall key number or the personal or system abbreviated dial number. If this stored number is changed, then the forwarding destination changes also.

Programming

Enable COS Option 208 (Call Forwarding - External) in the calling party's COS (extension's COS or trunk's COS).

For external call forwarding involving two trunks, ensure that the incoming and outgoing trunk can be connected together. CDE Form 30 - Device Interconnection Table specifies which devices can be connected together.

Enable COS Option 245 (Abbreviated Dialing Access) for the extension to allow forwarding to system abbreviated dial numbers or to key system personal speedcalls.

Disable COS Option 200 (Account Code, Forced - External Calls) in the extension's COS.

Enable System Option 21 (Incoming to Outgoing Call Forward) to allow trunks to be externally call forwarded.

Ensure Feature Access Code 03 (Call Forwarding - All Calls) has a feature access code programmed.

For feature key activation of call forwarding on a *SUPERSET 410* or *SUPERSET 420* telephone, program a CALL FORWARD feature key. (See Feature Keys.)

Operation

Operation varies depending upon the device type as described below.

Industry-standard and *SUPERSET 401+* Telephones:

To set up Call Forwarding - External at an extension:

- Lift the handset - dial tone is heard.
- Dial the Call Forward - All Calls access code.
- Dial the condition code for the desired type of forwarding.
- Dial the abbreviated dial index number, which contains the external telephone number to which calls are to be forwarded. Dial tone is returned if the above codes are valid; reorder tone is returned if the codes are invalid.
- Replace the handset - Call Forwarding - External is now active.

To cancel Call Forwarding - External at an extension:

- Lift the handset - dial tone is returned.
- Dial the Call Forward - All Calls access code (assuming Split Call Forwarding is disabled).
- Replace the handset - Call Forwarding - External is now inactive.

OR

- Lift the Handset - dial tone is returned.
- Dial the Clear All feature access codes - Dial tone is returned.
- Replace the handset.

***SUPERSET 410* Telephones:**

To set up Forwarding - External:

- Follow the same procedure as for industry-standard telephones; it is also possible to dial the codes and extension destination directly on the keypad and then press the SPEAKER ON/OFF key to go on-hook.

To activate or deactivate forwarding already programmed:

- Press the CALL FORWARD feature key, if programmed. The LCD indicator is dark (activated) to indicate forwarding active, clear to indicate not active.

***SUPERSET 420* Telephones:**

To display the current forwarding type and external destination:

- Press SUPERKEY.
- Press the NO softkey until CALL FORWARDING? appears in the display.

- Press the YES softkey. The display shows the current forwarding type and external destination, if programmed.
- Press SUPERKEY to return to normal display.

To activate forwarding already programmed:

- Press the CALL FORWARD feature key. The indicator beside the key turns on to indicate that call forwarding is activated. (Refer to feature keys for instructions on programming this feature key.)

OR, if a CALL FORWARD feature key isn't programmed,

- Press SUPERKEY.
- Press the NO softkey until CALL FORWARDING? appears in the display.
- Press the YES softkey.
- Press the CHANGE softkey.
- Press the TURNON softkey. An asterisk appears next to the destination to indicate that call forwarding is activated.
- Press SUPERKEY to return to normal display.

To deactivate forwarding to an external destination:

- Press the CALL FORWARD feature key. The indicator beside the key turns off to indicate that call forwarding is deactivated.

OR, if a CALL FORWARD feature key isn't programmed,

- Press SUPERKEY.
- Press the NO softkey until CALL FORWARDING? appears in the display.
- Press the YES softkey.
- Press the CHANGE softkey.
- Press the TURNOFF softkey. The asterisk next to the destination disappears.
- Press SUPERKEY to return to normal display.

To program the set to forward all calls to an external number:

- Press SUPERKEY.
- Press the NO softkey until CALL FORWARDING? appears in the display.
- Press the YES softkey. The display shows the current forwarding type and destination, if programmed.
- Press the CHANGE softkey.
- Press the PROGRAM softkey.
- Press the NO softkey until the desired type of call forwarding appears in the display.
- Press the YES softkey.
- Enter the destination number, or press a SPEED CALL key on the *SUPERSET* telephone or associated PKM. If an invalid number is

entered, the display shows INVALID NUMBER. Use the ← softkey to back space and erase the entry.

- Press the SAVE softkey. The type of forwarding, the forwarding destination, and an * appear momentarily in the display. The display then returns to showing the date and time. Call forwarding is now programmed and enabled.

Note: The asterisk indicates that call forwarding is enabled.

***SUPERSET 430* Telephones:**

To display the current forwarding type and destination:

- Press SUPERKEY.
- Find the FORWARDING softkey. (Use the MORE softkey).
- Press the FORWARDING softkey. The display shows the current forwarding type and destination, if programmed.
- Press SUPERKEY to return to normal display.

To activate forwarding already programmed:

- Press the TURN FWD ON softkey. <FWD ON> appears in the upper right corner of the LCD.

To deactivate forwarding:

- Press the TURN FWD OFF softkey. <FWD ON> disappears from the display.

To set up Call Forwarding - External:

- Press SUPERKEY.
- Find the FORWARDING softkey. (Use the MORE softkey).
- Press the FORWARDING softkey. The display shows the current forwarding type and destination, if programmed.
- Press the CHANGE softkey. Softkeys appear for the types of forwarding enabled in the set's COS. This only applies if Split Forwarding is disabled.
- Select the required type. The display requests the destination number.
- Enter the destination number, or press a SPEED CALL key on the *SUPERSET* telephone or associated PKM. If an invalid number is entered, the display shows INVALID NUMBER. Use the ← softkey to back space and erase the entry. Enter the correct number (can be a speedcall key, if programmed) or press the CURRENT NO. softkey to restore the previously programmed destination (if available).
- Press the SAVE/ON softkey. The display returns to normal. <FWD ON> appears in the top right corner. If Split Forwarding is enabled, the SAVE/OFF softkey is also available.

Call Forwarding - Internal / External Split

Description

Calls may be forwarded to different destinations according to the source of the call - either an internal call from within the PABX or an external call from outside of the PABX.

Conditions

The following conditions apply to this feature:

- COS Option 260 (Internal / External Split Call Forward) must be enabled for the extension to allow forwarding according to the source of the call.
- For call forwarding, all devices are treated as internal calls unless COS Option 709 (Follow External Call Forward) of the calling device is enabled.
- A feature key which turns forwarding ON or OFF, turns both internal and external on or off together.
- To allow call forwarding to an external destination, COS Option 208 (Call Forwarding - External) must be enabled.

Programming

Enable COS Option 260 (Internal / External Split Call Forward) for the extension to allow forwarding according to the source of the call.

Enable COS Option 709 (Follow External Call Forward) for devices to be treated as external calls.

Enable the desired call forwarding type in the extension's COS.

Ensure Feature Access Code 04 (Call Forwarding - Internal Only) and Feature Access Code 05 (Call Forwarding -External Calls Only) have feature access codes programmed.

For feature key activation of call forwarding on a *SUPERSET 410* or *SUPERSET 420* telephone, program a CALL FORWARD feature key. (See *Feature Keys*.)

Operation

Operation varies depending upon the device type as described below.

Industry-standard and *SUPERSET 401+* Telephones:

To set up call forwarding for internally originated calls:

- Lift the handset - wait for dial tone.
- Dial Access code 04 - Call Forwarding - Internal Calls Only.
- Dial the condition code for the desired type of forwarding.
- Dial the number to which calls are to be forwarded.
- Dial tone returns.
- Hang up - the extension is available for normal use.

To set up call forwarding for externally originated calls:

- Lift the handset - wait for dial tone.
- Dial Access code 05 - Call Forwarding - External Calls Only
- Dial the condition code for the desired type of forwarding.

- Dial the number to which calls are to be forwarded.
- Dial tone returns.
- Hang up - the extension is available for normal use.

To cancel forwarding (all types):

- Lift the handset - wait for dial tone.
- Dial the external forwarding access code.
- Hang up - external forwarding is canceled.
- Dial the internal forwarding access code.
- Hang up - internal forwarding is canceled.

or

- Lift the handset - wait for dial tone.
- Dial the Clear All Features access code (see *Clear All Features*). This also cancels all forwarding at the telephone.

***SUPERSET 410* Telephones:**

To set up Split Call Forwarding (all types except I'm Here):

- Follow the same procedure as for industry-standard telephones; it is also possible to dial the codes and extension destination directly on the keypad and then press the SPEAKER ON/OFF key to go on-hook.

To activate or deactivate forwarding already programmed:

- Press the CALL FORWARD feature key, if programmed. The LCD indicator is dark (activated) to indicate either internal or external forwarding active, clear to indicate not active.

***SUPERSET 420* Telephones:**

To display the current forwarding type and destination:

- Press SUPERKEY.
- Press the NO softkey until CALL FORWARDING? appears in the display.
- Press the YES softkey. The display shows the current forwarding type and destination, if programmed.
- EXTERNAL CALLS? * appears in the display. The asterisk (*) is present only if call forwarding is on.
- Press the REVIEW softkey to display the current call forwarding setting for the external incoming calls that arrive at your set.
- Press the NEXT softkey. INTERNAL CALLS? * appears in the display. The asterisk (*) is present only if call forwarding is on.
- Press the REVIEW softkey to display the current call forwarding setting for the internal calls that arrive at your set.
- Press SUPERKEY to return to normal display.

To activate call forwarding if it is already programmed:

- Press the CALL FORWARD feature key. The indicator beside the key turns on to indicate that call forwarding is activated. Note that if both external and internal call forwarding are programmed, both will be enabled. (Refer to *Feature Keys* for instructions on programming this feature key.)

  or, you can activate either external forwarding, internal forwarding, or both using the softkeys.

  To activate external forwarding using the softkeys:

- Press SUPERKEY.
- Press the NO softkey until CALL FORWARDING? appears in the display.
- Press the YES softkey.
- EXTERNAL CALLS? appears in the display.
- Press the CHANGE softkey. The current call forwarding setting for external incoming calls is displayed
- Press the TURNON softkey.
- Press SUPERKEY to return to normal display.

  To activate internal forwarding using the softkeys:

- Press SUPERKEY.
- Press the NO softkey until CALL FORWARDING? appears in the display.
- Press the YES softkey.
- EXTERNAL CALLS? appears in the display.
- Press the NO softkey. INTERNAL CALLS? appears in the display.
- Press the CHANGE softkey. The current call forwarding setting for internal calls is displayed.
- Press the TURNON softkey.
- Press SUPERKEY to return to normal display.

To deactivate forwarding:

- Press the CALL FORWARD feature key. The indicator beside the key turns off to indicate that call forwarding is deactivated. Note that if both external and internal call forwarding are programmed, both will be turned off.

  or, you can deactivate either external forwarding, internal forwarding, or both using the softkeys.

  To deactivate external forwarding using the softkeys:

- Press SUPERKEY.
- Press the NO softkey until CALL FORWARDING? appears in the display.
- Press the YES softkey.

- EXTERNAL CALLS? appears in the display.
- Press the CHANGE softkey. The current call forwarding setting for external incoming calls is displayed.
- Press the TURNOFF softkey.
- Press SUPERKEY to return to normal display.

To deactivate internal forwarding using the softkeys:

- Press SUPERKEY.
- Press the NO softkey until CALL FORWARDING? appears in the display.
- Press the YES softkey.
- EXTERNAL CALLS? appears in the display.
- Press the NO softkey. INTERNAL CALLS? appears in the display.
- Press the CHANGE softkey. The current call forwarding setting for internal calls is displayed.
- Press the TURNOFF softkey.
- Press SUPERKEY to return to normal display.

To program Call Forwarding - Internal/External Split:

To program forwarding of your external calls:

- Press SUPERKEY.
- Press the NO softkey until CALL FORWARDING? appears in the display.
- Press the YES softkey. EXTERNAL CALLS? * appears in the display. The asterisk (*) is present only if call forwarding is on.
- Press the CHANGE softkey.
- Press the PROGRAM softkey.
- Press the NO softkey until the desired call forwarding type appears in the display. When the desired type is displayed, press the YES softkey.
- Enter the number of the destination, or press a SPEED CALL key on the *SUPERSET* telephone or associated PKM. If an invalid number is entered, the display shows INVALID NUMBER. Use the ← softkey to back space and erase the entry.
- Press the SAVE softkey. The type of forwarding and the forwarding destination appear momentarily in the display. Then, the display shows INTERNAL CALLS?
- Press SUPERKEY to return to normal display.

To program the forwarding of your internal calls:

- Press SUPERKEY.
- Press the NO softkey until CALL FORWARDING? appears in the display.
- Press the YES softkey. EXTERNAL CALLS? * appears. The asterisk (*) is present only if call forwarding is on.
- Press the NO softkey. INTERNAL CALLS? * appears. The asterisk (*) is present only if call forwarding is on.

- Press the CHANGE softkey.
- Press the PROGRAM softkey.
- Press the NO softkey until the desired call forwarding type appears in the display. When the desired type is displayed, press the YES softkey.
- Enter the destination number. If an invalid number is entered, the display shows INVALID. Use the ← softkey to back space and erase the entry.
- Press the SAVE softkey. The type of forwarding, the forwarding destination, and an * appear momentarily in the display. Call forwarding of your internal incoming calls is now programmed and enabled.
  Note: The asterisk indicates that call forwarding is enabled.
- Press SUPERKEY to return to normal display.

SUPERSET 430 **Telephones:**

To display the current forwarding type and destination:

- Press SUPERKEY.
- Find the FORWARDING softkey. (Use the MORE softkey.)
- Press the FORWARDING softkey. The display shows both EXTERNAL and INTERNAL forwarding type and displays the condition keys.
- Press SUPERKEY to return to normal display.

To activate forwarding already programmed:

- Press the TURN FWD ON softkey. <FWD ON> appears in the upper right corner of the LCD.

To deactivate forwarding:

- Press the TURN FWD OFF softkey. <FWD ON> disappears.

To set up forwarding:

- Press SUPERKEY.
- Find the FORWARDING softkey. (Use the MORE softkey.)
- Press the FORWARDING softkey. The display shows the current forwarding types and destinations, for external and internal calls. Softkeys appear for the types of forwarding enabled in the set's COS.
- Select the desired type of forwarding by using the appropriate softkey.
- Settings according to the selected type are displayed for both EXTERNAL and INTERNAL. Select the EXTERNAL or INTERNAL softkey to change the programming.
- Enter the destination number. If an invalid number is entered, the display shows INVALID NUMBER. Use the ← softkey to back space and erase the entry. Enter the correct number (can be a speedcall key, if programmed) or press the CURRENT NO. softkey to restore the previously programmed destination (if available).
- Press the SAVE/ON or SAVE/OFF softkey. The display returns to allow the user to program the other type of forwarding, i.e., internal or external. SAVE/ON softkey saves the new destination and enables forwarding; SAVE/OFF softkey saves the new destination and disables forwarding.

- Pressing the SUPERKEY moves the user back to the idle set display. If either internal or external forwarding is enabled (SAVE/ON), <FWD ON> appears in the top right corner if the SAVE/ON softkey was pressed for either internal or external forwarding.

Call Forwarding - Display Prime as Forwarder

Description

This feature displays either the forwarder's prime line or the logical line on the forwardee's set display. If COS Option 258 (Display Prime as Forwarder) is enabled, the prime line of the set that forwarded the call is displayed. If this COS is disabled, the logical line appears as the forwarder for all types of forwarding.

Conditions

The following conditions apply to this feature:

- COS Option 258 is only checked if the forwarder is a logical line.
- Calls to logical lines are only forwarded if the logical line appears only on one set.
- Calls forwarded from a single appearance direct trunk select key that has COS option 815 enabled will always display the prime as forwarder.
- This feature is only applicable when the forwardee's set is a display set.

Programming

Enable COS Option 258 (Display Prime as Forwarder) in the COS of the extension where the logical line appears.

Operation

None.

Call Forwarding - Forward Call

Description

This feature allows a *SUPERSET 410*, *SUPERSET 420*, or *SUPERSET 430* telephone user to force an incoming call to be forwarded to a pre-programmed forward destination.

Users can forward both ringing calls and camped-on calls. *SUPERSET 430* telephone users can view the calling party identity on the LCD display, and decide if it is to be forwarded or not, rather than having the system forward it automatically.

Conditions

The following conditions apply to this feature:

- This feature is available to *SUPERSET 410, SUPERSET 420,* and *SUPERSET 430* telephones.
- This feature will apply to telephones which have forwarding programmed, but not enabled, or have *Call Forward - No Answer* programmed.

- The conditions which apply to *Call Forward - No Answer* also apply to this feature.
- The feature applies only to prime line (or intercom line) calls, and only to calls which can normally be forwarded (i.e., not callbacks or wake-ups, etc.).

Programming

The following programming is required:

- For *SUPERSET 410, SUPERSET 420,* and *SUPERSET 430* telephones, program call forwarding as described in *Call Forwarding.*
- To activate this feature from a *SUPERSET 410* telephone, you must program a FORWARD CALL feature key. (See *Feature Keys.*)

Operation

To forward a ringing call:

***SUPERSET 410* Telephones:**

- Press the FORWARD CALL feature key.

***SUPERSET 420* Telephones:**

- Press the FORWARD softkey.

***SUPERSET 430* Telephones:**

- Press the FORWARD CALL softkey.

Call Park

Description

This feature allows users of industry-standard telephones to put a call on hold (parked) and then replace the handset. The call may be retrieved at the extension at which the call was parked, or from any other extension in the system. If Music-on-Hold is available, the parked party hears music. The parking extension may not originate or receive new calls until the parked call is retrieved. It can only access paging equipment.

Conditions

The following conditions apply to this feature:

- This feature is available only at industry-standard telephones.
- If there is no system music then the caller on hold hears silence.
- COS Options 401 (Call Park) and 403 (Trunk Recall Partial Inhibit) are mutually exclusive.
- The paging and call park feature access codes are the only access codes that the parking industry-standard telephone can dial after the park is done.
- The industry-standard telephone is considered busy by the Campon and Callback features until the parked party is retrieved.
- The parked call can only be picked up from another extension once the parking extension has gone on–hook.

- When the call is parked, the timer for COS Option 254, Call Hold Recall Timer is started.
- When the Call Hold Recall timer times out, the industry-standard telephone audibly starts to ring. Forwarding on the extension is ignored and the timer for COS Option 115 (Attendant Timed Recall - No Answer) is started. Music or silence continues for the parked party. When the Recall No Answer timer expires, recall handling is done; see *Recall.*

Programming

Assign an access code to Feature 33 (Call Park).

Enable COS Option 254 (Call Hold Recall Timer) for the extension, to set the Call Park recall time.

Enable COS Option 401 (Call Park) for the extension that parks the call and for extensions using the Call Park access code to pick up the parked call.

Enable COS Option 218 (Directed Call Pickup) for others to pick up the parked call.

Operation

To park a call:

- Flash the switchhook (standard telephone) - wait for dial tone.
- Dial the Call Park access code - wait for dial tone.
- Replace the handset, or access paging equipment; see *PA Paging.*

To retrieve a parked call from the original extension:

- Lift the handset - connection is made.

To retrieve a parked call from another extension:

- Dial the Call Park or the Directed Call Pickup access code.
- Dial the number of the extension where the call was parked.
- The call is connected to the remote extension.

Callback

Description

The Callback feature allows a user to be notified when a busy device becomes free or when a set has been used after a no answer condition was encountered.

Callback busy allows a user who has encountered a busy set, hunt group or trunk group to have the call completed when the required set, hunt group or trunk group becomes idle. The system continuously monitors the originating set or console and the required device. When the originating set or console is idle and the call can now be completed, the system calls the originating set or console; when that set or console answers, it calls the required device.

Callback No Answer allows a user, after dialing an extension which does not answer, to have the call completed later after the called party uses the telephone. The system continuously monitors the originating set or console and the called set. When the called set goes off-hook and then returns to idle, the callback is handled in the same way as Callback Busy.

Up to 100 Callback requests may be active within the system at any time; however, a maximum of only 25 ARS callbacks is permitted in these requests.

See *Attendant Callback Busy/No Answer and Callback Busy/No Answer.*

Conditions

The following conditions apply to this feature:

- Use the *Callback* feature if you know that the person you are trying to contact is somewhere in the office. Use the *Message - Call Me Back* feature if you know that the person you are trying to contact is out of the office.
- If more than one callback request is active on any device, the requests are queued and serviced in order of placement.
- Duplicate callback requests supercede the original request.
- If the two parties involved in a callback hold a telephone conversation (not a conference) before the callback is honored, the callback is canceled automatically.
- Callbacks to a busy ARS digit string are NOT canceled if the same ARS digit string to which the callback was set is successfully dialed.
- A callback is canceled as soon as the telephone that originated it is rung, even if a conversation is not established.
- Internal callbacks outstanding for more than eight hours are canceled automatically; callbacks to a busy ARS digit string are canceled after 1 hour.
- If a callback is not answered by the originating set or console within 20 seconds, it is automatically canceled.
- If the called party becomes busy before the originating party answers the callback, the originating party hears busy tone when the callback is answered. The callback is purged.
- When a callback is placed to a hunt group, it is placed to the whole hunt group and not to any particular member.
- The scan for an idle hunt group member for the callback does not alter the next extension to hunt for in a circular hunt group.
- Callback - Busy must be activated within 10 seconds of receiving busy tone.
- Campons to extensions are honored before callbacks.
- Originate Only extensions do not have access to this feature.
- Call forwarding is ignored when a callback rings at the originating set.
- Callbacks ringing at a set cannot be picked up using CALL PICKUP.
- If a *SUPERSET* telephone originates a callback, the callback always rings the set's Prime Line. If the prime line has Key line appearances, the system places the key appearances into a simulated busy state to

prevent them from ringing during the callback. If the prime line has Multiline appearances, they remain unchanged.

- A callback only rings a *SUPERSET* or industry-standard telephone if the set and the prime line are available.

Programming None.

Operation None.

Callback - Busy

Description The Callback - Busy feature allows a user who has encountered a busy telephone, hunt group, or trunk group to have the call completed when the required telephone, hunt group, or trunk group becomes idle.

See *Callbacks* for details on callbacks.

See *Expensive Route Warning* for callbacks to less expensive ARS routes.

Conditions Originate Only extensions do not have access to this feature.

Programming Enable COS Option 300 (Automatic Callback) in the set's COS.

Assign an access code to Feature 20 (Callback - Busy).

For feature key activation of callback on *SUPERSET 410* telephones, program a CALLBACK feature key. (See Feature Keys.)

If Callback - Busy is to be permitted on outgoing trunks, also enable COS Option 236 (Outgoing Trunk Callback) in the set's COS.

Operation Operation varies depending on the type of telephone as described below.

Industry-standard and *SUPERSET 401+* Telephones:

The called extension, hunt group, or trunk group is busy. To set a callback:

- Dial the Callback - Busy access code within 10 seconds. Dial tone is returned. Your set is now ready for normal use.
- When the busy extension, hunt group, or trunk group becomes idle, your set rings.
- Lift the handset - ringback or audio for ARS dialing is heard.
- The required extension rings or the trunk is seized.

***SUPERSET 410* Telephones:**

The called extension, hunt group, or trunk group is busy. To set a callback:

- Press the CALLBACK feature key - dial tone is returned. Your set is available for normal use.
- When the busy extension, hunt group, or trunk group becomes idle, your set rings.

- Go off-hook - ringback or audio for ARS dialing is heard.
- The required extension rings or the trunk is seized.

Note: You can also follow the steps given for industry-standard telephones to set a Callback - Busy on *SUPERSET 410* telephones.

SUPERSET 420 **Telephones:**

The called extension, hunt group or trunk group is busy. To set a callback:

- Press the CALLBACK softkey - if you're off-hook, dial tone is returned; if you're using handsfree operation, the set goes idle. CALLBACK SET... appears briefly in the display. Your set is available for normal use.
- When the busy extension, hunt group, or trunk group becomes idle, your set rings. Your set's LCD indicates that a callback is calling.
- Go off-hook - ringback or audio for ARS dialing is heard.
- The required extension rings or the trunk is seized.

SUPERSET 430 **Telephones:**

The called extension or trunk group is busy. To set a callback:

- Press the CALL ME BACK softkey - if you're off-hook, dial tone is returned; if you're using handsfree operation, the set goes idle. CALLBACK SET... appears briefly in the display. Your set is available for normal use.
- When the busy extension, hunt group, or trunk group becomes idle, your set rings. Your set's LCD indicates that a callback is calling.
- Go off-hook - ringback or audio for ARS dialing is heard.
- The required extension rings or the trunk is seized.

Callback - No Answer

Description

Callback - No Answer allows a user, after dialing an extension that does not answer, to have the system complete the call after the called party returns and uses the telephone. The system continuously monitors the originating telephone and the called telephone. When the called telephone goes off–hook, the callback is handled in the same way as Callback - Busy.

See *Callbacks* for details on callbacks.

Conditions

The following conditions apply to this feature:

- Callback - No Answer cannot be activated by access code while listening to ringback; see *Operation.*
- Callback - No Answer can be activated on extension numbers only.
- Originate Only extensions do not have access to this feature.

Programming

Enable COS Option 300 (Automatic Callback) in the set's COS.

Assign an access code to Feature 43 (Callback - No Answer).

For feature key activation of callback on a *SUPERSET 410* telephone, program a CALLBACK feature key. (See Feature Keys.)

Operation

Operation varies depending upon the type of telephone as described below.

Industry-standard and *SUPERSET 401+* Telephones:

The called extension does not answer. To set a callback:

- Hang up.
- Go off–hook again.
- Dial the Callback - No Answer access code.
- Dial the unanswered set's extension number. Dial tone is returned. Your set is available for normal use.
- The next time the called (unanswered) set goes on–hook, your set rings.
- Lift the handset to call the extension.

***SUPERSET 410* Telephones:**

The called extension doesn't answer. To set a callback, follow the steps given for industry-standard telephones, or:

- Press the CALLBACK feature key - dial tone is returned. Your set is available for normal use.
- When the called (unanswered) set goes on–hook, your set rings.
- Go off–hook to call the extension.

Note: You can also follow the steps given for industry-standard telephones to set a Callback - Busy on *SUPERSET 410* telephones.

***SUPERSET 420* Telephones**

The called extension doesn't answer. To set a callback:

- Press the CALLBACK softkey - if you're off-hook, dial tone is returned; if you're using handsfree operation, the set goes idle. CALLBACK SET... appears briefly in the display. Your set is available for normal use.
- When the called (unanswered) set goes on–hook, your set rings. Your set's LCD indicates that a callback is calling.
- Go off-hook to call the extension.

***SUPERSET 430* Telephones:**

The called extension doesn't answer. To set a callback:

- Press the CALL ME BACK softkey - if you're off-hook, dial tone is returned; if you're using handsfree operation, the set goes idle. CALLBACK SET... appears briefly in the display. Your set is available for normal use.
- When the called (unanswered) set goes on–hook, your set rings. Your set's LCD indicates that a callback is calling.
- Go off–hook to call the extension.

Call Rerouting

Description

This feature provides flexibility for the routing of incoming calls, attendant access, call interception, and routing for various features. Different types of calls can be routed to different answering points in DAY, NIGHT 1, and NIGHT 2 service for each tenant. You program the call rerouting answering points in CDE Form 19 (Call Rerouting Table). Refer to *Form 19 - Call Rerouting Table* in the *Customer Data Entry Practice*, for a list of the call rerouting options.

Rerouted calls are processed differently than normal calls. The system considers rerouted calls to be important calls that must get through. Some features on the reroute point are ignored because of this.

Also see *Did/Dial-in/Tie Intercepts, Night Services, Trunk Support CO (LS/GS), and Uniform Call Distribution.*

Conditions

The following conditions apply to this feature:

- Do Not Disturb is ignored on reroute points.
- Rerouted callers are automatically camped on to the reroute point regardless of COS options. Campon is not done for consoles.
- Valid access codes for reroute types, with the exceptions listed below, are: LDNs, consoles, industry-standard telephones, *SUPERSET* telephones, logical lines, hunt groups (all types except data hunt groups), ACD paths and night bells.
- Console access codes are not valid rerouting points for the following entries in Form 19:

  DID always routing to this tenant.

  DID forward on busy/no answer.

  Dial-in Tie always routing to this tenant.

  Dial-in Tie forward on busy/no answer.
- LDN Access Codes are not valid rerouting points for Non-dial-in trunks alternate recall points.
- Only Recording Hunt Groups are valid for recording routing.
- CO trunks use rerouting when they originate, following the routing points programmed for each individual trunk; see *Trunk Support CO (LS/GS).*
- Forwarding is examined on a reroute point (regardless of the number of forwardings already done). See *Call Forwarding.*
- Whenever a DID trunk reroutes to the Night 1 point of its Attendant Access Night Point, it does not recall due to a no answer on that point. The same applies for Tie trunks and the Tie Attendant Access Night Point and for DISA trunks, and for internal extensions and the Dial 0 point.
- Device Interconnection checking is done between the rerouted party and the reroute point. For LDN keys as reroute points, the checking is done between the calling party and the console with the lowest

Bay/Slot/Circuit PLID where the LDN key appears. If the connection is not allowed then the reroute point is ignored.

Programming

All Call Rerouting entries are programmed in CDE Form 19 (Call Rerouting Table). Enter a valid extension number or LDN access code in one of the DAY, N1 or N2 service columns as desired. Whether the tenant of the calling or called party is used depends upon the feature selected.

Also see *Night Services.*

Operation

See *Attendant Night/Day Switching and SUPERSET Night/Day Switching* for changing the night/day service.

Campon

Description

A device is able to indicate to a busy party that communication is desired, and to be connected when the party is free. Also, the user can make a continuing request for a trunk when the trunk group is busy, and be connected to a trunk when one becomes free.

When calling an extension, hunt group, or ARS, if the destination is busy, the caller usually receives a tone for a period and then camps on to the busy device. The tone given indicates whether or not campon is allowed during the period, or is done at the end of the period. For some calls, the period is skipped.

The busy called extension receives a tone alerting the party that there is a call waiting; see *Campon Warning Tone.*

An extension can consult the first waiting party (in hunt groups as well) using the SWAP CAMPON feature.

Campon may be initiated on a trunk group that has been programmed to give the expensive route warning; see *Expensive Route Warning.*

For recall from Campon, see *Recall.*

Conditions

The following conditions apply to this feature:

- On *SUPERSET 430* telephones, the camped on party identification display takes precedence over the held party display.
- Calls camped on to a device are serviced in two groups, trunks and internal callers, with trunks being served first. Within each group calls are serviced on a first-come, first-served basis.
- When a party camps on to a busy trunk group, the DTMF receiver used to reach the trunk group is released. The campon to the trunk group is honored when a trunk group is free and a DTMF receiver is free.
- The attendant or an extension can transfer a call to a busy destination. The transferred party camps on immediately.

- The transferred party can camp on as long as the transferring party can camp on. COS checks are done on the transferring party and the transfer is disallowed if the transferring party cannot camp on.
- The transferred party hears system music while camped on; if there is no system music, silence is heard.
- All devices except the console can camp on; consoles and extensions can transfer a call into campon.
- If campon is allowed for a call, special busy tone is supplied for 10 seconds, after which the device is camped on automatically.
- If campon is not allowed for the call then busy tone is heard for 30 seconds and then the call is disconnected.
- DID and CO trunk calls that campon to busy devices receive ringback tone.
- Tie and DISA trunk calls that campon to busy devices receive busy tone (the same audio as internal callers).
- If there is no busy intercept then a DID trunk immediately camps on to a busy device if the trunk is allowed to campon to the device.
- On busy tone timeout, a serial call is dropped, and does not recall.
- On *SUPERSET 430* telephones a CAMPON softkey is provided. On *SUPERSET 420* telephones a Wait softkey is provided. On *SUPERSET 410* telephones, a CAMPON feature key can be programmed to activate campon. (See *Feature Keys.*)
- If a *SUPERSET* telephone is using the display or SuperKey feature while camped on to ARS, the campon is not honored when ARS becomes available; it is only honored after the telephone exits from the feature and ARS is available.
- Industry-standard telephones and *SUPERSET* telephones that camp on to a device without using a softkey hear busy tone while camped on.
- All trunk types and station and *SUPERSET* telephones can camp on.
- Calls can camp on to busy industry-standard telephones, *SUPERSET* telephones, logical lines, hunt groups, and trunk groups.
- Campon is done immediately for reroute points that are busy regardless of Campon COS options; see *Call Rerouting.*
- Campon is done for Automated Attendant hunt groups regardless of campon COS options; see *Automated Attendant.*
- Campon tones are not passed to lines which have COS Option 216 (Data Security) enabled in their COS.
- The campon warning tone applies to the party to which the call is being transferred.

Programming

Enable COS Option 301 (Campon) for the device to allow it to camp on.

If Campon is to be permitted on outgoing trunks, enable COS Option 237 (Outgoing Trunk Campon) for the device in addition to COS Option 301 (Campon).

Set the attendant campon recall timer through COS Option 117 (Attendant Timed Recall - Campon). See *Attendant Timed Recall.*

Operation

Operation varies depending upon the device as described below:

Industry-standard and *SUPERSET 401+* Telephones:

To camp on to a busy device:

- Dial the number - special busy tone is returned.
- After 10 seconds of special busy tone, campon is done and busy tone is returned. The called extension receives campon warning tone.
- When the called extension goes on-hook, the calling extension hears ringing tone and the called extension is rung.

  OR, if the camp on was performed on a trunk group.
- When the called trunk group becomes idle, the calling extension hears ringing tone the system dials the originally dialed digits.

***SUPERSET 410* Telephones:**

If the set does not have a CAMPON feature key programmed, use the procedure for industry-standard telephones, above. If the CAMPON feature key is programmed, use the following procedure.

To camp on:

- Dial the number - busy tone is returned.
- Press the CAMPON feature key. Remain off-hook.

***SUPERSET 420* Telephones:**

- While receiving special busy tone, press the WAIT softkey. Remain off-hook. WAITING XXXX appears in your display, where XXXX is the number of the busy extension.

***SUPERSET 430* Telephones:**

- While receiving special busy tone, press the I WILL WAIT softkey. Remain off-hook.

To Transfer into Busy:

- Put a party on consultation hold.
- Dial the extension to transfer the call to. Special busy tone is heard.
- Hang up.

Console

See *Attendant Transfer To Campon.*

Campon Warning Tone

Description

When a device camps on to an extension or hunt group, a warning tone is sent to the extension user over the current call. The warning tone can be programmed to repeat every 5 to 15 seconds.

Conditions

The following conditions apply to this feature:

- A different tone is given for internal and external calls. When an industry-standard telephone or *SUPERSET* telephone calls, a single burst of 440 Hz tone is given for 200 ms. For trunks and consoles calling, a double burst of the same tone is given.
- The tone is not heard by other parties involved in the call with the busy extension, but a short silent period may be noticed.
- COS Options 242 (Repeated Camp-On Beeps) and 216 (Data Security) are mutually exclusive.
- COS Option 216 (Data Security) prevents the tone from being applied to particular extensions. This has no effect on the SWAP CAMPON feature.
- Repeated campon beeps applies only when trunks are camped on.
- Trunks camped onto extensions with Repeated Campon Beeps do not recall.
- For hunt groups, the tone is given to the first busy member of the hunt group that is not Do Not Disturb and that is logged in (UCD).
- The COS of this member is used to check if repeated campon beeps on a hunt group applies to the first party in the hunt group. The tone is applied irrespective of the line that is in use.
- If the extension selected from the hunt group for the tone has Data Security enabled then no tone is given. No other extension is selected.
- The tone is not given to members of recording hunt groups.
- The feature applies to *SUPERSET* telephones only if their prime line is a key line (in which case all parties on the line get the tone) or their prime line has no appearances and they are on their prime line appearance.
- The feature applies to logical lines only if the line is a key line or a single appearance multicall line. The first *SUPERSET* telephone where the line appears gets the tone. All parties on the key line get the tone.
- The warning tone is applied only if the party to get the tone is talking to another party or is held by another party.
- Music is removed while the campon tone is applied (digital Bays only).
- Extensions in analog bays do not receive campon tone if they are connected to Music-on-Hold.

Programming

Enable COS Option 242 (Repeated Camp-On Beeps) for the extension if trunk calls should continue to notify the extension. For hunt groups, this option should be enabled for the first extension in the hunt group.

Enable COS Option 216 (Data Security) for the extensions that should not receive the warning tone. For hunt groups, this option should be enabled for each extension in the group.

Set the cycle time for the repeated beeps via COS Option 255, Repeated Camp-On Beeps Timer. The default setting is 10 seconds.

Operation

A call camps on and a tone is heard. For *SUPERSET* telephones, the SWAP CAMPON feature can be used to consult with the party that camped on. Alternatively, for any extension, finish the current call; hang up, and the camped on party rings the extension.

CENTREX™ Compatibility (Double Flash Over Trunk)

Description

This feature provides the ability to send a double switchhook flash out over a trunk. Flashing over a trunk enables a telephone on the PBX to use CENTREX features. Callers must follow the instructions specified by the local central office concerning when a double flash should be used.

A CENTREX caller can put a CENTREX extension on softhold and dial another extension (by sending out a Flash Over Trunk followed by the digits of the extension). If the caller hears busy or ringing, the caller can put the CENTREX extension on softhold again, and dial the Double Flash Over Trunk feature access code. This sends out two flashes on the trunk, with one second between the flashes. When the CENTREX service receives the double flash, it clears down the ringing/busy tone. The caller returns to a talking state with the trunk that was on softhold.

Conditions

The following conditions apply to this feature:

- Only one SMDR record will be generated by a call using this feature.
- The trunk must be a 6 circuit CO (LS/GS) trunk or a T1 LS/GS trunk.
- The trunk must be in a trunk group in order to flash over the trunk.
- This feature may be used by telephones only.
- This feature cannot be used with a PABX conference on consultation hold. It only works if there is one trunk is on consultation hold.
- The feature will not function if there is a conference on consultation hold. It will only function if there is one trunk on consultation hold.
- Extensions cannot send a flash signal over analog trunks.

Programming

Perform the following:

- To allow a CENTREX caller to flash over the trunk, enable COS Option 257 (Flash Over Trunk) in the COS of the telephone(s) permitted to flash over trunks.
- Assign a feature access code to Feature 53 (Double Flash Over Trunk) in Form 02 (Feature Access Codes).

- In the Select Options Subform of CDE Form 13 (Trunk Circuit Descriptors), set the Flash Over Trunk option to YES, set the Flash Type option to either LOOP FLSH or RING GND, and set the Flash Timer to an appropriate value (if applicable). **Note:** The trunk must be in a trunk group.

Operation

Generally:

- While on a busy or ringing CENTREX trunk, press the TRANS/CONF key, press the FLASH key (*SUPERSET 401+* telephones), or flash the switchhook (industry-standard telephone).
- Dial the Double Flash Over Trunk access code.
- Or press the DOUBLE FLASH key on *SUPERSET 410, SUPERSET 420*, and *SUPERSET 430* telephones.

Example CENTREX Transfer (Access Code Method):

- Answer call from a CENTREX line.
- Press the TRANS/CONF key, press the FLASH key (*SUPERSET 401+* telephones), or flash the switchhook (industry-standard telephone). The CENTREX trunk is placed on softhold.
- Dial the Flash Over Trunk feature access code.
- Dial the CENTREX number of a second CENTREX extension.
- When the second CENTREX extension does not answer or is busy, press the TRANS/CONF key, press the FLASH key (*SUPERSET 401+* telephones), or flash the switchhook (industry-standard telephone). Until the ARS unknown digit length timeout expires, the flashook is ignored and the TRANS/CONF key or FLASH key is not available.
- Dial the Double Flash Over Trunk feature access code. (The CO cancels the second call, clears the ringing/busy tone and reconnects the caller to the softheld CENTREX party. The PABX then returns the caller to the original softheld CENTREX trunk.)

Example CENTREX Transfer (Feature Key Method):

- Answer call from a CENTREX line.
- Press the SINGLE FLASH key on *SUPERSET 410*, *SUPERSET 420*, and *SUPERSET 430* telephones.
- Dial the CENTREX number of a second CENTREX extension.
- When the second CENTREX extension does not answer or is busy, FLASH key on *SUPERSET 410*, *SUPERSET 420*, and *SUPERSET 430* telephones. (The CO cancels the second call, clears the ringing/busy tone and reconnects the caller to the softheld CENTREX party. The PABX then returns the caller to the original softheld CENTREX trunk.)

CENTREX Compatibility (Single Flash Over Trunk)

Description

This feature provides the ability to send a switchhook flash out over a trunk. Flashing over a trunk allows for the use of CENTREX features by telephones within the PBX. Callers must follow instructions specified by the local central office concerning which access codes to dial, and when to wait for dial tone. After sending the flash over the trunk, the system will wait for dial tone from the central office, or for the Limited Wait For Dial Tone timer.

Speed Call

The access code for Flash Over Trunk, followed by digits which activate CENTREX features may be programmed into a speedcall key, or a system abbreviated dial number. Any central office access code with an asterisk (*) in it must be entered as **. See *Abbreviated Dial* in this document for information on special codes.

Conditions

The following conditions apply to this feature:

- Only one SMDR record will be generated by a call using this feature; also, digits dialed after the flash are not appended to the SMDR record.
- The trunk must be a 6 circuit CO (LS/GS) trunk or a T1 LS/GS trunk.
- The trunk must be in a trunk group in order to flash over the trunk.
- Ensure that the DTMF option in CDE Form 13 - Trunk Circuit Descriptors (Select Options Subform) is set up to the option which is understood by the local central office. If this is not done, the local central office will not receive the digits dialed after the flash.
- If the user makes an error while dialing digits to the local central office, the user must wait for it to timeout and return to the previous state (there is no CANCEL function, and no back arrow key).
- This feature may be used by telephones only.
- The feature will not function if there is a conference on consultation hold. It will only function if there is one trunk on consultation hold.
- Extensions cannot send a flash signal over analog trunks.

Programming

Perform the following:

- To allow a CENTREX caller to flash over the trunk and go directly into dial state, enable COS Option 257 (Flash Over Trunk) in the COS of the telephone(s) permitted to flash over trunks.
- To allow a CENTREX caller to go directly into talk state after a Flash Over Trunk is performed, enable COS Option 816 (CENTREX Flash Over Trunk) in the COS of the trunk permitted to flash. **Note:** The trunk must be in a trunk group.
- Assign a feature access code to Feature 46 (Single Flash Over Trunk) in Form 02 (Feature Access Codes).
- In the Select Options Subform of CDE Form 13 (Trunk Circuit Descriptors), set the Flash Over Trunk option to YES, set the Flash Type option to either LOOP FLSH or RING GND, and set the Flash

Timer to an appropriate value (if applicable). **Note:** The trunk must be in a trunk group.

Operation

Generally:

- While talking on a trunk, press the TRANS/CONF key, the **Flash** key (*SUPERSET 401+* telephones), or flash the switchhook (industry-standard telephone).
- Dial the Single Flash Over Trunk access code.
- Or press the SINGLE FLASH key on *SUPERSET 410, SUPERSET 420,* and *SUPERSET 430* telephones.
- Dial the applicable CENTREX codes, and follow the CENTREX instructions.

Example CENTREX Transfer:

- Answer call from CENTREX line.
- Press the TRANS/CONF key, the FLASH key (*SUPERSET 401+* telephones), or flash the switchhook (industry-standard telephone).
- Dial the Single Flash Over Trunk feature access code.
- Or press the SINGLE FLASH key on *SUPERSET 410, SUPERSET 420,* and *SUPERSET 430* telephones.
- Dial the CENTREX number of the second external party.
- Announce the caller.
- Hang up. The caller is transferred within the CENTREX network and trunk is released.

Class of Restriction (COR)

Description

Fifty Class Of Restriction (COR) groups are available in the system to provide 50 different levels of outgoing call capabilities. Each extension, *SUPERSET* telephone, dataset, console or dial-in trunk is assigned a COR which defines the outgoing call capabilities for that device. All devices with the same COR have access to the same outgoing call capabilities. By using CORs, the amount of ARS programming is reduced.

The Class of Restriction allows the system to restrict which trunk can not be accessed by a user. For example, COR 01 could restrict users from accessing CO trunks (local and DDD), WATS, and TIE lines. COR 02 could restrict users from accessing WATS, tie lines, and DDD (but allow local calls). COR 03 could restrict users from accessing DDD and WATS (but allow local calls and tie line calls).

Note: All extensions belong to COR group 1 in the default database.

Conditions The following conditions apply to this feature:

- CO trunks do not have a COR. When they access ARS, it is either using the COR of the transferring or forwarding party, or no COR at all, giving universal access.
- The COR may change temporarily for a caller through Verified Account Codes.
- The maximum dialed digits feature is based on the COR of the caller. Refer to the *Automatic Route Selection and Toll Control Practice*.

Programming Assign a COR number to each extension, *SUPERSET* telephone, dataset, Dial-In Trunk and console via CDE Form 09 (Station /*SUPERSET Telephones)*, 07 (Console Assignments), 15 (Dial-In Trunks) and 12 (Data Assignment).

Refer to the *Automatic Route Selection and Toll Control Practice*, for additional information on COR programming.

Operation None.

Class of Service (COS)

Description Each extension, trunk, *SUPERSET* telephone, dataset, ACD position, or console is assigned a Class Of Service (COS) which defines the features available for that device. All devices with the same COS (which defines the COS Options) have access to the same features. Fifty Classes Of Service are available in the system to provide 50 different levels of feature accessibility. Each COS can have a name associated with it.

In the default database, all devices belong to COS 1.

Conditions The following conditions apply to this feature:

- Several COS options are mutually exclusive; these are identified with the description of each feature.
- The COS may change temporarily for a caller through Verified Account Codes (Traveling Class Marks).
- The COS used is sometimes that of the caller and sometimes that of the called party. Check the feature to determine whose COS is used.
- LDNs do not have a COS. The COS for an LDN is determined from the lowest Bay/Slot/Circuit PLID of the console where the LDN is programmed.
- Logical lines do not have a COS. The COS is determined from the first *SUPERSET* telephone on which it is programmed. See *Logical Lines.*

Programming Assign the desired features to each COS via CDE Form 03 (COS Define).

Assign a COS to each console, extension, Dial-In Trunk, Non Dial-in Trunk, dataset, and ACD position via CDE Forms 07, 09, 12, 14, 15, 39 and 40.

Operation None.

Clear All Features

Description An extension user may cancel all Call Forwarding, Do Not Disturb, and Callbacks active at that extension.

Conditions None.

Programming Enable COS Option 221 (Clear All Features) for the extension.

Assign an Access Code to Feature 25 (Clear All Features).

Operation To cancel all Call Forwarding, Callbacks, and Do Not Disturb:

- Lift the handset - dial tone is returned.
- Dial the Clear All Features feature access code.
- Dial tone is returned.
- Replace the handset.

Conference

Description This feature allows a set user to establish a conference of up to five parties (including the originating extension), without the assistance of the Attendant.

See *Device Interconnection Control* for information on controlling trunk conferencing, and *Attendant Conference*.

Conditions The following conditions apply to this feature:

- A maximum of six conferences may take place at one time; the maximum number of conferees permitted at one time is 18; the maximum number of conferees permitted in one conference is 5.
- An attendant console cannot be a member of a set-initiated conference.
- Only one party may flash out of the conference at a time.
- An extension with COS Option 203 (Broker's Call) or 252 (Broker's Call With Transfer) enabled cannot set up a conference. The switchhook flash is interpreted as a SWAP and not as a conference attempt.
- A non-busy extension forms conferences automatically.
- The Override feature forms a conference call.
- If an industry-standard telephone attempts to conference with a trunk that gives answer supervision and the trunk has not given answer supervision yet then the trunk is dropped.
- Two conference calls cannot be conferenced together.
- All *SUPERSET* telephones in a conference can put the conference on hold.

- Another method of creating conferences is through key line privacy; see *Privacy Enable/Privacy Release.*
- See *Flash Control* for controlling flashing on an extension.

Programming Enable COS Option 302 (Flash-in Conference) for the extension. This allows an extension to create conferences of greater than three parties.

Operation Operation depends upon the type of device as described below.

Industry-standard Telephones:

To establish a conference:

- Establish a 2-party call.
- Flash the switchhook.
- Transfer dial tone is returned (if programmed).
- Dial the number of the next conferee.
- When the conferee answers, flash the switchhook - a conference is established.

Any extension in the conference may add additional conferees to the conference by repeating the above steps.

***SUPERSET 401+* Telephones:**

To establish a conference:

- While on a call, press the FLASH key.
- Transfer dial tone is returned (if programmed).
- Dial the number of the desired extension or the telephone number of the desired external party.
- When the called party answers, press the FLASH key to form the conference.

Any extension in the conference may add additional conferees to the conference by repeating the above steps.

***SUPERSET 410* and *SUPERSET 420* Telephones:**

To establish a conference:

- While on a call, press the TRANS/CONF key.
- Transfer dial tone is returned (if programmed).
- Dial the number of the desired extension or the telephone number of the desired external party.
- When the called party answers, press the TRANS/CONF key to form the conference. If you receive busy tone, or if the called party doesn't answer, press the CANCEL key to be connected with the original party.

Any extension in the conference may add additional conferees to the conference by repeating the above steps.

***SUPERSET 430* Telephones:**

To establish a conference:

- While on a call, press the TRANS/CONF softkey.
- Transfer dial tone is returned (if programmed).
- Dial the number of the desired extension or the telephone number of the desired external party.
- When the called party answers, press the TRANS/CONF softkey to form the conference. If you receive busy tone, or if the called party doesn't answer, press the BACK TO HELD softkey. You'll be connected with the original party.

Any extension in the conference may add additional conferees to the conference by repeating the above steps.

Conflict Dialing

Description

The system can differentiate between conflicting extension numbers such as "52345" and "5234". This implies that extensions can be programmed as 1-, 2-, 3-, 4-, or 5-digit numbers with the first digits being identical. The system selects the shorter extension number if the next digit is not dialed within a preselected time.

A conflict exists between two extension numbers if the first number is contained in the second number, starting with the first digit. For example, 1234 conflicts with 12345 but 1234 does not conflict with 123 (123 conflicts with 1234).

Conditions

The following conditions apply to this feature:

- First digit conflicts between the access codes assigned to Executive Busy Override and the Callback - Busy features, and other numbers within the numbering plan, are permitted.
- Feature access codes are not permitted to conflict with any other access codes in the system.
- Extension numbers may not conflict with feature access codes.
- ARS leading digit strings are not permitted to conflict with any other access codes in the system. Extension numbers may conflict with ARS leading digits.
- Modem Pool Hunt group access codes cannot conflict with any other access code in the system and no access code may conflict with a Modem Pool Hunt group access code.
- Conflict dialing applies to features that need to have access codes entered (call forwarding for example). This includes the programming features on the *SUPERSET* telephones.
- Normal system inter-digit timeout is 15 seconds (not ARS dialing). If a conflict exists, the Dialing Conflict Timer applies.

Programming Select an appropriate time-out period for System Option 50 (Dialing Conflict Timer 2-10 s).

Operation None.

Consoleless Operation

Description The system may be operated without the use of an attendant console. Under these conditions all features associated with the console are not available. *SUPERSET* telephones may be used as subattendant positions. These may switch the system night service (see *Night/Day Switching)* and have enhanced call handling and recall capabilities (see *Subattendant*).

Conditions Some attendant features are not available at subattendant positions.

Programming No attendant consoles are programmed.

Operation None.

Contact Monitor

Description This feature allows a station line circuit to be used for monitoring an alarm contact. The contact to be monitored is connected across Tip and Ring of the circuit. When the contact closes, a call is originated by the station line circuit and the call is directed to its tenant's Dial 0 or Priority Dial 0 Routing Point. The PABX handles the call as a call reroute; see *Call Rerouting, Attendant Access (Dial 0), and Priority Dial 0.*

Conditions The following conditions apply to this feature:

- Contact Monitor is available on OPS and ONS Line cards.
- COS Options 400 (Contact Monitor) and 241 (Receive Only) are mutually exclusive.
- The contact signal is non–latching; if the contact opens, the Dial 0 call is terminated.
- If the station line circuit has COS 239 (Priority Dial 0) enabled then the Priority Dial 0 Routing point is used.
- If there is no routing point then the origination is still processed (the station line circuit is now busy) and the station set is given reorder tone.
- If the call is answered at a console, the console display indicates that the caller is a contact monitor.

Programming Enable COS Option 400 (Contact Monitor) for the extension.

Operation Close the contact and a call is made to the Dial 0 Routing point or to the Priority Dial 0 Routing point.

Customer Data Entry

Description Customer Data Entry (CDE) is a full screen application, using softkey prompts and simple graphics, for entering and changing customer programming.

Enter customer data from a terminal via the RS-232 connector on the rear of the cabinet, or from an Attendant Console (a screen of reduced height is used).

Customer Data Entry can also be performed from a remote location, using a terminal connected to the PABX via modems (with a Null Modem Connector).

Conditions None.

Programming Refer to the *Customer Data Entry Practice*, for detailed programming information.

Operation None.

Customer Data Entry - Default Data

Description The system is preprogrammed with default Class-Of-Service (COS) Options and Class-Of-Restriction Options; if no alternates are programmed, the system defaults to the preprogrammed data. The Feature Access Codes can be entered during Customer Data Entry.

Conditions None.

Programming Refer to the *Customer Data Entry Practice.*

Operation Default data in the database includes:

- Device Interconnection Table
- T1 Link Descriptor
- Test Line - The location of the Test Line is Bay 1, Slot 1, Circuit 1
- Class of Service Timers
- System Timers
- Trunk Circuit Descriptors
- Data Terminal Equipment (DTE) Profile
- Data Circuit Descriptors

Customer Data Entry - Range Programming

Description This feature allows range programming for blocks of extensions. By entering a range of equipment numbers, one may assign extension numbers, COR, tenant, and COS to a selected block of equipment numbers. The start extension number and defaults for the other values are entered by the programmer. The extensions are assigned sequentially starting at the entered value, and the COS, tenant and COR are assigned to the entire group.

Conditions Refer to the *Customer Data Entry Practice*.

Programming Refer to the *Customer Data Entry Practice*.

Operation None.

Customer Data Print

Description This feature provides a means of displaying the current state of programming of the PABX. Each or all of the CDE forms may be printed, one at a time, in a presentable format.

Conditions The printer must be PABX compatible; refer to *Directed Input/Output*.

Programming See *Printer/Terminal Support* for details on directing the output to the desired printer.

Operation Refer to the *Customer Data Entry Practice* for details.

Data: Associated Data Line (ADL)

Description

The associated data line feature allows a telephone user to originate and disconnect a data call. The association between a telephone (used for dialing or disconnecting the call) and a dataset the call is switched to, once the setup is complete.

The user can establish a basic ADL call from an industry-standard telephone or a *SUPERSET* telephone by dialing an access code and the number of the destination.

The associated data line operates with either asynchronous or synchronous devices.

The data destination of any data call can be predetermined via the HOTLINE feature (see *Data: ADL Hotline).* Industry-standard DTMF and rotary telephones, *SUPERSET* telephones and Digital *SUPERSET* telephones can be used in ADL calls. Either the originator or destination can terminate the data call.

This feature allows the user to establish a basic ADL call by dialing an access code and the number of the destination from an industry-standard telephone or a *SUPERSET* telephone.

Data destinations include:

- data station extension number
- data hunt group access code
- modem hunt group access code followed by ARS digit string.

Included with originate/disconnect capabilities of ADL are the following:

- Data call originate using SPEED CALL key or Abbreviated Dial
- ADL queuing
- Data call disconnect using SPEED CALL key or Abbreviated Dial, or the Disconnect or CALL/ATTN key.

The associated data line can be used to originate or disconnect a data call, operating in either asynchronous or synchronous mode.

Conditions

The following conditions apply to this feature:

- The attendant console cannot use the ADL feature.
- Program the associated voice line (AVL) in CDE Form 12 (Data Assignment).
- Ensure that both the originating and destination datasets are both in the same Operating Mode (asynchronous or synchronous), as programmed in CDE Form 11 (Data Circuit Descriptor).

Programming

Assign Feature Access Code 28 (ADL Call Setup) and Feature Access Code 29 (ADL Disconnect) in CDE Form 02 (Feature Access Codes).

Enable COS Option 900 (Data Station Queuing). When COS 900 is enabled on a voice telephone, queuing to a busy data destination is allowed on an ADL call.

Program ADL Auto Baud and Operating Mode (Synchronous or Asynchronous) in the subform of CDE Form 11 (Data Circuit Descriptor). Operating Mode applies only to DATASET 2100 Series datasets.

To associate a voice telephone with a dataset, enter a valid extension number for the voice telephone in the AVL field of CDE Form 12 (Data Assignment).

Operation

ADL Call Originate

If the Originating dataset does not have ADL Autobaud:

- Dial the ADL access code (if the associated dataset is not idle; reorder tone is heard and the *SUPERSET* telephone displays the message "PLEASE TRY LATER").
- Dial the digits of the destination.
- A high pitched connect tone is heard (if the destination is busy, then busy tone is heard).
- The originating party hangs up the voice telephone and the call is switched to the associated dataset ("HANG UP" or "NEW CALL" prompt can also be used).
- If the *SUPERSET* telephone has a voice party on Hold, the CANCEL key establishes the data call and transfers the voice call to the party on hold.

When Originating dataset has ADL Autobaud:

- Dial the ADL access code (if the associated dataset is not idle; reorder tone is heard and the *SUPERSET* telephone displays the message "PLEASE TRY LATER").
- Dial the digits of the destination (busy tone is heard if the destination is busy).
- When ringback tone is heard, the originator must enter the autobaud character (usually carriage return) so that the dataset can determine the baud rate (this character must be entered within 30 seconds or the call is released; otherwise reorder tone is heard and PLEASE TRY LATER is displayed).
- Connect tone is returned to the originator.
- Hang up the voice telephone.
- Call is switched to the associated dataset.

Note: Once the ADL call has been established, the telephone is free to be used. If a voice call had been established, the other party can be placed on hold by pressing the TRANS/CONF key (FLASH key on the *SUPERSET 401+* telephones), and the ADL call can be established. Pressing the CANCEL or BACK TO HELD softkey reconnects the telephone to the party on hold.

When an ADL Call Origination is unsuccessful, reorder tone is returned and one of the following messages is displayed on the display *SUPERSET* telephone:

- The voice telephone has not been programmed as an AVL to an associated dataset. The set displays "NO ACCESS".
- The associated dataset is not idle. The set displays "PLEASE TRY LATER".
- The user dials the number of a device which cannot handle data. The set displays "INVALID #".
- The baud rates of the source and destination are not compatible and neither have Flow Control programmed. The call is barred. The calling set receives reorder tone and displays "NO ACCESS".
- The associated terminal is not asserting DTR and is NOT programmed for auto-answer. The set displays "NO ACCESS".

ADL Call Disconnect

The source or destination can disconnect an ADL call by:

- dialing the ADL disconnect access code.
- pressing a pre-programmed Attention or Break key on the terminal.
- pressing the DISCONNECT key or the CALL/ATTN key on the dataset.
- by either dataset dropping DTR for the preprogrammed time.

If the call is not set up via the ADL feature, doing an ADL disconnect causes the set to display "INVALID #".

Data: ADL Hotline

Description This feature establishes a data call between two preassigned DTEs when the user dials the ADL access code from the associated telephone.

Conditions Refer to *Associated Data Line.*

Programming Assign the hotline number in CDE Form 12 (Data Assignment).

Operation To originate an ADL hotline call:

- Dial the ADL access code. The call is automatically connected to the preassigned destination. Also, a carriage return <CR> may be required depending upon the programmed ADL Autobaud option. See *Associated Data Line* - "Operation: ADL Call Originate".
- If the destination is busy and queuing is enabled, the originating station is queued.
- If the destination is not a valid data device, the message "INVALID #" and reorder tone is returned.

Data: ADL Speed Call Originate

Description An ADL user can originate an ADL call using a personal Speed Call number.

Conditions None.

Programming The programming procedure to be followed varies depending upon the type of telephone to be programmed. Follow the appropriate procedure described below:

***SUPERSET 410* Telephones:**

At the user's telephone:

- Press SUPERKEY.
- Press a personal speed call key (if a previously defined speed call key is selected, the previously stored number is erased).
- Dial the ADL access code.
- If required, dial the account code access code and the account code.
- Dial the number of the data destination.
- Press SUPERKEY to save the number.

***SUPERSET 420* Telephones:**

At the user's telephone:

- Press SUPERKEY.
- Press the NO softkey until PERSONAL KEYS? appears in the display.
- Press the YES softkey.
- Press an unused personal key.
- Press the CHANGE softkey. SPEED CALL? appears in the display.
- Press the YES softkey.
- Dial the ADL access code.
- If required, dial the account code access code and the account code.
- Dial the number of the data destination.
- Press the SAVE softkey.
- Press SUPERKEY to return to the date and time display.

***SUPERSET 430* Telephones:**

At the user's telephone:

- With the handset on-hook, press SUPERKEY.
- Press the SPEED CALL softkey.
- Press an unused Line Appearance key.
- Dial the ADL access code.
- If required, dial the account code access code and the account code.
- Dial the number of the data destination.
- Check the number to be saved, if correct press the SAVE softkey.

Operation Operation varies depending upon the type of telephone as described below.

SUPERSET 410 and _SUPERSET 420_ Telephones:

To originate a data speed call:

- Press the programmed speed call key.
- Go on-hook when connect tone is heard.
- Call is transferred to the associated dataset.

To originate a data speed call with voice party on hold:

- Press the programmed speed call key.
- If ADL Autobaud is programmed, press Carriage Return key.
- When the connect tone is heard, press the CANCEL key to establish the data call and transfer the voice call back to the party on hold.

SUPERSET 430 Telephones:

To originate a data speed call:

- Press the programmed speed call key.
- Go on-hook when connect tone is heard.
- Call is transferred to the associated dataset.

To originate a data speed call with voice party on hold:

- Press the programmed speed call key.
- If ADL Autobaud is programmed, press Carriage Return key.
- When the connect tone is heard, press the BACK TO HELD softkey to establish the data call and transfer the voice call back to the party on hold, or press NEW CALL softkey to establish the data call and make a new voice call.

Data: Abbreviated Dial for ADL Calls

Description An ADL user can originate an ADL call using a system abbreviated dial number.

Conditions Reorder tone is returned if the user enters an invalid abbreviated dial number.

Programming Enable COS Option 245 (Abbreviated Dialing Access) in the class of service for the associated voice line.

Assign an access code to Feature 24 (Abbreviated Dial Access) in CDE Form 02 (Feature Access Codes).

Enter the desired abbreviated dial numbers into CDE Form 31 (System Abbreviated Dial Entry), or enter them from the attendant console. See *Attendant Abbreviated Dial Number Entry*.

Operation

To originate an abbreviated dial call:

- If the abbreviated dial number entry does not include the ADL access code, dial the ADL access code.
- Dial the abbreviated dial access code.
- Dial the desired abbreviated dial number (one to three digits).
- Hang up when connect tone is heard. The call is transferred to the associated dataset.

See also *ADL Call Originate* in this section.

Data: Associated Modem Line (AML)

Description

This feature allows voice only calls, data only calls, simultaneous voice/data calls, and alternating voice/data calls through the PABX. A standard ONS voice port is associated through CDE programming with one or more *SUPERSET* telephones. This ONS port is connected to data terminal equipment (DTE) via a modem. Data calls may be set up using the associated *SUPERSET* telephone.

Conditions

The following conditions apply to this feature:

- This feature is available only on *SUPERSET 420* and *SUPERSET 430* telephones. AML is not available on *SUPERSET 410* telephones.
- The only effect of AML on voice calls is that the SWAP (Trade Calls) softkey appears on calls to a station which has an associated modem line or on trunk calls.
- When two telephones assigned the same modem port converse, the SWAP (Trade Calls) softkey does not appear; data communication between these telephones is not possible.
- If the set user presses the SWAP (Trade Calls) softkey when the data port is already in use, the message DATA BUSY appears on the display and the user may hang up or press the CANCEL softkey.
- An AML data device may be associated with more than one telephone (only one can use it at a time); however, a telephone may be associated with only one AML data device.
- In a data call, both modems must operate at the same baud rate; one modem must be in originate mode, the other in answer mode.
- In a data call, the destination number must be followed by a pound sign (#). Otherwise, reorder tone will be returned to the modem.

Programming

Enable COS Option 607 (*SUPERSET* Telephone - Associated Modem Line) for the *SUPERSET* telephone's class of service.

Enter the associated modem ONS port extension number in the ASSOC field of the associated telephone entry in CDE Form 09 (Station/ *SUPERSET* Telephones).

If the terminal is to be able to dial a call on behalf of the associated *SUPERSET* telephone, enter the extension number of the associated *SUPERSET* telephone in the ASSOC field of the modem port entry in CDE Form 09 (Station/*SUPERSET* Telephones).

Operation

To make a Data Call from the *SUPERSET* telephone:

- Dial the number of the destination modem; when the destination modem answers, it returns a modem answer tone.
- Press the SWAP (Trade Calls) softkey.

Note: If the modem line is already connected to a call, or the modem is attempting to make a data connection, the message "DATA BUSY" appears. The CANCEL softkey is then the only key available.

- Put the modem on-line; the data call is now established.
- Press the HANG UP (or New Call) softkey.

The *SUPERSET* telephone is now available for voice calls. In this way, it is possible to have simultaneous voice and data calls. The data call terminates when the modem goes on-hook.

To make an alternating voice/data call:

- At the *SUPERSET* telephone, establish a normal voice call with the destination set.
- At both telephones, press the SWAP (Trade Calls) softkey to switch to data communication and put the modem on-line.
- When data communication is complete, press the SWAP (Trade Calls) softkey at both telephones to return to voice communications.

To make a data call from the modem port:

- Following the modem manufacturer's instructions, have the modem go off-hook and dial the destination modem number, followed by the pound sign (#). Note that if a modem dials the destination number WITHOUT the #, the modem will receive reorder tone, and the call will not complete.

To make a voice call on behalf of the associated voice line:

- Following the modem manufacturer's instructions of how to dial a call from the attached DTE, have the modem go off-hook and dial the destination telephone number, followed by an asterisk (*). The call is transferred to the *SUPERSET* telephone. The modem is available for data calls.

Data: Auto-Answer

Description

This feature supports automatic answering destination data devices. When the destination data device detects ringing, it answers, then signals to the calling device that it is ready to receive data.

Conditions

The destination data device must have its DTR signal low when idle.

Programming In the destination data device's CDE Form 11 (Data Circuit Descriptors) subform (Data Circuit Descriptor Options), set Action Taken If the Idle DTE Has DTR Low (Auto Answer) to one of the following:

- RI - the system toggles RI high 2.5 seconds then low 2.5 seconds until the dataset responds with DTR high.
- DSR - the system raises DSR. The dataset responds with DTR high.
- DCD - the system raises DCD. The dataset responds with DTR high.

Operation The destination data device must respond with DTR high within 1 minute of receiving the auto-answer signal.

Data: Automatic Data Route Selection (ADRS)

Description Outgoing trunk data calls are processed by the same Automatic Route Selection (ARS) system as voice trunk calls; see *Automatic Route Selection.*

Conditions None.

Programming Trunks should be grouped into trunk groups according to the maximum data rate they can support. For more information, refer to the *Automatic Route Selection and Toll Control Practice.*

Operation None.

Data Account Codes

Description Verified and non-verified account codes can be applied to data calls, internal, external or long distance. The account code appears in the DATA SMDR and TRUNK SMDR records.

Internal data calls generate a DATA SMDR record only; external data calls generate both a DATA SMDR and a TRUNK SMDR record.

Conditions The following conditions apply to this feature:

- For data or modem pooling calls, the user can enter an account code only during call setup. Only the last account code entered is stored in DATA SMDR and TRUNK SMDR records.
- If variable length account codes are enabled, a user dialing an account code must follow it with the # symbol. This is not necessary if the entered account code is twelve digits long. Thus, rotary dial telephones cannot have access to variable length account codes of less than twelve digits.
- Account code entry for incoming callers is permitted only on DISA or Tie trunks. When PABX dial tone is returned, a Special DISA caller can

enter another account code during call setup. Only this second code appears in the TRUNK SMDR and DATA SMDR reports.

- The user can include account codes in speed call numbers.

Programming

To use Verified Account Codes, enable System Option 05 (Verified Account Codes) in CDE Form 04 (System Options/System Timers); enter the required Account Codes into CDE Form 33 (Account Code Entry).

Set System Option 55, Account Code Length in CDE Form 04 (System Options/System Timers) to a value between 4 and 12, or to VARIABLE.

To require account code entry on internal ADL or DTRX data calls, enable COS Option 326 (Account Code Forced Entry - Data Internal Calls) for the originating dataset.

To require account code entry for incoming or internal modem pooling calls, enable COS Option 326 (Account Code, Forced Entry - Data Internal Calls) in the COS for the pooled modem.

Enable one of the following options in the COS form for the originating data device, as required, for outgoing calls:

- COS Option 327 (Account Code, Forced Entry - Data External Calls)
- COS Option 328 (Account Code, Forced Entry - Data Long Distance Calls).

Disable COS Option 700 (SMDR - Does Not Apply) in the class of service form for the data device.

Enable COS Option 808 (Special DISA) in the COS for the data device.

Disable COS Option 906 (Data SMDR - Does Not Apply) in the originating data device's Class of Service for ADL or DTRX calls.

To record incoming calls, enable COS Option 806 (SMDR - Record Incoming Calls).

For incoming and internal modem pooling calls, disable COS Option 906 (Data SMDR - Does Not Apply) in the originator's pooled modem's Class of Service.

Operation

ADL

From the set:

- Dial the ADL access code.
- Dial the account code access code.
- Dial the account code.
- Dial the destination.

DTRX

From the terminal, at the * prompt, enter:

CALL destination ACCOUNT account code.

DISA (Incoming)

When the PABX answers:

- Dial the DISA access code - the system returns dial tone.
- Dial the account code access code.
- Dial the account code.
- Dial the destination number.

Special DISA (Incoming)

When the PABX answers:

- Enter the account code - if it is valid, the PABX returns dial tone.
- Dial the destination number.

Incoming Tie Trunk or Internal Modem Pooling Caller

- Dial the account code access code.
- Dial the account code.
- Dial the destination number.

Direct-In-Line (to an Attendant)

When the attendant answers:

- Have the attendant dial the account code access code.
- Have the attendant dial the account code.
- Have the attendant dial the destination number.

Data Hunt Groups

Description

Datasets can be assigned to a Data Hunt Group. The system supports both terminal and circular hunting; see *Hunt Groups.*

Conditions

The following conditions apply to this feature:

- Only datasets can be assigned to a Data Hunt Group.
- The system can support a total of 50 hunt groups; a hunt group may have a maximum of 50 members.

Programming

Enter the dataset extension numbers into CDE Form 17 (Hunt Groups). When the first dataset is entered, the system sets the hunt group type to Data. The system then accepts only datasets as subsequent entries.

Specify TERMINAL or CIRCULAR hunting as desired.

Program an access code for the hunt group.

If desired, program a name for the data hunt group. This is useful for DTRX calls. See *DTRX Call By Name.* If desired, program an overflow point for the data hunt group.

Operation

Call the data hunt group using its access code or name. See *ADL Call Origination, DTRX Call Origination, and Modem Pooling.*

Data Peripherals

Description

The PABX supports several different data devices; MITEL also provides several data support devices for use with MITEL data devices.

MITEL data devices are proprietary limited-distance modems that provide data facilities for terminals, digital *SUPERSET* telephones, and other types of data devices. These data sets serve as the interface between the DNIC digital line card in the PABX, and the data devices connected to the system, transmitting data and control signals over a single twisted pair of wires. The datasets are available in two series - DATASET 1100 series and DATASET 2100 series.

DATASET 1103 Standalone supports asynchronous data communications at rates from 110 bps to 19.2 kbps. The data sets are RS-232C compatible, provide end-to-end error correction, auto-baud detection and auto-parity generation.

The DATASET 1100 series is available in two options:

- DATASET 1103 Standalone
- MILINK™ Data Module.

DATASET 2103 Standalone supports asynchronous data communications from 110 bps to 19.2 kbps, and synchronous data communications at rates from 1200 bps to 19.2 kbps. In asynchronous mode this data set has the same operating characteristics as the DATASET 1100 series. In synchronous mode, the DATASET 2103 uses X.31 protocol.

All datasets interface to a PABX digital line card circuit through a single pair.

Refer to the *Peripherals Devices Practice*, for information on the cabinet and shelf that is available for rack mounted data devices.

A Modem Interconnect Panel is available for installation within the DATACABINET 9000 cabinet; refer to the *Peripherals Devices Practice*. The Modem Interconnect Panel is required for Modem Pooling.

For information on using data devices as printer ports, see CDE Form 34 (Directed Input/Output).

For details on the SUPERCONSOLE 1000™ printer port, see CDE Form 34 (Directed Input/Output).

For details on data circuit descriptor options, refer to the *Customer Data Entry Practice.*

Refer to the *Engineering Information Practice*, for a description of the Digital Line Card.

Conditions

Refer to the *Peripherals Devices Practice.*

Programming

Program a Digital Line Card into the system's physical configuration table via CDE Form 01 (System Configuration).

Select options for the data circuit descriptors via CDE Form 11 (Data Circuit Descriptors).

Assign a circuit descriptor and select options for specific data circuits via CDE Form 12 (Data Assignment).

Operation

None.

Data Security

Description Data Security protects an established data call from receiving any intrusion or warning tones (for example, Campon or Override) that would interfere with the data signals present on the line. Any extension with Data Security in its COS cannot be overridden or receive campon tone; it may be camped on to, but is secure against any form of audio intrusion.

Conditions The following COS Options must be disabled to permit correct operation of Data Security:

- COS Option 242 (Repeated Camp-On Beeps)
- COS Option 607 (*SUPERSET* Telephone - Associated Modem Line).

Programming Enable COS Option 216 (Data Security) for the extension.

Operation None.

Data Station Message Detail Recording (Data SMDR)

Description The Data SMDR feature produces a detailed record of all internal and external data calls. The printout format is similar to the trunk SMDR record and includes "data call indicator", "type of data call" and "reason for disconnect" fields. Data SMDR records differ in appearance from voice SMDR records. This allows a call costing device to maintain separate totals for data and voice calls.

The "External Data Call Indicator" field shows EX if the data call used a pooled modem and a trunk.

The "Associated SMDR Record Indicator" field shows an asterisk when the data call has an associated "Trunk SMDR" record.

Refer to the *Station Message Detail Recording Practice* for details.

Conditions None.

Programming The following COS options and System Options/Timers can be selected:

COS Option 906 (Data SMDR-Does Not Apply)

COS Option 907 (Data SMDR-Extended Record)

COS Option 908 (Data SMDR-Overwrite Buffer)

System Option 39 (Data SMDR - Indicate Long Calls)

Specify a port to print Data SMDR records in CDE Form 34 (Directed I/O).

Operation None.

Data Station Queuing

Description

Data Station Queuing is similar to the Campon feature provided for telephones. An extension user encountering a busy dataset or data hunt group can camp on to the destination and remain in the queue until a destination becomes available or terminate the call at any time.

Conditions

The following conditions apply to this feature:

- The user can camp on to datasets, data hunt groups, trunk groups, or modem pool hunt groups only.
- The destination cannot be camped onto if it is busied out or in Do Not Disturb.

Programming

Enable COS Option 900 (Data Station Queuing) in the originating device's Class of Service.

Operation

Data Station Queuing is described below.

ADL from Industry-standard, *SUPERSET 401+*, and *SUPERSET 420* Telephones:

The telephone calls a busy data device:

- The telephone receives special busy tone for 2 seconds, then the telephone receives busy tone while it is queued.

ADL from *SUPERSET 420* Telephones:

The set calls a busy data device:

- The set receives special busy tone. The display shows BUSY. The WAIT softkey appears.
- Press the WAIT softkey within 10 seconds. While the set is queued, the display shows the extension number of the busy data device followed by the WAITING prompt (e.g., 4410 WAITING).

ADL from *SUPERSET 430* Telephones:

The set calls a busy data device:

- The set receives special busy tone. The display shows BUSY. The I WILL WAIT softkey appears.
- Press the I WILL WAIT softkey within 10 seconds. The display shows WAITING FOR and the extension number while the set is queued.

DTRX

Queuing is automatic. The originator receives the message "Call is Queued".

Data Transceiver (DTRX)

Description

The Data Transceiver (DTRX) allows the data user to set up a data call from a data terminal (DTE). The system has four DTRX channels; four users can communicate with the DTRX simultaneously. When the requested call is established, the DTRX is dropped, making it available for other data callers.

Users requesting a busy destination are queued until the destination is available. Up to 20 callers can queue. Subsequent callers requesting busy destinations are sent the message “System Busy, Try Again Later”.

When all DTRX channels are busy, callers queue for DTRX service. Up to 28 callers can queue, including those queued for destinations. Subsequent callers are prompted to try again later.

Conditions

The following conditions apply to this feature:

- A dataset connects each Data Terminal (DTE) to the system on a DNIC circuit.
- The DTE connected to the dataset must be an ASCII data device using an RS-232C interface.
- All DTRX commands (except ACCOUNT) may be abbreviated to its minimum unique character string.
- To connect to a data transceiver, a data device must have a DTE Profile number programmed in CDE Form 12 (Data Assignment).

Programming

The DTRX feature is associated with the DTE Profile and Data Circuit Descriptor options. Refer to *CDE Form 29 (DTE Profile) and CDE Form 11 (Data Circuit Descriptor Options).*

Enable or disable COS Option 900 (Data Station Queuing) as desired. This option, when enabled, allows automatic queuing to the destination when the originator encounters a busy data device.

Enable or disable COS Option 901 (DTRX Herald Display) as desired. When enabled, this option allows display of the programmable herald (e.g. Welcome to MITEL Data Switching).

If DTRX Herald Display is enabled, set COS Option 905 (DTRX Herald Text Select) to 1, 2, 3, or 4. The options are as follows:

[1] Welcome to MITEL *SX-200* Data Switching
[2] Welcome to MITEL Data Switching
[3] Welcome to Data Switching
[4] Data Call Switching

Enable or disable COS Option 902 (DTRX Message Code) as desired. Enabling this option displays the machine message code. The code is divided into three segments:

- the first character describes the application; either “G” for general messages, or “D” for DTRX application messages

- the next two digits identify the message category
- the last two digits identify the message

for example, D0102, where:

D = DTRX application
01 = Information
02 = Connected

Enable or disable COS Option 903 (DTRX Message Code Text) as desired. Enabling this option allows display of the message text (e.g. Connected); the message category is not displayed. The text also displays a user action if appropriate (e.g. System Busy, Please Wait). If the message code option is also enabled, the complete message is displayed.

Enable or disable COS 904 (DTRX Complete Message Text) as desired. This option displays the message code, the message category, and the text (e.g., D0207 Call Failure - Destination Not Ready).

Operation See *Data: DTRX Call Originate/Disconnect.*

Data: DTRX Call by Name

Description The Data Transceiver (DTRX) Call by Name feature allows the user to call a data hunt group or modem pool hunt group by name instead of number.

Conditions The hunt group must have a name programmed.

Programming Assign names to data and modem pool hunt groups in CDE Form 17 (Hunt Groups).

Operation At the DTE, in response to the * prompt:

1. Enter CALL or C.
2. Enter the name. (Example: C SYS)

or,

1. Enter CALL
2. Enter the modem pool hunt group name,
3. Enter the destination number. (Example: C MODEMS 5551234)

Data: DTRX Call Originate/Disconnect

Description

The DTRX Call Originate/Disconnect call is initiated and controlled from a terminal keyboard.

Conditions

The following conditions apply to this feature:

- Unless the user originates the call from the keyboard, the device is assigned the last baud rate used. (The default baud rate is used if Always Use Default Baud Rate When Called is enabled in the device's data circuit descriptors).
- Autobaud is invoked when keyboard origination is used. (Keyboard Origination (Autobaud) must be enabled in the device's data circuit descriptors).
- Heralds and call progress messages are only displayed if programmed; see *Data Transceiver* (DTRX).

Programming

Assign the originating dataset a DTE Profile number in CDE Form 12 (Data Assignment).

Program the Data Circuit Descriptor Options in CDE Form 11 (Data Circuit Descriptors) subform.

Program the DTE Profile in CDE Form 29 (DTE Profile).

Operation

DTRX operation is described below.

DTRX Originate

- Press the RETURN or ENTER key on the DTE keyboard. (Keyboard Origination must be enabled in the data circuit descriptors).

or

- Press the ATTENTION key on the dataset.

or

- Switch power ON at the data device (i.e., raise DTR). (The option, "Originate a DTRX Call With Low to High Transition of DTR" must be enabled in the data circuit descriptors).

The system attempts to seize DTRX. If DTRX is available, the * prompt is displayed.

- Enter CALL (or C).
- Enter the destination directory number at the keyboard, followed by RETURN or ENTER.
- DTRX sends the digits to system's main control for call processing.
- Call progress messages are supplied, if programmed.

DTRX Disconnect

DTRX is dropped when the originator is connected to the destination. To disconnect the current DTRX-originated data call and reconnect to DTRX:

- Switch the DTE off and on. (The option "Originate a DTRX Call With Low to High Transition Of DTR" must be enabled in the Data Circuit Descriptors). It must remain off for longer than the duration of the DTR OFF Disconnect timer duration.

  or
- Press the BREAK key (if enabled).

  or
- Press the PBX Attention Character key (if programmed).

  or
- Press the ATTENTION key on the dataset.

  To disconnect from DTRX and return to idle state:
- At the * prompt, enter BYE or <CTRL E>. "Goodbye" message is displayed.
- While connected to DTRX, wait for the DTRX inactivity time out. (This is programmed in the DTE Profile).
- Turn off the data terminal (DTR dropped).
- Press the DISCONNECT key on the dataset.

The first two disconnect methods are possible only while communicating with the DTRX, i.e., no call in progress. The other methods can be used at anytime and also disconnect the current call. Also, the system can automatically disconnect users who remain inactive too long while connected to a destination. If programmed, the Session Inactivity Disconnect Timer controls the time-out period.

Data: DTRX Help

Description

The Data Transceiver (DTRX) Help feature provides on-line help. The topics covered include:

- CALL
- HELP
- ACCOUNT
- MONITOR (ACD only)
- BYE

Conditions

None.

Programming

None.

Operation

At the * prompt, type HELP and the topic name, followed by <RETURN>.

DTRX responds with the required information and returns to the * prompt.

If the topic is not known, just type HELP followed by <RETURN>.

DTRX responds with a list of topics and the ? prompt. Type the required topic name, followed by <RETURN>.

To exit from Help, press <RETURN> at the ? prompt.

Data: DTRX Hotline

Description A dataset can be programmed to connect automatically to a predetermined destination when a user originates a DTRX data call.

Conditions Unlike normal DTRX call origination, if the user presses ATTENTION or BREAK while the dataset is queued or connected to its hotline destination, the system drops the call and does not reconnect the user to DTRX.

Programming Assign the hotline number in CDE Form 12 (Data Assignment).

Operation See DTRX *Call Originate/Disconnect.*

Data: DTRX Messages

Description The Data Transceiver (DTRX) Messages feature provides several messages to inform the user of call progress and error conditions.

Table 2-4 DTRX Messages

Message	Meaning
D0101 Information - Ringing	The destination is available and is now ringing.
D0102 Information - Connected	The user is now connected to the destination. This is the last message displayed in a successful call. The DTRX is now dropped.
D0201 Call Failure - Busy	The destination is busy and queuing is not enabled. The user is then returned to the DTRX * prompt.
D0203 Call Failure - Invalid Number, Check Directory	The directory number does not exist. The user is returned to the DTRX * prompt.
D0204 Call Failure - Out of Order, Call Communications Department	The destination data device has been busied out. This can indicate a fault in the destination data device.
D0205 Call Failure - No Answer	The destination auto answer dataset did not respond. The user is returned to the DTRX * prompt.
D0207 Call Failure - Destination Not Ready	The DTR signal of the destination device (non auto answer) indicates it is not ready. The user is returned to the DTRX * prompt.
D0208 Call Failure - Destination Powered Off, Try Again Later	The destination dataset has no power or is not connected to the PABX.

Page 1 of 3

Table 2-4 DTRX Messages (continued)

Message	Meaning
D0209 Call Failure - Destination Not Compatible	The destination is a voice device and thus incompatible.
D0210 Call Failure - Address Incomplete, Check Directory	Further addressing information is required to complete the call. The user is returned to the DTRX * prompt.
D0211 Call Failure - Modem Failure, Try Again	The setup to the Pooled Modem has failed. The user is returned to the DTRX * prompt.
D0212 Call Failure - No Modem Available, Try Again Later	There are no idle members in the Modem Pool Hunt Group and queuing is not enabled. The user is returned to the DTRX * prompt.
D0218 Call Failure - Destination Not Compatible	The destination device's data rate is not compatible with that of the source device. (Speed conversion is allowed if XON/XOFF Flow Control is enabled). The user is returned to the DTRX * prompt.
D0301 Call Not Allowed - No Access, Call Communications Department	Access to the destination device is not allowed. This can be due to tenanting, because the data station is programmed as Receive Only, or because the destination is not accessible by the caller. The user is returned to the DTRX * prompt.
D0503 Facility - Call is Queued	The destination is busy and queuing is enabled. The "Queue Position" message follows if the destination is a dataset. The DTRX is dropped until it is needed to display the call progress messages, or the next queue position. When the destination device becomes free, the "Ringing" and "Connected" messages follow.
D0504 Facility - Queue Position nn	This message occurs after every system clock minute. It indicates the user's new queue position, if changed. If the position has not changed, "." is appended to the displayed position number. (e.g. Queue Position 02.) The DTRX is obtained for this write operation and is dropped after it is completed.
D0506 Facility - Destination Queue Full	There is no room in the queue. The user is returned to the DTRX * prompt.
D0507 Facility - Invalid Account, Check Account Code	The entered account code is programmed as inactive (Verified Account Code), is invalid or is not the length specified in CDE Form 04 (System Options/System Timers).
D0512 Facility - Facility Not Available	For ACD only, when "MONITOR ACD" is selected. This message is displayed if the user enters MONITOR ACD in an ACD software load and System Option 104 (Automatic Call Distribution) is disabled.
D0517 Facility - Change Your Speed to [xxxxx]	The system has changed the data rate of the dataset to be compatible with the destination. The user must change the terminal's data rate as indicated. The DTRX is dropped at this point.
G0101 Information - Welcome to MITEL Data Switching	If DTRX Herald is not enabled, only the "*" (asterisk) prompt appears.
G0104 Information - Goodbye	DTRX dropped due to timeout or user request ("BYE" command or <CTRL E> at DTRX * prompt).
G0105 Information - System Busy, Please Wait	There are no DTRX links currently available.
G0108 Information - Call Disconnected	The system has disconnected the call. This could be due to a link failure. Check the DATA SMDR reports for more information.
G0109 Information - Call Disconnected By User	The user disconnected the call using the Disconnect key.

Page 2 of 3

Table 2-4 DTRX Messages (continued)	
Message	**Meaning**
G0110 Information - Call Disconnected By Destination	The destination data station disconnected the call.
G0201 System Error - System Busy, Try Again Later	There are no free DTRX work areas in system memory.
G0301 User Error - User Has been Inactive for too Long	DTRX or data call dropped due to inactivity timer expiry. If in DTRX, "Goodbye" message follows.
G0302 User Error - Unrecognized Command, Check User Manual	The system could not understand some part of the input command string. Consult the HELP facility.
G0303 User Error - Ambiguous Command, Check User Manual	The command the user entered was too abbreviated. The system could not determine which command was intended.
G0307 User Error - No Help Information Available	The help facility has no information on the command specified.
	Page 3 of 3

Data: Modem Pooling

Description

Modem Pooling is a data application feature using voice communication lines. Modem pooling permits communication between modems and DATASET 2100 Series datasets. Each pooled modem consists of a modem and a dataset. The dataset connects to a Digital Line Circuit, and communicates with datasets within the PABX. The pooled modem connects to an ONS or OPS Line Circuit, which communicates directly with internal modems, or with external modems via trunks.

A collection of pooled modems organized into a hunt group is called a modem pool hunt group.

A dataset user can access a modem pool hunt group to obtain a pooled modem for data transmission to a modem user over the public network. If the modem pool hunt group or the trunk group is busy, the user can queue for it; see *Modem Pooling Queuing*.

A modem user, internal or external, can access a modem pool hunt group to obtain a pooled modem for communication with a dataset within the PABX. If the modem pool hunt group or the destination dataset is busy, the user can queue for it; see *Modem Pooling Queuing.*

Modem Pooling calls may also be made by using system abbreviated dial numbers.

Conditions

The following conditions apply to this feature:

- COS Option 217 (Direct to ARS) is ignored for outgoing data calls.

- If the originator does not dial a specific modem pool hunt group access code, a pooled modem is selected from the default modem pool hunt group, if programmed.
- On incoming or internal calls, the system prints TRUNK SMDR reports for the trunk and Data SMDR reports for the pooled modem if they are enabled in the respective COS. The reports can be correlated by the originating device's identity and the time the call was started. COS Option 806 (SMDR - Record Incoming Calls) must be enabled to generate SMDR records for incoming calls. COS Option 906 (Data SMDR - Does Not Apply) must be disabled for pooled modems to generate SMDR records for incoming calls.
- On external ADL or DTRX-originated calls, the system generates a TRUNK SMDR report for the pooled modem and a Data SMDR report for the originating data station.
- Only DATASET 2102 and 2103 datasets can be used as modem pool devices.
- All pooled modems in the Modem Pool Hunt Group must have the same Operating Mode (all Originate or all Answer or all Both).
- The pooled modems in the default modem pool hunt group can only be programmed for Both (Answer and Originate) mode.
- A dictation trunk cannot be used for an outgoing modem pool.

Programming

Programming varies depending upon the modem configuration as described below.

Internal/Incoming Modem Users

Modem users can be a trunk, station (ONS/OPS), *SUPERSET* telephone (with acoustically-coupled modem), or a console (can only transfer a modem).

To permit a modem user to make calls to datasets, enable COS Option 704 (Incoming/Internal Modem Pooling Access) in its COS. This is required because the system normally bars calls between voice and data devices. Modems of pooled modems within the modem pool hunt group should be in auto-answer mode.

If desired, enable Option 900 (Data Station Queuing) in the modem user's COS; see *Modem Pooling Queuing*.

Pooled Modem

Pooled modems are programmed in CDE Form 36 (Modem Assignment); for each pooled modem, assign:

- Bay/Slot/Circuit location of its ONS (or OPS) Line (system-supplied),
- Data Circuit Descriptor Number (CDN) that applies to its dataset,
- Bay/Slot/Circuit location of its dataset's Digital Line,
- extension number,
- tenant (default is 1),
- DTE Profile Number (optional),

- Class of Service (COS) (default is 1) and Class of Restriction (COR) (default is 1),
- mode: Answer, Originate, or Both (default is Both),
- auto-dial capability: Auto dial or Non Auto dial (default is Non Auto dial).

Modem Pool Hunt Group

- Program the extension numbers of the required pooled modems to a Hunt Group in CDE Form 17 (Hunt Groups).
- Assign an access code to the modem pool hunt group.
- If desired, make this the default modem pool hunt group. Users do not need to dial the modem pool access code. Only one default modem pool hunt group can be programmed in the system.
- Only pooled modems of the same operating mode can be assigned to the same modem pool hunt group.
- Assign a hunt group name to allow access via the DTRX Call by Name feature.
- To provide immediate DTRX access for incoming or internal modem pooling callers, enable DTRX for the modem pool hunt group in the Options subform of CDE Form 17 (Hunt Groups). If DTRX is disabled, DTRX callers must wait for the interdigit time-out (up to 15 seconds) before receiving a response. The DTRX option does not affect outgoing modem pooling calls. The DTRX option can only be enabled if DTE is programmed in the pooled modem and the mode of the hunt group is Answer or Both.

Operation

Internal Modem User (Station, *SUPERSET* Telephone, or Console)

- Dial:
  - Data Station or Data Hunt Group access code (where default modem pool hunt group is selected, if programmed)
    or,
  - Modem Pool Hunt Group access code for DTRX access
    or,
  - Modem Pool Hunt Group access code plus Data Station or Data Hunt Group access code.
- When answer tone is heard, the modem pooling call is established. Put the modem on line.

Incoming DISA Trunk Modem Call

- From the Public Network, using a DTMF telephone, dial the DISA telephone number. The PABX answers and returns dial tone.
- Dial the DISA Access Code (normal DISA trunk) or Account Code (if a Special DISA trunk). Once access is allowed, the system returns dial tone.
- Dial:
  - Data Station or Data Hunt Group access code (where default modem pool hunt group is selected, if programmed)
    or,

- Modem Pool Hunt Group access code for DTRX access
  or,
- Modem Pool Hunt Group access code plus Data Station or Data Hunt Group access code.

- When answer tone is heard, the modem pooling call is established. Put the modem on line.
- If DTRX access is used, enter CALL followed by the destination data number or name at the DTRX prompt.

Incoming Tie Trunk Modem Call

- Dial the TIE number to reach the remote PABX.
- Dial:
  - Data Station or Data Hunt Group access code (where default modem pool hunt group is selected, if programmed)
    or,
  - Modem Pool Hunt Group access code for DTRX access
    or,
  - Modem Pool Hunt Group access code plus Data Station or Data Hunt Group access code.
- When answer tone is heard, the modem pooling call is established; put the modem on line.
- If DTRX access is used, enter CALL followed by the destination data number or name at the DTRX prompt.

Incoming DID Trunk Modem Call

The central office passes on to the PABX a pre-arranged number of the caller's dialed digits. The resulting digit string formed is one of the following:

- Data Station or Data Hunt Group access code where default modem pool hunt group is selected, if programmed.
- Modem Pool Hunt Group access code for DTRX access.
- Modem Pool Hunt Group access code plus Data Station or Data Hunt Group access code.

- When answer tone is heard, the modem pooling call is established; put the modem on line.
- If DTRX access is used, enter CALL followed by the destination data number or name at the DTRX prompt.

Incoming Direct-In-Line (DIL) Modem Call

- From the public network, dial the DIL telephone number.
- The call is directed to the programmed Answer Point,
  - a data station or a data hunt group where the programmed default modem pool hunt group is selected
    or,
  - a modem pool hunt group access code for DTRX access.

Transferring a Modem Pooling Call (Incoming/Internal Call On Hold)

- The transferor must have COS Option 704 (Incoming/Internal Modem Pooling Access) enabled; the transferor can be a station, *SUPERSET* telephone, or a console.
- Dial:
  - Data Station or Data Hunt Group access code (where default modem pool hunt group is selected, if programmed) or,
  - Modem Pool Hunt Group access code for DTRX access or,
  - Modem Pool Hunt Group access code plus Data Station or Data Hunt Group access code.
- Once the call is successful, the pooled modem returns answer tone.
- When answer tone is heard, transfer the call as follows:
  - press RELEASE key at console or,
  - press RELEASE ME softkey or hang up at a *SUPERSET 420* or *SUPERSET 430* telephone. or,
  - hang up at an industry-standard telephone, a *SUPERSET 401+*, or a *SUPERSET 410* telephone.
- To cancel the modem pooling call setup, proceed as follows:
  - press CANCEL key at console or,
  - press BACK TO HELD softkey at a *SUPERSET 420* or a *SUPERSET 430* telephone or,
  - press the TRANS/CONF key at a *SUPERSET 410* telephone or,
  - press the FLASH key at a *SUPERSET 401+* telephone or,
  - flash the hook switch at an industry-standard telephone.

External ADL Modem Pooling Call

- Dial the ADL access code.
- Dial the modem pool access code, unless the default modem pool is to be used.
- Dial the external destination (dial the ARS leading digit string and the destination number). Re-order tone is returned if the data connection cannot be made.
- When answer tone is heard, hang up the voice telephone. The data call is established.

External DTRX Modem Pooling Call

- Originate a DTRX session (see *Data: DTRX Call Originate/Disconnect*)
- Enter "CALL" followed by the Modem Pool Hunt Group access code, (omit the Modem Pool Hunt Group access code if the default modem pool hunt group is to be used),
- Enter the destination name or number. (For external calls, dial the ARS leading digit string and the destination number).
- Press RETURN.
- The DTE displays RINGING while the destination is ringing and CONNECTED when the data connection is established.

Data: Modem Pooling Queuing

Description

Modem Pooling callers who encounter a busy modem pool hunt group, destination dataset, data hunt group, or trunk group can queue for it.

Conditions

The following conditions apply to this feature:

- DID trunk callers cannot queue. They receive busy tone unless an answer point is programmed in the Call Rerouting table for DID Recall Points on Busy.
- DIL trunks can queue for a modem pool hunt group, dataset, or data hunt group, but they receive ringback tone, not busy.
- Trunk callers have higher priority than internal callers for queuing.

Programming

Enable COS Option 900 (Data Station Queuing) in the caller's Class of Service.

Operation

Operation varies depending upon the modem configuration as described below.

Incoming Modem Pool Queuing

The modem pool hunt group or destination is busy. Special busy tone is returned for 2 seconds, followed by busy tone when the campon occurs. ADL Outgoing Modem Pool Queuing

The modem pool hunt group or trunk group is busy. Special busy tone is returned for 2 seconds, followed by busy tone when the campon occurs.

DTRX Outgoing Modem Pool Queuing

The originator receives the message "Call is Queued".

Device Interconnection Control

Description

This feature provides a means of disallowing connection between devices of different types. The feature is primarily intended to control trunk interconnection but applies to other devices as well. This is intended to provide a method of meeting interconnection regulations imposed by various regulatory authorities.

The checks apply when a device calls another device, when a transfer (supervised or unsupervised) is attempted and when the console attempts to perform operations on devices. The sections in this practice mention Device Interconnection checks where they apply.

See *Flash Control* for details on controlling the ability of extensions to flash when extensions are involved with trunk calls.

There are three aspects of the feature: Tenant Interconnection, the Device Interconnection Table, and COS Options.

Tenant Interconnection, CDE Form 05 (Tenanting Interconnection Table) is described in the *Tenanting Practice*. The check for tenant connection is done when the Device Interconnection Table check is done. Other aspects of interconnection are done only if the tenant connection check passes.

The Device Interconnection Table is a table that the installer can program to limit interconnection from one device type with another and with itself. This restriction can be applied in either direction if desired. The Device Interconnection Table alone applies when the two devices in question are not both trunks. When two trunks are to be connected together, the Device Interconnection Table restrictions apply. In this case, COS options may apply additional restrictions.

The COS options apply additional checks when trunks are to be connected together. The COS options do not override the Device Interconnection Table but add finer control to the table. The options apply when a third party (a console, extension or trunk) attempts to leave two or more trunks connected together by a supervised transfer. The COS of the third party is used for the checks.

The following rules apply for the COS option checks:

- CO to CO trunk checks COS Option 313 (CO Trunk To CO Trunk Connect)
- CO to Tie checks COS Option 314 (CO Trunk To Tie Trunk Connect)
- Tie to CO checks COS Option 314 (CO Trunk To Tie Trunk Connect) and 319 (Extension non-CO Trunk To Trunk Connect)
- CO to DID checks COS Options 315 (CO Trunk To DID Trunk Connect)
- DID to CO checks COS Options 315 (CO Trunk To DID Trunk Connect) and 319 (Extension Non-CO Trunk To Trunk Connect)
- Tie to Tie checks COS Option 316 (Tie Trunk To Tie Trunk Connect) and 319 (Extension Non-CO Trunk To Trunk Connect)

- DID to Tie, Tie to DID checks COS Option 317 (Tie Trunk To DID Trunk Connect) and 319 (Extension Non-CO Trunk To Trunk Connect)
- DID to DID checks COS Option 318 (DID Trunk To DID Trunk Connect)
- For DISA trunks, the underlying circuit descriptor trunk types are checked against the device interconnection COS Options.
- Incoming Tie trunks have no COS option checking.

Conditions

The following conditions apply to this feature:

- When a device hangs up from a three party call, the Device Interconnection feature applies between the remaining parties. The third party that is used in the COS option checks is the party that hung up. If the two remaining parties cannot connect, then the call is cleared down.
- When a device hangs up from a four or five party conference call, if there are only trunks left in the call, the Device Interconnection feature is applied between each trunk in the call (both ways). The third party that is used in the COS option checks is the party that hung up from the call. If at least one trunk can connect to one other trunk in the conference then the Device Interconnection check passes. If no trunks can connect to any other trunks in the call then the call is completely cleared down.
- If the console is not allowed connection to a device, it cannot exercise features affecting that device.
- Device Interconnection applies directly to trunks, stations, sets, data stations, modem pools and consoles.
- Connection is always allowed to night bells, call announce ports and LDNs.
- Device connection checking is done when an ACD position is directly dialed. The checks apply between the caller and the set where the position is logged in.
- No checking is done for ACD calls from an ACD path.
- No checking is done between callers and members of recording or auto attendant hunt groups.
- Checking is done with hunt groups using the tenant of the first programmed member of the hunt group.
- Checking is done with logical lines using the tenant of the *SUPERSET* telephone where the first appearance of the line resides.
- When a trunk is accessing ARS directly, or for trunks transferred into ARS before a trunk is seized, the COS used is that of the trunk itself.
- When an extension times out from reorder tone with a conference on hold, the above rules apply to the call that is left; it is as if the extension had hung up from the call.

Programming

Set the interconnection rules in CDE Form 30 (Device Interconnection Table) as required.

Enable the appropriate trunk connection COS options, see *Description.*

Operation

None.

Dial Tone Disable

Description Assignment of this feature to a dial-in trunk suppresses dial tone on an incoming trunk call. If this feature is assigned to an extension, the extension does not receive dial tone whenever dialing is initiated.

Conditions When applied to DISA trunks, the option suppresses the initial dial tone returned for dialing the DISA feature access code. The second dial tone, after a successful code is dialed, is not affected by the option.

Programming Enable COS Option 701 (No Dial Tone) for the device.

Operation None.

Dial Tone - Discriminating

Description An extension having a feature enabled that prevents calls from ringing the extension hears a distinct dial tone (350/440 Hz, 400 ms on, 100 ms off for six cycles, then continuous tone) when going off-hook to make a call. These features include Do Not Disturb, Call Forwarding - Follow Me, or Call Forwarding - I Am Here.

Conditions The following conditions apply to this feature:

- The tone heard is the same one that is used for transfer dial tone.
- COS Option 701 (No Dial Tone) must be disabled.

Programming Enable COS Option 219 (Discriminating Dial Tone) for the extension.

Operation None.

Dictation Trunks

Description This feature indefinitely extends the dialing stage on a trunk to allow tone-to-pulse or pulse-to-tone conversion, based on trunk circuit descriptors, of all digits dialed during a trunk call. Without the feature, only DTMF signaling is possible from extensions so equipped.

Conditions The following conditions apply to this feature:

- A DTMF receiver circuit is used for the duration of the call.
- Dictation trunks cannot be put on hold but callers can be transferred to dictation trunks.
- The dialing stage cannot be terminated on the trunk during the call.
- A dictation trunk can't be used for an outgoing modem pool.

Programming Set the switches on the analog CO trunks cards to the appropriate setting; refer to the *Installation Information Practice.*

Enable appropriate options for the trunks via CDE Form 13 (Trunk Circuit Descriptors); also see *Trunk Circuit Descriptor Options.*

Operation Dial the necessary digits to access the dictation trunk. The system keeps the extension in dialing mode on the trunk and converts all digits dialed to the appropriate format for the trunk selected (tone or pulse).

DID/Dial-in/Tie Intercepts

Description This feature allows a customer to specify that all DID and Dial-in Tie Trunk calls directed to a busy extension (or one which does not answer within a selected time period) are rerouted to a call rerouting point. As well, the trunks can be programmed to be redirected immediately or to be redirected under certain error conditions. See *Recall* for how this fits in with general recall operation.

When a DID trunk dials in, the following conditions are tested in the following order:

- If calling an LDN and the DID Attendant Access Night point is defined, then the trunk is rerouted to that point.
- If calling a station, *SUPERSET* telephone, hunt group, or logical line and the DID Routing Point is defined, then the trunk is rerouted to that point.
- If calling a busy industry-standard telephone, *SUPERSET* telephone, or logical line and the extension has forwarding - busy/always programmed, then the trunk is forwarded.
- If calling a busy industry-standard telephone, *SUPERSET* telephone, logical line, or hunt group and the DID Recall Points on Busy is defined, then the trunk is rerouted to that point.
- If calling a station or *SUPERSET* telephone with Do Not Disturb enabled and a Do Not Disturb Intercept point is defined, then the trunk is rerouted to that point.
- If the station or *SUPERSET* telephone called does not answer and the extension has forwarding - no answer programmed, then the trunk is forwarded.
- If the industry-standard telephone, *SUPERSET* telephone, hunt group member, or logical line called does not answer and the DID no answer point is defined, then the trunk is rerouted to that point.

The above applies, except for DND intercept, for Tie trunks, but using the equivalent Tie routing points.

Conditions

The following conditions apply to this feature:

- If an illegal number is dialed or an illegal feature is attempted and the DID Intercept Routing point is defined, then the DID trunk is rerouted to that point. The tenant of the trunk is used. The same applies for Tie trunks and the Dial-in Tie Intercept point. If no point is specified, reorder tone is given.
- Calls routed to an LDN or to a console through the DID/Tie intercepts are identified as intercepted calls on the console display, when they are answered.
- Calls routed to an LDN through DID Intercept busy/no answer routing point, show the tenant name of the called party when answered at the console.
- If a vacant number is dialed and the DID Vacant Number Routing point is defined, then the DID trunk is rerouted to that point. The tenant of the trunk is used. The same applies for Tie trunks and the Dial-in Tie Vacant Number Routing point.
- The DID or Tie no answer point applies in recall situations if the trunk was ringing a party and has not been answered yet by any device in the system (excluding recordings). See *Last Number Redial.*
- The reroute for no answer does not occur if calling a console, LDN or Night Bell. It also does not occur if already ringing the reroute point.
- The reroute point on busy or no answer cannot be a console.
- The tenant of the called party is used to determine the rerouting point. When a logical line is called, the tenant of the first appearance of the line is used. When a hunt group is called, the tenant of the first member of the hunt group is used. When an LDN is called, the tenant of the console with the lowest Bay/Slot/Circuit plid where the LDN appears is used.
- The answer time timing for reroute/no answer is taken from COS Option 115 (Attendant Timed Recall - No Answer) of the trunk used for the reroute on no answer. If the time value is zero, then no reroute is done.
- The application of busy and attendant night access points applies only when the trunk initially dials in. The no answer point applies when ringing any extension unless the trunk has been answered. This includes ringing an extension after a call forward no answer on the initial dialed destination.
- COS Option 210 (Call Forwarding Inhibit on Dial-In Trunks) for the called party is checked on the always, busy, and no answer routing points. If the COS option is enabled for the point, then the reroute is not done.
- Extensions may be restricted from receiving DID trunk calls directly from DID trunks, by selecting COS Option 226 (Inward Restriction - DID) for the extension.
- For DID and Tie calls, the routing for all calls is done once dialing is completed.
- The busy and no answer points do not apply when calling consoles, LDNs, or Night Bells.

- If the console is programmed as the DID/TIE Busy/No Answer recall point (in Form 19, Call Rerouting Table), DID/TIE calls that are redirected to the console will not go to Attendant Automatic Overflow.

Programming Enter the desired rerouting points via CDE Form 19 (Call Rerouting Table) in the appropriate tenant. Refer to *Form 19 - Call Rerouting Table* in the *Customer Data Entry Practice*, for a list of the call rerouting options.

To disable rerouting when a particular station or *SUPERSET* telephone is called, enable COS Option 210 (Call Forwarding Inhibit on Dial-In Trunks) for the required extension(s).

See Inward Restriction (DID) for details on blocking DID trunk calls altogether to a particular extension.

Operation None.

Digit Translation

Description You can program the PABX to provide one of four Digit Translation Plans for rotary telephone sets. The default Digit Translation Plan is Plan 0.

Table 2-5 Digit Translation Plans

Digit	Plan 0	Plan 1	Plan 2	Plan 3
1	1 pulse	2 pulses	9 pulses	10 pulses
2	2 pulses	3 pulses	8 pulses	9 pulses
3	3 pulses	4 pulses	7 pulses	8 pulses
4	4 pulses	5 pulses	6 pulses	7 pulses
5	5 pulses	6 pulses	5 pulses	6 pulses
6	6 pulses	7 pulses	4 pulses	5 pulses
7	7 pulses	8 pulses	3 pulses	4 pulses
8	8 pulses	9 pulses	2 pulses	3 pulses
9	9 pulses	10 pulses	1 pulse	2 pulses
0	10 pulses	1 pulse	10 pulses	1 pulse

Conditions Only one translation plan may be in effect at a time.

Programming Select the desired translation plan via System Option 46 (Digit Translation Plan).

Operation None.

Direct-in Lines (DIL)

Description

This feature allows non-dial-in type trunks to ring specific answering points, rather than at the attendant console; this may vary with night service changes. An answering point may be one of the following in addition to a console or LDN:

- an ACD Path
- an extension number (industry-standard telephone, *SUPERSET* telephone, logical line)
- a hunt group access code
- a modem pool access code
- a night bell access code.

See *Recall* for effects on recall.

Conditions

Refer to *Trunk Operation - Non-dial-in CO.*

Programming

In CDE Form 14 (Non-Dial-In Trunks) enter the desired answering points for each of the DAY, NIGHT1 and NIGHT2 modes of operation.

Operation

None.

Direct Station Select (DSS) Key

Description

You use the Direct Station Select (DSS) keys on a *SUPERSET* telephone, or Programmable Key Module (PKM) in conjunction with the Busy Lamp Field (BLF) indicators located beside each DSS key. A DSS key is used to call, and connect calls to a device. The BLF indicator corresponding to the DSS key indicates the busy status of the device.

The DSS key operates when the line appearance of the device is:

- idle - pressing the DSS key will initiate a call to the associated device.
- talking to another party that can be put on soft hold - pressing the DSS key will put the other party on consultation hold, and initiate a call to the associated device.
- listening to dial tone - pressing the DSS key will initiate a call to the associated device.

The DSS key is inoperable in all other states. For example, if the device is talking to one party with another party on soft hold, the DSS key will have no effect.

Secretarial Option

The BLF indicator associated with the DSS key always displays the status of the BLF party, thus limiting the need for supervised transfers. An

Automatic Transfer option is available to users with the SECRETARIAL enabled on their DSS keys. To transfer a call, the DSS key is pressed - the call is transferred with an automatic release to the BLF party.

If the BLF party is idle, the transferred party rings it. If the BLF party is busy, the system camps on the transferred call. If the two parties are not allowed to be connected together because of device interconnection restrictions, the system prevents the automatic transfer; i.e., the same conditions apply to these transfers as to any others.

See also *Busy Lamp Field* in this document.

Conditions

The following conditions apply to this feature:

- DSS keys will not function on line appearances associated with trunks.
- DSS keys will not function when any feature associated with a template such as Hotel/Motel and Automatic Call Distribution are placed on the line appearance.

Programming

***SUPERSET* telephones**

To program DSS keys:

- While in CDE Form 09 (Station/*SUPERSET* Telephones) or CDE Form 45 (Key System Telephones), position the cursor on the desired device and select the EXPAND SET softkey (Expand Set sub-form is displayed).
- Move cursor to the desired key.
- Move cursor to the TYPE column (use TAB key if necessary).
- Select the BLF/DSS softkey (Type becomes "Busy Lamp").
- Press the TAB key twice to get to the DSS column
  OR
  To program the key with Secretarial Option, press the TAB key once to get to the SEC column. Then press the SECRETARIAL softkey (YES appears in the SEC column). Press the TAB key again to get to the DSS column.
- Select the DSS softkey (YES appears in the DSS column).
- Press the TAB key to get to the EXT NUM column.
- Enter the extension number of the device to be monitored by the BLF appearance.
- Select the ENTER softkey.

Programmable Key Modules

To program DSS keys:

- Display CDE Form 09 (Stations/*SUPERSET* Telephones) or CDE Form 45 (Key System Telephones).
- Move the cursor to the bay/slot/circuit number of the *SUPERSET 410*, *SUPERSET 420*, or *SUPERSET 430* telephone. An asterisk (*) appears to the left of the set type (410, 420, 430) if PKMs are programmed for the set.

- Select the EXPAND PKM softkey. "ENTER PKM NUMBER (1-3):" appears in the command line.
- Type the address setting (1, 2, or 3) of the desired PKM in the command line.
- Select the Enter softkey. The system displays the Expand PKM Set Subform for the PKM. The Expand PKM Set Subform allows you to program the functions of the 30 available PKM keys. Keys 31 and 32 in the form are listed as RESERVED since the PKM only has 30 keys.
- Move cursor to the desired key.
- Move cursor to the TYPE column (use TAB key if necessary).
- Select the BLF/DSS softkey (Type becomes "Busy Lamp").
- Press the TAB key twice to get to the DSS column
  OR
  To program the key with Secretarial Option, press the TAB key once to get to the SEC column. Then press the SECRETARIAL softkey (YES appears in the SEC column). Press the TAB key again to get to the DSS column.
- Select the DSS softkey (YES appears in the DSS column).
- Press the TAB key to get to the EXT NUM column.
- Enter the extension number of the device to be monitored by the BLF appearance.
- Select the ENTER softkey.

Operation

To call the BLF party:

- Press the DSS key.

To transfer a call to a station (without Secretarial option):

- Press the DSS key.
- press the RELEASE key/softkey.

To transfer a call to a station (Secretarial option):

- Press the DSS key.

Direct to ARS

Description

This option allows an industry-standard telephone to be directly routed to ARS without dialing, and for other devices to be routed after dialing a valid account code. The system dials up to five digits automatically for the extension.

Conditions

The following conditions apply to this feature:

- For industry-standard telephones, the feature only applies when the station initially starts a call. Once the initial trunk call has been established, normal operation resumes (for example, flashing is not affected).
- This feature applies to all devices that can access account codes, only after a valid account code is entered; see *Account Codes*.
- Direct to ARS conflicts with the maintenance test-line function.
- The Contact Monitor and Manual Line features cannot access the Direct To ARS feature.
- Industry-standard telephones cannot have the Forced Account Code feature enabled.
- There must be an existing ARS Leading Digit string that matches the Direct to ARS code.

Programming

Enable COS Option 217 (Direct To ARS) for the extension.

To program an ARS, use the leading digits string as the feature access code.

Assign an access code to Feature 37.

Operation

For industry-standard telephones: lift the handset and the PABX automatically dials the ARS access code.

For other devices: enter a valid account code and the PABX automatically dials the ARS access code.

Direct Trunk Select

Description

This feature allows the user to directly access an outside trunk for both incoming and outgoing calls without the need of trunk access codes. The trunk is assigned to a line appearance of the telephone through system programming. Telephones having the Direct Trunk Select feature can be programmed for ring, delayed ring, or no ring.

Direct trunk select calls bypass the system's Automatic Route Selection feature and are therefore unaffected by COR (Toll Control). An account code can be entered while established in a call.

Conditions

The following conditions apply to this feature:

- Direction and Ring variants can be programmed independently for each appearance. The Direction variant allows control of the type of trunk call, incoming only or both directions (incoming and outgoing).
- If a DTS trunk has answer points (DAY, NIGHT 1 or NIGHT 2) programmed in Form 14, they will be ignored. Instead, incoming calls on the DTS trunk will be routed to the telephone(s) with the DTS appearance.
- The trunk ignores Do Not Disturb on the *SUPERSET* telephone when it originates if there is only one appearance of the line.
- A DTS trunk must be in a trunk group in order to make outgoing calls.
- It can appear on several *SUPERSET* telephones.
- This feature is not available on *SUPERSET 401+* telephones.
- There can be only one appearance of any one Direct Trunk Select line on a *SUPERSET* telephone.
- The calls on the trunks bypass ARS but do use the SMDR feature if enabled.
- The user can transfer calls on this line to other extensions.
- When the far end hangs up on a direct trunk select line, all parties connected are cleared down, and the trunk occupies the line while it clears down.
- When the trunk hangs up, all parties in the call on the line are put idle.
- DTS trunks are unavailable to ARS, regardless of ARS programming.
- Only CO type trunks can be programmed in a direct trunk select configuration. Incoming-only trunks are not compatible.
- Incoming direct trunk select calls follow call forwarding programmed on the set only if COS Option 815 "DTS Key Honors Forwarding" is enabled for the trunk and the DTS key appears on only one set.
- Do Not Disturb is overridden by an incoming direct trunk select call.
- DTS trunks are not available on key system telephones.

Programming

Program trunks that are to be selected directly, instead of through ARS, in the *SUPERSET* Telephone Lines subform of CDE Form 09 (Station/*SUPERSET* Telephones). These trunks must be programmed into trunk group(s).

Operation

To access Direct Trunk Select:

- Lift the handset. PABX dial tone is heard.
- Press a Direct Trunk Select line key. Central Office dial tone is returned.
- Dial the external number.

To answer an incoming Direct Trunk Select call:

- Lift the handset and press the Direct Trunk Select line key when the set rings and/or the line appearance flashes.

Disconnect Alarm

Description This feature provides an alarm indication when a *SUPERSET* telephone is unplugged.

Conditions The alarm is not cleared by the *SUPERSET* telephone being plugged back in.

Programming Enable COS Option 603 (*SUPERSET* Disconnect Alarm) in the *SUPERSET* telephone's COS.

Operation Unplug the *SUPERSET* telephone - an alarm is generated; see *Attendant Alarm Readout* for reading alarms.

Display Keys

Description This feature allows users of *SUPERSET 420* and *SUPERSET 430* telephones to display the function of certain keys on their sets.

Conditions None.

Programming Refer to *Feature Keys* for instructions on how to program feature keys on *SUPERSET 420* and *SUPERSET 430* telephone sets.

Refer to *Speedcall* for instructions to program personal speedcall numbers into personal keys on *SUPERSET 420* and *SUPERSET 430* telephones.

Operation Operation varies depending upon the type of set as described below.

***SUPERSET 420* Telephones**

You can display the line numbers, speedcall numbers, or features that are programmed into the personal keys on your set.

- Press SUPERKEY.
- Press a personal key. The line number, speedcall number or feature that is programmed for the key is displayed.

  If the key is not programmed for, the display shows UNUSED KEY. If the line key is programmed with a private speedcall number, PRIVATE NUMBER appears.

  For feature keys and speedcall keys, the CHANGE and CLEAR softkeys also appear. Press the CHANGE softkey if you want to reprogram the personal key. Press CLEAR if you want to clear the feature or speed call number from the personal key.
- Press SUPERKEY to exit.

On a *SUPERSET 420* telephone, you can also display the functions of some of the hardkeys on the set:

- Press SUPERKEY.
- Press the desired hardkey. The function of the hardkey appears in the display.

  If you press the REDIAL hardkey, the number that is currently stored against it appears in the display. If no number is stored in the REDIAL key, NO NUMBER STORED appears.

  If you press the MESSAGE key, and you have don't have any messages waiting, NO MESSAGES appears in the display.
- Press SUPERKEY to exit.

***SUPERSET 430* Telephones:**

You can display the line numbers, speedcall numbers, or features that are programmed into the line keys/personal keys on your set.

- Press SUPERKEY.
- Press the MORE softkey until the DISPLAY KEYS softkeys appears.
- Press the DISPLAY KEYS softkey.
- Press a line (personal) key. The line number, speedcall number or feature that is programmed for the key is displayed.

  If nothing is programmed for the key, the display shows NO NUMBER SAVED. If the key is programmed with a private speedcall number, PRIVATE NUMBER appears.
- Press SUPERKEY to exit.

Do Not Disturb

Description

Do Not Disturb prevents incoming calls from ringing the telephone. It also prevents incoming directed and group paging announcements from occurring over the set speaker. Outgoing calls are unaffected.

Callers to a telephone with Do Not Disturb active receive reorder tone; the message DO NOT DISTURB appears on display sets. The alternative is to program the system to reroute such calls to a predetermined answering point; see *Call Rerouting.*

Consoles and *SUPERSET 430* telephones can override Do Not Disturb.

Conditions

The following conditions apply to this feature:

- If Do Not Disturb is activated, users hear broken dial tone followed by normal dial tone when they go off-hook.
- Either the set user or the attendant can set up or cancel Do Not Disturb.

- The console or a *SUPERSET 430* telephone can override Do Not Disturb if COS Option 500 (Override) is enabled in its COS, and COS Option 238 (Override Security) is disabled in the called extension's COS.
- When overriding Do Not Disturb, the Auto-Answer feature is ignored.
- When overriding Do Not Disturb, if the called set becomes busy in the meantime, then the overriding console or *SUPERSET 430* telephone will attempt to intrude on the call; see *Override (Intrude)*.
- A hunt group member that activates Do Not Disturb is treated as busy during hunting.
- While Do Not Disturb is active, forwarding works as if the extension were busy.
- If the set user activates a callback, the callback is honored, even if Do Not Disturb is active.
- When a Reminder expires, Do Not Disturb is ignored.
- Call Rerouting ignores Do Not Disturb.
- If Discriminating Dial Tone is enabled then discriminating dial tone is heard at an extension when it has Do Not Disturb activated; see *Discriminating Dial Tone.*
- DID trunks ignore DND if there is no Do Not Disturb routing point for the extension's tenant.
- If a call is placed on hold by the telephone with Do Not Disturb set, is not retrieved, and recalls when the hold times out, the telephone with Do Not Disturb set does not ring and does not see the recalling call.
- If there are multiple call line appearances of the Do Not Disturb station or *SUPERSET* telephone then the behavior of the feature changes. If the extension is dialed, busy tone is heard if there are no free appearances to ring and all *SUPERSET* telephones where the line appears have Do Not Disturb activated. If all *SUPERSET* telephones where the line appears have Do Not Disturb activated, then the caller gets Do Not Disturb Reorder Tone. If at least one appearance is free then the call to the extension proceeds as if the Do Not Disturb was not activated.
- Consoles are never directed to the Do Not Disturb reroute point.
- Do Not Disturb settings are not maintained through a system reset.

Programming

To enable the Do Not Disturb feature, assign an access code to Feature 10 (Do Not Disturb) in Form 02. To permit a set user to activate Do Not Disturb, enable COS Option 220 (Do Not Disturb) in the set's COS.

For feature key activation of Do Not Disturb at a *SUPERSET 410, SUPERSET 420 or SUPERSET 430* telephone, program a DO NOT DISTURB feature key. (See *Feature Keys.*)

To intercept calls dialed to a Do Not Disturb extension, program CDE Form19 (Call Rerouting Table) for the extension's tenant.

Operation

Operation varies depending upon the type of device as described below.

Industry-standard and *SUPERSET 401+* Telephones:

To activate Do Not Disturb:

- Dial the Do Not Disturb access code followed by the digit 1.
- Dial tone is heard.
- Hang up.

To remove Do Not Disturb from the extension:

- Dial the Do Not Disturb access code followed by the digit 2.
- Dial tone is returned.
- Hang up.

***SUPERSET 410, SUPERSET 420,* and *SUPERSET 430* Telephones:**

To activate Do Not Disturb:

- Press the DO NOT DISTURB feature key. The adjacent LCD indicator darkens.

To remove Do Not Disturb from the extension:

- Press the DO NOT DISTURB feature key. The adjacent LCD indicator, which was dark, clears.

DTMF-To-Rotary Dial Conversion

Description

This feature automatically converts DTMF tones from DTMF equipment to rotary dial outpulsing on outgoing trunks which have been programmed as rotary dial trunks. The DTMF digits also appear on the trunk, as early line split is not provided.

Conditions

None.

Programming

Specify the trunk as non-DTMF in CDE Form 13 (Trunk Circuit Descriptors).

Operation

None.

Expensive Route Warning

Description

A trunk route can be programmed to give an expensive route warning tone (three short tones). On *SUPERSET 420* and *SUPERSET 430* telephones, and on the console, the LCD displays a message. The user can continue with the call or hang up and try again later when a less expensive route may be available. *SUPERSET 420* and *SUPERSET 430* telephones provide the additional options of camping on to wait for a less expensive route or placing a callback on a less expensive route.

Conditions

The following conditions apply to this feature:

- If Callback is activated, the callback is to the least cost route.
- If the caller waits (camps on), the caller waits for the least cost route.
- Campon and callback to an expensive route are available to *SUPERSET 420* and *SUPERSET 430* telephones. These options are also available to *SUPERSET 410* and *SUPERSET 430* telephones that have CAMPON and CALLBACK feature keys programmed.
- Warning tone is given to all devices (including incoming tandem trunks).

Programming

Program Warning Tone (WT) in CDE Form 24 (ARS Route Lists). Refer to the *Automatic Route Selection and Toll Control Practice*, for designating expensive routes.

Enable COS Option 301 (Campon) and COS Option 237 (Outgoing Trunk Campon) for the extension to allow campon for less expensive routes.

Enable COS Option 300 (Automatic Callback) and COS Option 236 (Outgoing Trunk Callback) in the COS for the extension to allow callbacks to less expensive routes.

Operation

A trunk or extension attempts an external call and ARS cannot find a route without the expensive route designation. The warning tone is provided; if no action is taken by the caller, the call proceeds after a 5 second delay.

For *SUPERSET 420* and *SUPERSET 430* telephones, see Callback - Busy for the operation of the CALLBACK softkey and see Campon for the operation of the softkeys CAMPON, WAIT and I WILL WAIT.

Feature Keys

Description

The programmable line keys on *SUPERSET 410*, *SUPERSET 420*, and *SUPERSET 430* telephones and Programmable Key Modules that are commonly used for speedcall and line appearances, may also be used for feature activation; the user simply presses a feature key.

The feature keys available on *SUPERSET 410* telephones include:

- Speedcall
- *Call Forward
- Account Code
- *Do Not Disturb
- *Auto Answer
- *Music
- Direct Page (key system telephones only)
- PA Paging
- Call Pickup
- Campon (I Will Wait)
- Callback

- Swap (Trade Calls)
- Privacy Release
- Override (Intrude)
- Night Answer
- Forward Call
- ** Single Flash
- ** Double Flash
- Release

For the feature keys above marked with an asterisk (*), an indication is given on the adjacent LCD display when the feature is active. For the feature keys above marked with two asterisks (**), the adjacent LCD indicators are inoperative. For the remaining feature keys listed, an indication on the adjacent LCD indicator indicates when the feature becomes available to the user. The LCD indicator for Direct Page is not used.

Fewer Feature Keys are available on the *SUPERSET 420* and *SUPERSET 430* telephones because most features are provided via softkeys.

The Feature Keys available on a *SUPERSET 420* telephone are:

- Speedcall
- Call Forward
- Account Code
- Do Not Disturb
- Auto Answer
- Music
- Direct Page (Key Set telephones only)
- PA Pager Access
- Privacy Release
- Intrude (Override)
- Night Answer
- ** Single Flash
- ** Double Flash
- Release

For the feature keys listed above and marked with two asterisks (**), the adjacent LCD indicators are inoperative.

The Feature Keys available on *SUPERSET 430* telephones are:

- Speedcall
- Do Not Disturb
- Auto Answer
- Direct Page (Key Set telephones only)
- ** Single Flash
- ** Double Flash

For each of the keys listed above and not marked with two asterisks, an indication on the adjacent LCD indicator is provided when the feature becomes active. For the feature keys listed above and marked with two asterisks (**), the adjacent LCD indicators are inoperative.

Conditions

The following conditions apply to this feature:

- The keys are effective when the feature is applicable. For example, the Override feature key will not be effective if there are no calls in progress.
- The Single Flash and Double Flash feature keys operate only if COS 257 "Flash Over Trunk" is enabled in the set's Class of Service Options Assignment form and "Flash over trunk" is enabled in the trunk's circuit descriptor. These keys apply only to two-way calls between trunks in trunk groups and keysets that do not have any other calls on hold.
- The Auto-Answer, Call Forward, and Do Not Disturb feature keys are effective at all times.
- The feature keys do not preclude the use of feature access codes.
- Line/personal keys on *SUPERSET 410* telephones that are not assigned to a line default to being Speed Call keys. However, if you program a personal key as another type of feature key, for example as a DO NOT DISTURB key, you can't program a speed call number into the key until you reprogram it as a PERSONAL SPEED CALL key.

Programming

Assign an access code to Feature Access Code 47 (Program Feature Key) in CDE Form 02.

Feature keys may be programmed in 2 ways:

1. **From CDE:**
   - Enter the Expand Set Subform of CDE Form 09 (Station/*SUPERSET* Telephones) or CDE Form 45 (Key System Telephones).
   - In the TYPE field, select feature key via the FEATURE softkey.
   - Press the softkey corresponding to the desired feature.
2. **From the *SUPERSET* Telephone.**

***SUPERSET 410* telephones:**

- Dial the Program Feature Key access code.
- Press an unused personal key.
- Dial the feature code for the desired feature key (see following table).
- Press the SPEAKER key.

***SUPERSET 420* telephones:**

- Press SUPERKEY.
- Press the NO softkey until PERSONAL KEYS? appears in the display.
- Press the YES softkey.
- Press a personal key that isn't assigned to a line.
- Press the CHANGE softkey. SPEED CALL? appears.
- Press the YES softkey to program the key as a speed call key
  or
  Press the NO softkey. Another feature key option is displayed.
- Press the NO softkey until the desired feature key option is displayed.

- When the desired option is displayed, press the YES softkey to program the personal key.
- Press SUPERKEY to return to the date and time display.

To determine what feature, if any, is programmed for a personal key:

- Press SUPERKEY.
- Press the personal key. The function of the key is displayed.
- Press SUPERKEY to exit.

***SUPERSET 430* telephones:**

- Press SUPERKEY.
- Press the MORE softkey.
- Press the FEATURE KEY softkey.
- Press the desired line key.
- Press the CHANGE softkey.
- Press the softkey of the desired feature.
- Press SUPERKEY.

Feature Code	Feature Key
00	Speedcall*
01	Call Forward
02	Account Code
03	Do Not Disturb
04	Auto Answer
05	Music
06	Direct Page
07	PA Paging
08	Pickup
09	Campon (I Will Wait)
10	Callback
11	Swap (Trade Calls)
12	Privacy Release
13	Override (Intrude)
14	Night Answer
15	Forward Call
18	Release
19	Single Flash
20	Double Flash

* After dialing 00, dial the number to be stored.

Operation

Rather than dialing the feature access code for the desired feature, simply press the feature key. Refer to the specific feature for further details.

Flash - Calibrated

Description

Telephone users access many PABX features using a switch hook flash from industry-standard telephones. Calibrated Flash allows the system to consistently create the proper flash time thus preventing confusion between flash and hang up attempts. On rotary dial sets, the user sends a calibrated flash by dialing the digit "1". On DTMF sets equipped with a flash key, the user presses this key to send a switch hook flash to the PABX.

Conditions

The following conditions apply to this feature:

- DTMF sets equipped with a flash button must have a calibrated flash time of 50 to 140 ms.
- Rotary dial users can perform a calibrated flash, except when dialing a number. (The system interprets the digit 1 as part of the dialed number.)
- For ONS and OPS line cards, flash timers have no effect on the calibrated flash detection circuits.

Programming

In CDE Form 04 (System Options/System Timers):

- Enable System Option 37 (Calibrated Flash).
- Set System Option 52 (Minimum Flash Timer) to 200 ms if a Calibrated Flash line card is used.
- Enable System Option 38 (Switch-hook Flash).

In the CDE Form 03 (COS Define):

- Disable COS Option 223 (Flash Disable) to allow a flash to be processed.

Operation

In an established call:

- From a rotary set, dial the digit 1 and proceed as appropriate for the feature.
- From a DTMF set, press the calibrated flash button and proceed as appropriate for the feature.

Flash Control

Description

This set of options limits the use of consultation hold (switch–hook flash) under certain conditions when an extension is in a call with a trunk or attempts to establish a call with a trunk.

Flash On Incoming Trunk

This feature allows extension users to place an incoming trunk on consultation hold. This enables the trunk call to be transferred, held, or added to a conference. The option does not apply when the extension is talking to a DISA trunk that has dialed into the system.

Flash on Outgoing Trunk

This option is the same as the previous option but it applies to outgoing trunks.

Cannot Dial a Trunk After Flashing

This option prohibits the extension user from accessing a trunk, through dialing or picking up a trunk on hold at another extension, while a consultation hold is in progress. The option does not apply to industry-standard telephones with COS Option 203 (Broker's Call) or COS Option 252 (Broker's Call With Transfer) in their COS or when picking up trunk calls that are ringing at another extension.

Cannot Dial a Trunk If Holding or in Conference with a Trunk

This option prevents devices from dialing a trunk call or picking up a trunk from another extension while another trunk is in a call (conference or two party) on consultation hold. This option does not apply to industry-standard telephones with COS Option 203 (Broker's Call) or COS Option 252 (Broker's Call With Transfer) in their COS.

Conditions

The following conditions apply to this feature:

- The options are disabled if the extension has COS Option 223 (Flash Disable) enabled in its COS.
- The options are disabled for industry-standard telephones if System Option 38 (Switch–hook Flash) is disabled.

Programming

Enable any or all of the following COS Options for the extension:

212 (Can Flash if Talking To An Incoming Trunk)
213 (Can Flash if Talking To An Outgoing Trunk)
214 (Cannot Dial a Trunk After Flashing)
215 (Cannot Dial a Trunk if Holding or in Conference with a Trunk)

Operation

None.

Flash Disable

Description An extension may be inhibited from using all services requiring the use of the switch hook flash. For *SUPERSET* telephones, this prevents the extension from putting a call on consultation hold.

Conditions The following conditions apply to this feature:

- COS Options 223 (Flash Disable) is mutually exclusive with:

  COS Option 224 (Flash for Attendant)
  COS Option 203 (Broker's Call)
  COS Option 252 (Broker's Call With Transfer)
  COS Option 302 (Flash-In Conference).

Programming Enable COS Option 223 (Flash Disable) for the extension.

Enable System Option 38 (Switch–hook Flash) to allow flashing for all industry-standard telephones in the system.

Operation None.

Flash For Dial 0 (Attendant)

Description An extension can be set to ring the Dial 0 Routing Point (usually the Attendant) automatically if a transfer is attempted while in an established call.

Conditions The following conditions apply to this feature:

- The extension cannot access any other feature requiring a switch hook flash; such as Broker's Call, or Transfer/Conference, or Call Hold.
- The conditions for consultation hold must be met before the call can proceed; see *Transfer*.
- The extension calls the DIAL 0 or PRIORITY DIAL 0 routing, based on the extension's tenant, COS, and the current NIGHT/DAY service. If there is no point specified for the given conditions then no call is made and reorder tone is supplied.
- This only applies if there is not a call on Consultation Hold.

Programming Enable COS Option 224 (Flash for Attendant) for the extension.

Operation While on an established call, flash the switch–hook or press the **Transfer** softkey. The extension rings the Dial 0 (Attendant) Routing Point; the other party is placed on consultation hold.

Flash For Waiting Call

Description

COS Option 205 (Flash for Waiting Call) allows a user to place a call on consultation hold and connect to a waiting call by performing a switchhook flash.

If COS Option 252 (Broker's Call with Transfer) is also enabled, a user can toggle between the two calls using the switchhook. The user is unable to form conferences if COS Option 252 is enabled.

See *Campon Warning Tone* to allow the telephone to receive an audible notification of waiting calls.

Conditions

The following conditions apply to this feature:

- This feature is for standard telephones. It is not intended for use on *SUPERSET* telephones.
- Other features that are also activated with a switchhook flash affect the operation of this feature.

  For example, if COS Option 302 (Flash in Conference) is also enabled, when the station user performs a flash to answer a camped on call, the call is conferenced in with the current call. The user can add each new call that camps on to the conference by performing another flash.
- This feature takes precedence over any other feature which uses the switchhook flash.
- The feature *Flash Disable* will not prevent this feature from functioning.
- When this feature is enabled, the features *Flash For Attendant*, *Broker's Call*, and *Broker's Call With Transfer* will operate normally, provided that there are no calls waiting.
- The use of this feature in conjunction with other features which use the switchhook flash may be complicated. The system will provide an indication that there is a call waiting (campon warning tone); however, the system does not provide an indication when the call is no longer waiting.
- The first waiting caller is connected; see *Campon* for details on the ordering of waiting calls.

Programming

The following steps are required:

- Enable System Option 38 - Switch-hook Flash.
- Enable COS Option 205 - Flash For Waiting Call in the telephone's COS.
- COS Option 252 - Broker's Call with Transfer in the telephone's COS (recommended).
- Enable campon for calling devices - see programming under *Campon.*

Operation

Upon hearing campon warning tone, flash the switchhook. The current call is placed on consultation hold, and the waiting call is connected.

If COS Option 252 - Broker's Call with Transfer is also enabled:

While on a call you hear campon tone.

- Flash the switchhook.

  Your current call is placed on consultation hold and you are connected to the camped on call.

- Flash the switchhook to toggle between the two calls.

To transfer one of the parties:

- Inform the party that you will perform a transfer.
- Flash the switchhook to connect to the other party and ask this party to hang up.

  After this party has hung up, you hear dial tone.

- Dial the number of the desired station and hang up.

  The call is transferred.

Flash Timing

Description

The flash timer is a systemwide programmable item; its value applies to all industry-standard telephones in the system.

Minimum Flash Timer (10 ms units) 20 - 50 units 200 to 500 ms
Maximum Flash Timer (10 ms units) 20 - 150 units 200 to 1500 ms

Conditions

None.

Programming

Set the desired times in Form 04 (System Options/System Timers) - System Options 52 - Minimum Flash Timer and 53 - Maximum Flash Timer.

Operation

None.

Forward Campon

Description

Calls that camp on to a *SUPERSET 410*, *SUPERSET 420* or *SUPERSET 430* telephone can be selectively forwarded to the telephone's call forwarding destination. When a party camps on to a busy telephone that has call forwarding programmed (it may be active or inactive), the person at the busy telephone can press a softkey or the FORWARD CALL feature key to forward that waiting party to the call forwarding destination.

See *Call Forwarding* for selecting a call forwarding destination. See *Campon* and *Swap Campon* also.

Conditions

The following conditions apply to this feature:

- The softkey on the *SUPERSET 420* telephones is only displayed for the first 10 seconds after a new caller camps on. After that time the softkey is removed.
- The feature key on *SUPERSET 410* telephones is only displayed for the first 10 seconds after a new caller camps on. After that time the feature key is removed.
- The feature forwards the first waiting caller. See *Campon* for information on the ordering of waiting callers.
- The feature is only available if the forwarding destination is idle; see *Call Forwarding.*
- The feature is not available if the forwarding destination is a Speedcall key or system abbreviated dial number, unless the destination translates into an ONS voice mail station or hunt group number.

Programming

Program a forwarding destination for the *SUPERSET* telephone.

For *SUPERSET 410* telephones, program a FORWARD CALL feature key.

Operation

Operation varies depending upon the type of device as described below.

***SUPERSET 410* Telephones:**

- A new caller camps on to the telephone while you are talking to another party.
- Press the FORWARD CALL feature key. The waiting caller is forwarded to the forward destination.

***SUPERSET 420* Telephones:**

- A new caller camps on to the telephone while the telephone is talking to another party.
- Press the FORWARD softkey. The waiting caller is forwarded to the forward destination.

***SUPERSET 430* Telephones:**

- A new caller camps on to the telephone while you are talking to another party.
- Press the CALL WAITING softkey.
- Press the FWD WAITING softkey. The waiting caller is forwarded to the forward destination.

Handsfree Operation

Description This feature enables users of *SUPERSET 410, SUPERSET 420* and *SUPERSET 430* telephones to have a telephone conversation without lifting the handset.

Conditions The following conditions apply to this feature:

- The Microphone key can be used to turn off the microphone on *SUPERSET* telephones. Refer to the *Peripherals Devices Practice.*
- Calls can be changed from handsfree to the handset by lifting the handset from the set and pressing the SPEAKER key. The call can be changed back to handsfree by pressing the SPEAKER key and then putting the handset back into the handset cradle.
- Handsfree cannot be used while the *SUPERSET* telephone is using the pager.
- Calls can be ended by using the SPEAKER or CANCEL key or the hangup prompt.
- COS Option 604 (*PBX SUPERSET* Telephone - Automatic Outgoing Line) affects handsfree dialing; see *Line Selection.*
- Key Set Line Preference affects handsfree dialing.

Programming None.

Operation To initiate handsfree dialing:

- Press a dial pad key, a SPEEDCALL key, a line select key, or the SPEAKER key; the system selects a line to dial on.

Handset Receiver Volume Control

Description *SUPERSET 401+, SUPERSET 410, SUPERSET 420* and *SUPERSET 430* telephone users can adjust the volume of the set's handset receiver. The handset receiver volume is independent of the set speaker (see Speaker Volume Control).

Conditions None.

Programming None.

Operation To adjust the handset receiver volume:

- During a conversation using the handset, press the VOL and ∧ VOL ∨ keys repeatedly to increase or decrease the handset receiver volume.

Headset Operation

Description *SUPERSET* telephones can be equipped with headsets instead of handsets. The headset user answers incoming calls by pressing the line select key or the SPEAKER ON/OFF key and hangs up by pressing the CANCEL key or HANGUP softkey. The user can combine headset operation with the auto-answer feature for complete handsfree operation.

Conditions The following conditions apply to this feature:

- Sets with Headset Operation enabled must be operated only with headsets.
- *SUPERSET* telephone compatible headsets must be supplied by the user. Headsets used with the 400 series *SUPERSET* telephones must be externally powered.
- The handset must be out of its cradle for the headset to operate.

Programming Enable COS Option 612 (*SUPERSET* Telephone - Headset Operation) in the set's COS.

Operation A call is ringing at the set:

- Press the appropriate line select key, or,
- Press the SPEAKER ON/OFF key, or,
- With Auto-Answer, the call is answered automatically.

To terminate the call:

- Press the SPEAKER ON/OFF key or the HANGUP softkey.

Hold

Description

This feature enables the telephone user to place the current call on hold, then replace the handset or use the telephone for other calls. While the call is held, the user can select all features normally available on the telephone. The held call can be retrieved at the telephone that placed it on hold or at another telephone.

Do not confuse the call hold feature described here with the temporary consultation hold that occurs during a call transfer; see *Transfer*.

See *Add Held* And *Auto-hold* for set hold features.

See *Attendant Hold Positions* for attendant hold features.

see *Subattendant Hold Positions* for subattendant hold features.

Conditions

The following conditions apply to this feature:

- When the hold is done, the *SUPERSET* telephone user enters the select line feature to choose another line; see *Line Selection.*
- System music is heard if it is available and continues even if the held party starts recalling to the *SUPERSET* telephone. If music is not programmed, silence is provided until the hold timeout period, when ringback is provided.
- All *SUPERSET* telephones in a conference call can put the conference call on hold.
- The HOLD button is only available when talking to another party or when in a conference call. A hold can be done implicitly at other times through AUTO-HOLD.
- For industry-standard telephones, COS Option 211 (Call Hold and Retrieve Access) must be enabled in the set's COS.
- At industry-standard telephones, the Call Hold feature does not operate if COS Option 223 (Flash Disable) or COS Option 224 (Flash For Attendant) is enabled in the set's COS.
- An industry-standard telephone can have only one extension or trunk on hold at a time. It cannot put a conference call on hold.
- When an industry-standard telephone puts a call on hold it is free to make and receive other calls.
- When the user of an industry-standard telephone attempts to place a second call on hold, the call currently on hold is swapped with the call being placed on hold (i.e., when the second call is placed on hold, the user is reconnected to the first call that was placed on hold).
- When an industry-standard telephone puts a call on hold, the call is not occupying the station's key line if it has one programmed. A *SUPERSET* telephone with an appearance of the line can put a call on hold on the industry-standard telephone's key line as well.
- The Call Hold access code does not apply to *SUPERSET* telephones that have a HOLD key.

- If a call is on hold at a *SUPERSET* telephone's prime line, other calls can be made only if the set has other line appearances programmed.
- There is no time limit on holding conference calls on a line.
- There is no time limit on holding calls on a personal outgoing line.

Hold Retrieve

- Calls can be retrieved remotely when calls are held on industry-standard telephone key lines or *SUPERSET* telephone prime line appearances (key line appearances only).
- Calls cannot be retrieved from logical lines, direct trunk select, private trunk and private outgoing lines or from multicall line appearances.
- A call held on a key line appearance of a *SUPERSET* telephone can be retrieved by pressing the line key of the held call. Calls held on key line appearances on a *SUPERSET* telephone can also be retrieved from an industry-standard telephone.
- A held call cannot be retrieved if the retrieving party has a consultation hold in progress and the held party is an industry-standard telephone.
- A held call cannot be retrieved if the retrieving party has a consultation hold in progress and the held party is a *SUPERSET* telephone with COS Option 233 (Never a Consultee) enabled.
- A held call cannot be retrieved if COS Option 215 (Cannot Dial A Trunk If Holding A Conference With A Trunk) is enabled and the retrieving extension has a trunk in the call on consultation hold.
- When a two party call on a key, private trunk line or direct trunk select line is put on hold, other *SUPERSET* telephones where the line appears can select the held line to connect to the held party.
- Conference calls on hold cannot be retrieved from other sets. They can be taken from hold through other appearances of the line only if the line was put on hold at those appearances as well.
- When a call is retrieved from an industry-standard telephone, the first check is for a call held by the telephone itself. If no call is being held, and if the industry-standard telephone has a key line appearance, then the line is checked for a held call.
- The activity of the holding set does not affect the ability of other devices to retrieve held calls.
- Industry-standard telephones, *SUPERSET* telephones, and TIE trunks can retrieve remotely-held calls.

Hold Timeout

- When the held party starts ringing the holding set, the ringing is done as if the held party had just called the holding line. That is, all appearances of the line ring, with delay ring etc. operational.
- If Auto-Answer is activated on the holding *SUPERSET* telephone then it is ignored when the held party rings the set.
- If call forward - don't answer is activated on the holding set, the forwarding is done unless COS Option 222 (Call Forwarding Inhibit on Hold Timeout) is in the holding set's COS. Other types of call forwarding are ignored.

- Once the held party starts ringing the holding set, the Call Hold Retrieve feature access code no longer picks up the held party because it is ringing the holding party. The Directed Call Pickup feature can then be used.
- Recall no answer is operational when the held party rings the holding set.
- If a party held by an industry-standard telephone attempts to ring the holding telephone, while the holding telephone is busy, then a recall is performed. See *Recall.*
- The Do Not Disturb feature is ignored when a held party rings an industry-standard telephone.
- The party is held on the line for the time specified in COS Option 254, Call Hold Recall Timer in the holding extension's COS. After that time, the held party starts ringing the line it is held on. For PBX sets only. At key sets, the party does not ring the line back at all, only a periodic beep is given. See *Key Set Hold Reminder.*

A call cannot be put on hold:

- if an attendant is in the call
- if there is a party in the call with the non-busy extension feature enabled
- if a party in the call has the call on consultation hold
- if there is an override in progress and the set is not the party being overridden
- if the set is a *SUPERSET 430* telephone using the call waiting feature
- if there is a direct trunk select trunk in the call
- if the call is on consultation hold
- if the trunk in a two party call is still dialing digits out.

Programming

Assign an access code to Feature 21 (Call Hold) for industry-standard telephones to put a call on hold.

Assign an access code to Feature 22 (Call Hold Retrieve - Local) to allow industry-standard telephone users to retrieve calls on hold at their sets.

Assign an access code to Feature 23 (Call Hold Retrieve - Remote) to allow devices to retrieve calls at other extensions.

Enable COS Option 211 (Call Hold and Retrieve Access) for the station set to put a call on hold, and for all devices to retrieve a call.

Set COS Option 254, Call Hold Recall Timer in the holding extension's COS to the desired Call Hold recall time (1 to 10 minutes).

Enable COS Option 222 (Call Forwarding Inhibit On Hold Timeout) in the extension's COS, if forwarding is not desired on a hold time-out.

Operation

Operation varies depending upon the type of device as described below.

Industry-Standard Telephones:

To place a call on hold:

- Flash the switch–hook.
  Transfer dial tone is returned.
- Dial the Call Hold code.
- Dial tone is returned.

The caller is held and hears music, if provided. The extension may make or receive calls or access features in the normal manner.

To retrieve the call locally:

- Obtain dial tone.
- Dial the Call Hold Retrieve - Local access code.
- The set is reconnected to the held call.

To retrieve the call remotely (from another extension):

- Obtain dial tone.
- Dial the Call Hold Retrieve - Remote access code.
- Dial the number of the extension holding the call.
- The call is connected to the remote extension.

SUPERSET 401+ **Telephones:**

To place a call on hold:

- Inform the caller, then press the red HOLD key.
- Dial tone is returned.

The caller is held and hears music, if provided. The extension may make or receive calls or access features in the normal manner.

To retrieve the call locally:

- Press the red HOLD key.
- The set is reconnected to the held call.

To retrieve the call remotely (from another extension):

- Dial the Call Hold Retrieve - Remote access code.
- Dial the number of the extension holding the call.
- The call is connected to the remote extension.

SUPERSET 410, SUPERSET 420, **and** ***SUPERSET 430*** **Telephones:**

To place a call on hold:

- Inform the caller, then press the red HOLD key. The call is held and the caller hears music, if provided.
- The line status display associated with the call on hold flashes as a reminder.

- The set may select another line to make calls or to access features in the normal manner, or hang up.

To retrieve the call at a *SUPERSET* telephone that has an appearance of the held line:

- Press the line select key associated with the call on hold. The call is connected to the remote set.

The set user can add a call on hold to another line to form a conference or to move an established call from one line to another; see *Add Held.*

To retrieve the call at a *SUPERSET* telephone that does not have an appearance of the held line:

- Obtain dial tone.
- Dial the Call Hold Retrieve - Remote access code.
- Dial the number of the extension holding the call.
- The call is connected to the *SUPERSET* telephone.

Holiday Messages

Description *SUPERSET* telephones can display a holiday message at Christmas and New Year's.

Every minute, the holiday messages alternate with the usual time and date message that appear on the *SUPERSET* telephone display.

Conditions The Christmas message is displayed from December 23 to December 27 and the New Year's message from January 1 to January 4.

Programming Enable System Option 20 (Holiday Messages) to show holiday messages.

Operation The regular message and the holiday message alternate every minute.

Hot Line

Description Individual telephones can be programmed through CDE as hot lines. When a hot line telephone goes off-hook, the system automatically dials a pre-programmed number (internal or external). The number is specified as the call forwarding destination for that extension.

Conditions The following conditions apply to this feature:

- The extension must be programmed as a Manual Line.
- Manual Line and Automatic Callback are mutually exclusive.
- The extension must have access to Abbreviated Dial, and External Call Forwarding (if it is to dial an external number).
- The type of forwarding programmed has no effect on the feature.
- If the extension is called, then the forwarding feature still works as usual.
- The hot line call is a reroute to the forward number; see *Call Rerouting.*
- Valid hot line destinations include all valid call forwarding destinations.
- A DTMF receiver circuit is needed for the hot line call regardless of the type of the destination.

Programming Enable COS Option 228 (Manual Line) for the extension.

The attendant must program a "Call Forward - Follow Me" destination number for that extension; see *Attendant Call Forward Setup And Cancel.*

If programming the destination number from the extension then enable one of the call forwarding COS options in the extension's class of service.

Operation Lift handset; the extension is rerouted to the forwarding destination automatically.

Hotel / Motel (Lodging)

Description For information on this feature, refer to the *Hotel/Motel Feature Package Description Practice.*

Conditions Hotel/Motel is not compatible with the Property Management System (PMS).

Programming The following System Options apply:

- 02 - Message Lamp Test Enable
- 04 - Message Waiting and Message Register Clear Print
- 09 - Attendant Call Block
- 11 - Automatic Wakeup
- 12 - Auto Wakeup Alarm

- 13 - Auto Wakeup Print
- 14 - Auto Wakeup Music
- 23 - Message Register Count Additional Supervisions
- 24 - Message Register Audit
- 25 - Message Register Zero After Audit
- 27 - Room Status Audit
- 32 - Outgoing Call Restriction (mutually exclusive with 33)
- 33 - Room Status (mutually exclusive with 32 and 108)
- 34 - Auto Room Status Conversion / Auto Wakeup Print
- 40 - Message Register Follows Talker
- 49 - Pseudo Answer Supervision Timer
- 56 - Auto Room Status Conversion / Auto Wakeup Print Timer
- 57 - Vacant/Reserved Room Default Call Restriction
- 58 - Occupied Room Default Call Restriction
- 107 - Lodging (Hotel/Motel) (mutually exclusive with 108)
- 108 - Property Management System (mutually exclusive with 33 & 107)

The following COS Options apply:

- 101 - Attendant O/G Restriction / Room Status Setup
- 105 - Attendant Guest Room Key
- 113 - Attendant Call Block Key
- 202 - Alarm Call
- 204 - Call Block Applies
- 220 - Do Not Disturb
- 230 -Message Register Overflow Alarm
- 231 - Message Waiting Setup - Bell (mutually exclusive with 232)
- 232 - Message Waiting Setup - Lamp (mutually exclusive with 231)
- 244 - Room Status Applies
- 608 - *SUPERSET* Telephone - Room Status Display
- 610 - *SUPERSET* Telephone - Guest Room Template
- 703 - Message Register Applies

The following Feature Access Codes apply:

- 35 - Maid In Room
- 36 - *SUPERSET 4* Telephone Room Status Display
- 40 - *SUPERSET 4* Telephone Maid In Room Status Display

Refer to the *Hotel/Motel Feature Package Description Practice*, for further information.

Operation

Refer to the *Hotel/Motel Feature Package Description Practice*.

Hunt Groups

Description

Hunt groups, or master number hunting, allows a collection of devices to share a common access code. A caller can be routed to or dial the access code (the master hunt number of the hunt group), and have the call completed to an available extension in that hunt group. Extensions within a hunt group may still be accessed directly by dialing the extension number.

A hunt group is busy if all members of the hunt group are busy.

If all devices in a hunt group are busy and there is an overflow point programmed, the call is forwarded to the overflow point (if it is available). If there is no overflow point programmed or if it is busy, then the caller finds the hunt group busy. See *Campon* for information on campon to busy hunt groups.

If the caller camps on then the first extension in the hunt group list without Do Not Disturb activated receives a campon beep and Swap Campon capability for the first party camped on to the hunt group; see *Swap Campon.*

See *Trunk Groups* for details on trunk groups.

Some special types of hunt groups include:

- Recording hunt group; see *Recording Device Support.*
- UCD Agent hunt group; see *Uniform Call Distribution.*
- Automated Attendant hunt group; see *Automated Attendant.*
- Voice Mail hunt group; see *Voice Mail - ONS Port.*
- Recall Appearance Back to Originating Set (RABTOS) hunt group; see *Mitel Application Interface (MAI).*

Two types of hunting are provided by the system, circular and terminal:

- Circular hunting starts at the extension after the last extension in the hunt group to which a call was completed (the extension rung), and hunts overall extensions in the hunt group in the sequence programmed. Hunting stops at the first idle extension found.
- Terminal hunting starts at the first extension in the hunt group and terminates at the first idle extension found. Hunting takes place in the order in which the extensions were programmed into the hunt group.

Conditions

The following conditions apply to this feature:

- An extension must be programmed before programming it into a hunt group.
- A maximum of 99 hunt groups may be programmed.
- A hunt group may have a maximum of 50 members.
- The overflow is only checked once, when the call is initially made to the hunt group.
- An extension cannot be assigned to more than one hunt group.

- An extension in a hunt group is passed in the hunt if:

  it is busy
  Do Not Disturb is set
  it is busied-out
  it is locked out
  it is a *SUPERSET* telephone with a busy prime line.

- The following overflow points are allowed: hunt groups, sets, extensions, datasets, consoles, LDNs, Night Bells, and ACD paths.
- Data hunt groups can have only other data devices (or data hunt groups) for an overflow point.
- A hunt group can have only one overflow point; however, a device can be the overflow point for more than one hunt group.
- Overflow is not permitted if the overflow point has forwarding active.
- Do not delete programming for a hunt group overflow point. A hunt group should always have a valid overflow point.
- When a hunt group overflows to another hunt group, overflow to the second hunt group's overflow point is not permitted.
- All tenant and COS checks involving a hunt group use the first programmed member of the hunt group.
- Recall never involves hunt groups (unless a reroute through CALL REROUTING) - recall can only occur to the individual hunt group members.
- If there is no hold timeout point and recordings are given, then for loop start trunks that do not provide release supervision, the system does not disconnect the trunk. It is important to provide a routing point for these unanswered calls to ensure that the loop start trunk is either answered, or timed out for no answer and then released.

Programming

Program all extensions via CDE Form 09 (Station/*SUPERSET* Telephones). Program datasets via CDE Form 12 (Data Assignment).

Program the hunt group via CDE Form 17 (Hunt Groups).

Enter the desired extension numbers of sets, datasets.
Enter the hunt group access code.
Enter the appropriate options for the hunt group.
Assign the appropriate hunt group type.

Operation

None.

Illegal Access Intercept

Description

Calls to restricted access codes or extension numbers can be routed to a given answering point for completion. This point can be an LDN position on the attendant console (see *Console LDN Keys*) or any valid reroute point. Illegal number intercept points can be programmed to be different or the same for DAY, NIGHT1, and NIGHT2 operation.

Conditions

The following conditions apply to this feature:

- If the required programming is not done, such calls receive reorder tone.
- Only sets, DISA trunks, and CO trunks are routed to the answering point.
- See *DID/Dial-in/Tie Intercepts* for illegal number handling for DID and Tie trunks.
- If the call is routed to a console, the call is shown as an intercept call at the console.

Programming

To cause all calls to restricted numbers to be routed to a specific answering point, access CDE Form 19 (Call Rerouting Table) and enter the desired answering point access code, into the appropriate column for the "Station Illegal Number Routing For This Tenant" Call Type.

Operation

None.

Inhibit Trunk Ring-Me-Back During Dialing

Description

This feature inhibits the operation of a particular instance of the Station Transfer Security feature. If an industry-standard telephone is dialing and goes on–hook while a trunk is on consultation hold, that trunk does not ringback the station and is instead dropped. This prevents trunk lock-ups when the flash on the trunk was intended as a hang-up and the station user did not expect a trunk to be on consultation hold.

Conditions

The following conditions apply to this feature:

- COS Options 401 (Call Park) and 403 (Trunk Recall Partial Inhibit) are mutually exclusive.
- This only operates for industry-standard telephones.
- This only operates for single trunks on consultation hold. Non-trunks and conference calls ring back the industry-standard telephone.
- Serial trunks are not dropped; they recall back to the console.
- If the station has called another party that has external call forwarding enabled and hangs up during the dialing of the external number, the held trunk is transferred to the external number and is not dropped.

Programming

Enable COS Option 403 (Trunk Recall Partial Inhibit) for the extension.

Operation Establish a trunk call.

Flash. Dial tone is returned.

Hang up. Trunk is dropped.

Intercept to Recorded Announcement

Description Incoming trunk calls can be intercepted to groups of recording devices after dialing vacant numbers, reaching busy extensions, obtaining no answer, or as required.

Conditions None.

Programming See *RAD Support.*

Enter the extension of the recording device hunt group into CDE Form 19 (Call Rerouting Table) for the appropriate routing.

Operation None.

Inward Restriction (DID)

Description An extension may be restricted to not receive calls directly from DID trunk calls.

Conditions None.

Programming Enable COS Option 226 (Inward Restriction - DID) for the extension.

Operation None.

Key Set/System Support

Description

Key system functionality is supported. Combined with tenanting and other features, this allows for departmental key applications. The following key system features are provided:

- Key Sets - Auto Answer for Directed Page Calls
- Key Sets - CO Line Group Key
- Key Sets - CO Line Key
- Key Sets - CO Line - Direct Select
- Key Sets - Handsfree Answerback to a Directed Page
- Key Sets - Hold Reminder
- Key Sets - Intercom Key
- Key Sets - Line Preference
- Key Sets - Off-hook Voice Announce
- Key Sets - Paging
- Key Sets - Personal Speed Call.

Key sets are *SUPERSET 401+*, *SUPERSET 410*, *SUPERSET 420* and *SUPERSET 430* telephones that have been programmed as key system telephones. In addition, as key system telephones, these sets can be associated with PROGRAMMABLE KEY MODULES. PKMs are described in this document. Refer to the conditions of individual feature to determine what features are available on each type of key set.

Conditions

Key system features will not function on PABX telephones.

Programming

Programming for key system telephones, sub-attendants, and PKMs is done in CDE Form 45 (Key System Telephones) and CDE Form 46 (Key System Toll Control). Refer to the specific feature for programming details.

Refer to the *Customer Data Entry Practice*, for further details on programming.

Operation

Refer to the specific feature.

Key Sets - CO Line Group Key

Description

This type of key is available only on key system telephones to allow a selection of an idle CO line from a CO line group. The key accesses a group of CO lines without having a dedicated appearance for each line on the set. Toll control is handled by ARS. The LCD indicator corresponding to the key has no function.

Conditions

The following conditions apply to this feature:

- The CO line group key needs an available internal line to make the call. The CO line group key cannot be used while listening to external dial tone.
- If all the CO lines in the CO line group are busy when a user presses the CO line group key, the user hears busy tone. The user then has the option of camping on, or setting a callback (see *Campon* or *Callback-Busy*). If the user sets a callback, the callback will ring the Intercom key when a CO line becomes available.
- Redial does not operate on a CO line group key.
- CO line group keys can only be programmed on *SUPERSET 410, SUPERSET 420,* and *SUPERSET 430* telephone sets that have been programmed as key system telephones.

Programming

The following programming steps are required:

- Enter trunks in CDE Form 14 (Non-Dial-In Trunks) for the CO line group.
- Assign those trunks to CO Line Groups in CDE Form 16 (Trunk Groups).
- Enter ARS leading digits into CDE Form 26 (ARS: Digit Strings) for the CO line groups.
- Enter the CO line group numbers into CDE Form 23 (ARS: Route Definition).
- Program the remaining ARS tables (CDE Forms 20 - 27).
- Enter the CO line group key and the associated ARS leading digit(s) into the Expand Set Subform of CDE Form 45 (Key System Telephones).

Operation

To originate a call:

- If the origination line preference is programmed as a CO line group key, all the user has to do is to go off–hook to originate the call, providing there is a free internal line.
- Acquire internal dial tone (via the Intercom key, or some other means).
- Press the CO line group key.
- Dial the required digits to complete the call.

Key Sets - CO Line Key

Description

This is a key type which is available only on key system telephones. It is used to originate and answer calls to or from parties outside the system. The key accesses a specific trunk directly. A CO line key may be shared by any number of key system sets, but only one may access it at a time. One other party can join in on a call on the line if the CO line is non-private, or privacy is released. The LCD indicator corresponding to the key functions as described in the *Peripherals Devices Practice.* Dialing is checked against Key System Toll Control; if a call is not answered, it reroutes as programmed in CDE Form 19.

Conditions

The following conditions apply to this feature:

- Calls made via a CO line key are subject to key system toll control restrictions.
- Auto answer will not operate on a CO line key.
- If another key is selected while dialing on a CO line, the CO line will be cleared down. If another key is selected while there is a call on the CO line, the CO line call will be put on hold.
- When camping on to a busy CO line, there is no campon warning tone given.
- When a callback is set on a busy CO line, the call will be returned to the INTERCOM key. When the callback is answered, the call will switch back to the CO line key - both must be idle for a successful callback.
- On SMDR records, the ARS leading digit field will be blank for outgoing CO Line calls.
- The back arrow, for deleting misdialed digits, is not available when dialing on the CO line key.
- CO line keys can only be programmed on *SUPERSET 410, SUPERSET 420,* and *SUPERSET 430* telephone sets that have been programmed as key system telephones.

Programming

The following programming steps are required:

- Enter the trunk into CDE Form 14 (Non-Dial-In Trunks).
- If answer points are programmed for the trunk in Form 14, delete them. Deleting the answer points will allow calls to ring the CO line key, and allow the user to access the trunk using the CO line key.

  Note: If answer points are programmed then the calls will be directed to the answer points based on the system service mode (Day, Night 1, or Night 2). Although the calls will be indicated on the key set by the flashing line key appearance, the set user will not be able to answer the call or access the trunk.

- Enter a routing point for CO Line Routing Points on No Answer in Form 19 (Call Rerouting Table).
- Enter the trunk into a group in CDE Form 16 (Trunk Groups).

- Enter the CO line key into the Expand Set Subform of CDE Form 45 (Key System Telephones).
- Enter digit strings into CDE Form 46 (Key System Toll Control) for analysis of dialed digits on the CO line. To add to restriction, enter a suitable COR.
- Enter COR group members into CDE Form 20 (ARS: COR Group Definition) to restrict users from dialing specific digit strings.
- Specify the maximum permitted number if dialed digits for each COR group in CDE Form 27 (ARS: Maximum Dialed Digits).

Operation

To originate a call:

- Press an idle CO line key

  or
- Go off-hook (if CO line key is the preference -- see *Key Sets - Line Preference).*
- Dial outgoing digits; ARS codes are not required.

To answer a call:

- Press the ringing CO line key

  or
- Go off-hook (if it is the first ringing line).

Key Sets - CO Line - Select Direct

Description

This feature allows a key system telephone user to directly access a specific CO trunk which may or may not appear on the user's telephone set. This feature must be accessed through any internal line.

Conditions

The following conditions apply to this feature:

- Calls made in this manner are subject to key system Toll Control restrictions.
- If the CO trunk is busy, the user is not permitted to barge in under any circumstances.
- If the CO trunk is busy, the user may set up campon or callback. If callback is set up, the call is returned to the intercom line.
- It is possible to use speed call to access this feature: store the Direct CO Line access code and the three-digit trunk number.
- CO Line - Select Direct keys can only be programmed on *SUPERSET 410*, *SUPERSET 420,* and *SUPERSET 430* telephone sets that have been programmed as key system telephones.

Programming

The following programming steps are required:

- Enable COS Option 680 (Key System - Direct CO Access) in the telephone's COS.

- Assign an access code to Feature Access Code 50 (Direct CO Line Select) in Form 02 (Feature Access Codes).
- See *Key Sets - CO Line Key* for information on toll control programming.

Operation

To access a CO trunk:

- Dial the Direct CO Access access code.
- Dial the 3-digit trunk number. (Trunk numbers are 1 thru 200 - if a user is accessing trunk number one, one, two or three digits can be dialed, e.g., 1, 01 or 001.)

Note: It is possible to use speed call to access this feature: store the Direct CO Line access code and the three-digit trunk number. See *Key Sets - Personal Speed Call.*

Key Sets - Direct Access on CO Line Keys - Bypass Key System Toll Control

Description

This feature allows an extension seizing a CO trunk with a line key to bypass PBX dial tone and Key System Toll Control. Instead, dial tone from the CO is immediately received. This allows users to hear stutter CO dial tone on their CENTREX lines, indicating the presence of voice mail messages. Users may then access their voice mail or other CENTREX features (see *CENTREX Compatibility* (*Flash Over Trunk*)) or they may dial an external destination number.

Conditions

The following conditions apply to this feature:

- The trunk must be a 6-circuit CO trunk, a 6-circuit DISA trunk, a 4-circuit CLASS trunk, a T1 (LS/GS, DID/TIE, or E&M) trunk, an E&M module or a T1 DISA (TIE, E&M, or CO).
- Personal Speedcall does not work on direct CO line access because no digits are analyzed, but Speed Call, Redial, SMDR record, and other CO Line Key functionality is unaffected by this option.

Programming

Perform the following:

- Set the "Direct Access on CO Line Keys: Bypass Key System Toll Control" option to "Yes" in the "Outgoing" parameter of the Options subform of the Trunk Circuit Descriptor form (CDE Form 13) for the CO Line Key's trunk.

Operation

To originate a call:

- Press an idle CO line key.

  or
- Go off-hook (if CO line key is the preference -- see *Key Sets - Line Preference*).
- Dial outgoing digits; neither ARS codes nor Key System Toll Control digits are required.

Key Sets - Handsfree Answerback to a Directed Page

Description

This feature allows users of *SUPERSET 410, SUPERSET 420,* and *SUPERSET 430* key system telephones to respond handsfree to a directed page. In order to respond to a directed page handsfree, the user must turn on the set's microphone lamp in advance. Then, if a directed page from a key system telephone is broadcast over the users set, the set microphone is activated allowing the user to speak handsfree to the calling party.

The *Key Sets - Handsfree Answerback to a Directed Page* feature is only available on *SUPERSET 410, SUPERSET 420,* and *SUPERSET 430* key system telephones (refer to *Key Set/System Support*).

Conditions

The following conditions apply to this feature:

- This feature is only available on *SUPERSET 410, SUPERSET 420,* and *SUPERSET 430* telephones that have been programmed as key system telephones (refer to *Key Set/System Support*).
- A directed page can only be made from a *SUPERSET 401+, SUPERSET 410, SUPERSET 420,* or *SUPERSET 430* that has been programmed as a key system telephone (refer to *Key Set/System Support*).
- During a handsfree answerback call, pressing the MICROPHONE key toggles the set microphone on and off.
- The telephone can be put in handsfree answerback mode while the telephone is idle (i.e., if you answer a directed page call by pressing the MICROPHONE key, when you go back on hook the set will not be in handsfree answerback mode).
- The user can switch from handsfree answerback operation to the handset by lifting the receiver, or to the headset by pressing the SPEAKER key or the prime line key.
- During a handsfree answerback call, users have access to all the features that are available during a normal call.
- A directed page cannot be made to a set in DND.
- A directed page cannot be made to a *SUPERSET 401+* telephone.

Programming

Enable COS Option 683 (Key System - Direct Paging Handsfree Answerback) in the set's class of service. By default, this COS Option is disabled.

Operation

To set your telephone for handsfree answerback:

- While your set is idle, press the MICROPHONE key. The microphone lamp on the set turns on. You are now able to answer any directed paging calls to your set using handsfree answerback.

To answer a directed page handsfree:

- When you hear a short tone from the set speaker listen for the caller.
- Speak towards the set in a normal tone of voice. Note that your Microphone lamp must be on.

- Press the SPEAKER key or CANCEL key to end the call.

Note: If you receive a directed page call while your set microphone lamp is off, you can answer the page call by pressing the MICROPHONE key. However, when you hang up, the microphone lamp will turn off and you will not be in handsfree answerback mode.

Key Sets - Hold Reminder

Description

This feature reminds a user that there is a call on hold at the set. The user hears a single burst of tone at regular intervals until the call is retrieved from hold.

You can program the length of time that the system waits before providing the first reminder tone, as well as the time interval between the reminder tones.

Conditions

The following conditions apply to this feature:

- This feature is available only on key system telephones.
- A held party will never recall.
- The Hold Reminder feature is available only on *SUPERSET 410, SUPERSET 420,* and *SUPERSET 430* telephone sets that have been programmed as Key System telephones.

See *Hold* for more details on holding calls.

Programming

Complete the following programming:

- Program the length of time that the system should wait before providing the first reminder tone. Enter the desired time (1 to 10 minutes) for COS Option 254 (Call Hold Recall Timer) in the key set telephone's COS.
- Set the time interval between the reminder tones by entering the desired time interval (0 to 600 seconds) for COS Option 681 (Key Set/Sub Att - Call Hold Notify Timer) in the key set telephone's COS. Note that a time of 0 indicates that there will be NO call hold reminder.

Operation

To retrieve a call after hearing the hold reminder:

- Press the flashing held party's key.

Key Sets - Intercom Key

Description

By default, the first key on a key system telephone is the Intercom key - it can be neither deleted nor moved. This is the line with which the user may place/accept internal calls, enable/disable features, access CO line groups and access specific CO lines (via the Direct CO Line Select feature). There may be only one Intercom line on a telephone set.

Conditions

An Intercom key can only be programmed on *SUPERSET 410*, *SUPERSET 420,* and *SUPERSET 430* telephone sets that have been programmed as Key System telephones.

Programming

It is not necessary to program the Intercom key - it is hard-coded, and is always the first entry in the Expand Set Subform of CDE Form 45 (Key System Telephones).

Operation

To originate a call:

- Press the INTERCOM key

  or
- Go off-hook (if the INTERCOM key is the preference -- see *Key Sets - Line Preference)*

To answer a call:

- Press the ringing INTERCOM key

  or
- Go off-hook (if it is the first ringing line).

Key Sets - Line Preference

Description

Allows the system to automatically select which line is used when the set goes off-hook to originate a call. One of the following may be selected:

- Intercom key
- CO line key
- CO line group key
- Personal O/G key
- Manual (user must press a line key to originate a call).

The user may override the line preference by pressing another line key prior to going off-hook for a call origination.

Conditions

The following conditions apply to this feature:

- This feature has no effect on the answering of calls.
- This feature is available only to *SUPERSET 410*, *SUPERSET 420,* and *SUPERSET 430* telephone sets that have been programmed as key system telephones.

Programming To select or change the line preference:

- From within the Expand Set Subform of CDE Form 45 (Key System Telephones), press the LINE PREF softkey, and choose the desired line type.

Operation None.

Key Sets - Off-hook Voice Announce

Description Allows a party to place a directed page to a busy *SUPERSET 420 or SUPERSET 430* key system telephone. A short burst of ringing precedes the voice announcement, advising the busy party that an announcement is following. The announcement is heard through the handset, only by the paged party. The other party will hear silence.

After initiating an Off-Hook Voice Announce, a short burst of ring-back tone is returned to the paging set if the paged set is idle. If the paged set is busy, the paging party will hear a short burst of busy tone.

If the paged party has the COS option *Override Announce* enabled, the paging party makes the announcement immediately after the burst of ringing. If the COS option *Override Announce* is not enabled, the paging party cannot make the announcement until invited to do so by the paged party.

Conditions The following conditions apply to this feature:

- An off-hook voice announce can only be made to *SUPERSET 420* and *SUPERSET 430* telephones that have been programmed as key system telephones (refer to *Key Set/System Support*).
- An off-hook voice announce can only be made from a *SUPERSET 410*, *SUPERSET 420*, or *SUPERSET 430* telephone that has been programmed as a key system telephone (refer to *Key Set/System Support*).
- An off-hook voice announce can only be made to a *SUPERSET 420* and *SUPERSET 430* telephone that is in the talking state, and is not in handsfree mode.
- An off-hook voice announce cannot be made to a set that has DND enabled.
- An off-hook voice announce cannot be made to a set that is involved in a conference.
- An off-hook voice announce cannot be made to a set that has a call on hold, or that is talking to a party that has a call on hold.

Programming Enable COS Option 501 (Override Announce) in the caller's COS.

Program the Direct Paging feature access code (48) in Form 02, or program a Direct Paging feature key in the Expand Set Subform of CDE Form 45 (Key System Telephones).

Operation

To Initiate an Off-hook Voice Announce:

SUPERSET 410 Telephones:

- Dial the Direct Paging feature access code, or press the DIRECT PAGE feature key.
- Dial the extension number, or press the DSS key of the party to be paged.
- When the paged party telephone is idle, a burst of ringing will sound, indicating you may make your announcement.
- When the paged party telephone is busy, a short burst of busy tone precedes the voice announcement.
- If the paged party has COS option *Override Announce* enabled, you may make your announcement immediately following the burst of ringing. If the COS is not enabled, you must wait for the paged party to respond.

SUPERSET 420 and SUPERSET 430 Telephones:

- Dial the Direct Paging feature access code, or press the DIRECT PAGE feature key.
- Dial the extension number, or press the DSS key of the party to be paged.
- When the paged party telephone is idle, a burst of ringing will indicate you are making a paged call. Your set LCD displays PAGING, followed by the called extension number.
- When the paged party telephone is busy, a short burst of busy tone precedes the voice announcement.
- If the paged party has COS option *Override Announce* enabled, you may make your announcement immediately following the burst of ringing. If the COS is not enabled, you must wait for the paged party to respond.

To Answer an Off-hook Voice Announce:

SUPERSET 420 and SUPERSET 430 Telephones which have COS option *Override Announce* enabled:

- You will hear a short burst of ringing, then the paging party's announcement.
- Press, and hold, the RESPOND softkey if you wish to respond to the paging party. Your existing caller will be placed on hold.
- Release the RESPOND softkey to end the connection with the paging caller and return to your held call.

SUPERSET 420 and SUPERSET 430 Telephones without the COS option *Override Announce* enabled:

- You will hear a short burst of ringing, but the paging party cannot be heard until you press, and hold, the RESPOND softkey.
- Your existing caller is placed on hold, and you are connected to the paging party.
- Release the RESPOND softkey to end the connection with the paging caller and return to your held call.

Key Sets - Paging

Description

Key system telephones have access to three different types of paging:

1. PA Paging - based on a Pager and paging zones (see *PA Paging* in this document).
2. Group Page (and Meet Me Answer) - based on the telephone speaker.
3. Directed Page - also based on the telephone speaker.

The last 2 items are applicable to key system telephones only. In both types of key set paging, both the paging party and the paged party receive a single burst of tone to indicate that a page is about to occur. When the paged set is busy or if all sets in the paged group are busy, a short burst of busy tone precedes the voice announcement.

Directed Page: Allows a party to page a specific key system telephone set via its telephone speaker. The connection is one-way audio, and is terminated when the paging party hangs up. Another party attempting to call a set that is being paged in this manner will receive busy tone. The paged party can answer the page as if it were a normal incoming call to the **Intercom** key.

Group Page (Also known as All Page): Allows a party to page all telephones in a paging group simultaneously via their telephone speakers. The connection(s) are one-way audio to each telephone in the page group, and are terminated when the paging party hangs up. A telephone being paged in this manner may originate and receive calls - when this occurs, the paging on that telephone is terminated.

If all the telephones in a paging group become busy during a user's group paging announcement, the user hears two beeps. The user receives the two beeps as soon as the last idle telephone in the paging group becomes busy. The two beeps indicate to the user that nobody can hear the remainder of his or her announcement.

You can program up to 50 paging groups. Each group can have a maximum of 16 members.

Meet Me Answer: Allows a party to respond to a group page. It does not apply to directed page calls. A paged party may respond in this manner if the paging party and the paged party are in the same page group. If a party is involved in a call, but hears the page from another telephone, they may put the current call on hold, and respond to the page. The paged party must respond to the group page within 15 minutes of the termination of the group page - after this, the system cancels the page. A paged party should not try to respond after another group page has been made to another party. If the paging party has not hung up, a Meet Me Answer response will be connected immediately as a normal 2-way conversation. If the paging party has hung up, the response will be treated like a normal telephone to telephone call. Once connected, the call is treated like a normal call, with all other existing features accessible.

Conditions

The following conditions apply to this feature:

- This feature is available only to *SUPERSET 401+, SUPERSET 410, SUPERSET 420,* and *SUPERSET 430* telephones that have been programmed as key system telephones.
- A directed page or group page cannot be made to a set that has DND enabled.
- A party may initiate a group page only if they are a member of a page group.
- Only one group page may be performed to a page group at any one time.
- If a telephone was listening to background music, a group page will cause the music to be turned off. The music will be turned on again when the paging is terminated.
- COS Option 600 (Auto Answer) does not enable *SUPERSET 410, SUPERSET 420,* or *SUPERSET 430* telephone users to respond handsfree to a directed page.
- Key System telephone users cannot make directed page calls to *SUPERSET 401+* telephones.
- *SUPERSET 410, SUPERSET 420,* or *SUPERSET 430* telephone users can respond handsfree to a directed page. Refer to *Key Sets - Handsfree Answerback to a Directed Page* for details.

Programming

For group page and directed page:

- Program the Direct Paging feature access code (48) in Form 02, or program a Direct Paging feature key in the Expand Set Subform of CDE Form 45 (Key System Telephones).
- Program a page group number for each key system telephone in the PAGE field in CDE Form 45 (Key System Telephones) for group page.

For Meet Me Answer:

- Assign an access code to Feature Access Code 49 (Key System - Group Page - Meet Me Answer) in Form 02 (Feature Access Codes).

Operation

To Initiate a Directed Page:

***SUPERSET 401+* Telephones:**

- Lift the handset.
- Dial the Direct Paging feature access code.
- Dial the extension number.
- Broadcast the page message.

***SUPERSET 410* Telephones:**

- Dial the Direct Paging feature access code, or press the DIRECT PAGE feature key.
- Dial the extension number, or press the DSS key of the party to be paged.
- Broadcast the page message.

***SUPERSET 420* and *SUPERSET 430* Telephones:**

- Dial the Direct Paging feature access code, or press the DIRECT PAGE feature key.
- Dial the extension number, or press the DSS key of the party to be paged.
- Broadcast the page message.

To answer a Directed Page:

- Lift the handset, press the flashing line key, or press the SPEAKER key.

To initiate a Group Page:

***SUPERSET 401+* telephones:**

- Lift the handset.
- Dial the Direct Paging feature access code.
- Dial the end-of-dial character (#).
- Broadcast the page message.

***SUPERSET 410* telephones:**

- Dial the Direct Paging feature access code, or press the DIRECT PAGE feature key.
- Dial the end-of-dial character (#).
- Broadcast the page message.

***SUPERSET 420* and *SUPERSET 430* telephones:**

- Dial the Direct Paging feature access code, or press the DIRECT PAGE feature key.
- Dial the end-of-dial character (#).
- Broadcast the page message.

To answer a Group Page:

- Lift the handset.
- Dial the Group Page - Meet Me Answer access code.

Key Sets - Personal Speed Call

Description

Allows the user of a key system telephone to program and access up to five personal speed call numbers. The telephone user enters the numbers at the telephone - they may then be accessed via an access code, followed by an index number. These personal speed call numbers may only be accessed from the telephone on which they were entered.

This feature applies only to key system telephone sets. Note that, in addition to this feature, key system users also have access to the Abbreviated Dial feature - see *Abbreviated Dial* in this document.

Conditions

The following conditions apply to this feature:

- Personal speed call numbers may be dialed at any time on any line, provided that they are the first digits dialed, and that an ACD agent or supervisor is not currently logged in at the set.
- There may be a maximum of five stored numbers on one telephone set.
- The maximum length of a stored number is 25 digits.
- This feature is available only on *SUPERSET 401+*, *SUPERSET 410*, *SUPERSET 420*, and *SUPERSET 430* telephone sets that have been programmed as key system telephones.

Programming

Assign feature access codes 51 (Key System - Store Personal Speed Call) and code 52 (Key System - Retrieve Personal Speed Call) in Form 02 (Feature Access Codes).

Note: The Retrieve Personal Speed Call access code must begin with an asterisk (*).

Operation

To store a personal speed call number:

- Dial the Store Personal Speed Call access code
- Dial the desired index number (1 - 5)
- Dial the number to be stored.
- Hang up.

To dial a personal speed call number:

- Dial the Retrieve Personal Speed Call access code
- Dial the index number for the desired stored number.

Language Change

Description This feature allows *SUPERSET 420* and *SUPERSET 430* telephones to be used with a second language. The display text and softkey text are in the second language.

Conditions The language for the *SUPERSET 420* and *SUPERSET 430* telephones is selected by a softkey. The language chosen stays with the Digital Line Card port if the telephone itself is changed.

Programming None.

Operation Operation varies depending upon the type of set as described below.

***SUPERSET 420* Telephones:**

- Press SUPERKEY.
- Press the NO softkey until LANGUAGE? appears in the display.
- Press the YES softkey.
- Press the CHANGE softkey.
- Select the desired language.
- Press SUPERKEY.

***SUPERSET 430* Telephones:**

- Press the SUPERKEY.
- Find and press the LANGUAGE softkey.
- Choose the desired language.

Last Number Redial

Description This feature allows the attendant console and *SUPERSET* telephone users to redial the last manually-dialed internal or external number with a single key operation. Single line telephones may use the Last Number Redial feature using an access code.

Conditions The following conditions apply to this feature on attendant consoles:

- The redial number is changed when an internal destination is successfully dialed or a trunk group is dialed and found busy or an external number is successfully dialed.
- The REDIAL softkey does not appear until a destination has been dialed and the number can be changed.
- The number is not updated after a call is made using the Abbreviated Dial feature.
- The REDIAL softkey can be used while calls are ringing the console.

- If a call is directed to an LDN programmed at the same key as the REDIAL key (softkey 9), the LDN label replaces the REDIAL softkey and the Redial feature is not available.
- The REDIAL key does not appear when the console is locked out; see *Attendant Console Lockout*.
- The redial number is lost after a system reset.

The following conditions apply to this feature on telephones:

- The Redial feature is not available when a consultation hold is in progress.
- The redial number is changed when an internal destination is successfully dialed or a trunk group is dialed and found busy or an external number is successfully dialed.
- A redial is permitted when the conditions mentioned immediately above are met. On display telephones, the REDIAL softkey is presented.
- The number is not updated when dialing on direct trunk select keys, private trunk lines, or on CO line group keys.
- The number is not updated after an external call is made using a speedcall or a system abbreviated dial.
- The redial number is lost after a system reset.
- When the REDIAL softkey on *SUPERSET* display telephones is pressed, the rules for initiating dialing and selecting a line apply as if the user had selected a speedcall key; see *Line Selection*.
- The redial number contains only the digits dialed until ARS dialing is completed, and the two parties are talking without further ARS processing. Digits dialed during the established call are not stored in the redial number.
- The Forced Account Codes feature disables Last Number Redial.
- Hotel/Motel internal only call restrictions applied to this extension disables Last Number Redial.

Programming

Enable System Option 29 (Telephone Last Number Redial).

Assign an access code to Feature 30 (Last Number Redial) for use with single line telephones.

Operation

Operation varies depending upon the device as described below.

***SUPERSET 410* and *SUPERSET 420* Telephones:**

- Press the REDIAL key.

Users of *SUPERSET 420* telephones can display the telephone number that is stored in the REDIAL key.

- Press the SUPERKEY and then press the REDIAL key.
  The redial number appears in the display. If the number contains more than 16 digits, the MORE softkey will be present.
- Press the MORE softkey to display the remaining digits.
- Press the SUPERKEY to exit.

Attendant Console and *SUPERSET 430* Telephones:

- Press the REDIAL softkey.

Industry-standard and *SUPERSET 401+* Telephones:

- Lift the handset - dial tone is returned.
- Dial the Last Number Redial feature access code.

Last Party Receives Dial Tone

Description This feature allows the last party left on a call, after the other party(s) hang up, to receive dial tone and be able to dial. Normally, this party would receive silence and after 30 seconds be locked out; see *Line Lockout.*

Conditions The following conditions apply to this feature:

- The feature applies only to station and *SUPERSET* telephones.
- The feature has no effect on *SUPERSET* telephones which have the handset in the cradle, have the auto answer feature set, or are on a line that cannot be used for originating calls.
- The feature has no effect on station sets that are members of recording hunt groups.
- This feature does not work on telephones with either COS Option 228 (Manual Line) or COS Option 241 (Receive Only) enabled in their class of service.

Programming Enable System Option 22 (Last Party Clear - Dial Tone).

Operation Establish a call. All of the parties in the call except one hang up. The remaining extension hears dial tone and can dial a new call.

Line Lockout

Description The PABX locks out an extension if the extension goes off–hook and does not dial digits or go back on–hook for a length of time. Lockout also occurs if the extension does not hang up at the end of a call. In the locked-out state, the extension cannot originate or receive calls, and appears busy to potential callers.

See *Lockout Alarm* for an alarm generated from line lockout.

Conditions The following conditions apply to this feature:

- Dial tone time-out is 15 seconds, with an additional 30 seconds of reorder tone before lockout is applied to the extension.
- Calls cannot be transferred to locked out extensions.
- Locked out extensions are not hunted in hunt groups.

Programming None.

Operation To remove the extension from the locked-out state, go back on-hook.

Line Privacy

Description This feature ensures that calls on key lines, direct trunk select lines, private trunk lines and CO line keys are private. Line privacy is assigned to extensions and trunks. If an extension is assigned with line privacy, calls made from that extension are private. Other users cannot enter the call by selecting another appearance of the line. If a trunk is assigned with line privacy, a user cannot enter a call on that trunk by selecting another line appearance of the trunk.

A *SUPERSET* telephone user can override Line Privacy using Privacy Release to allow other sets to join a conversation. Refer to *Privacy Enable/Privacy Release* for details.

Conditions The following conditions apply to this feature:

- You control line privacy by enabling or disabling COS Option 240 (Line Privacy) in the COS of an extension or trunk. By default, COS Option 240 (Line Privacy) is enabled in the COS of extensions and trunks.
- You can only disable line privacy on key lines. Line privacy cannot be disabled on key lines of logical lines, direct trunk select lines, private trunk lines and CO lines.
- *SUPERSET 410, SUPERSET 420,* and *SUPERSET 430* telephones can use the Privacy Release feature to disable privacy during a call.
- A maximum of five parties can be on the line at a time (including the industry-standard telephone, *SUPERSET* telephone, or trunk).
- Direct line selection by the user is the only way to join a call in progress on a line. The system does not choose an occupied line automatically; see *Line Selection.*

Programming To disable line privacy, disable COS Option 240 (Line Privacy) in the COS of the industry-standard telephone, *SUPERSET* telephone, or trunk.

Operation The two modes of operation are described below.

Line Privacy Enabled

When the line is in use, attempts by other *SUPERSET* telephones with appearances to access the line are ignored. If it is a key line of an industry-standard telephone, and the telephone goes off–hook then it receives busy tone.

Line Privacy Disabled

When the line is in use, any other appearance that accesses the line joins the conversation, preceded by the override warning tone. For key lines of industry-standard telephones this includes the station when it goes off–hook.

Line Selection

Description

The *SUPERSET* telephones are equipped to have many line appearances programmed on them. When the user of the telephone initiates dialing, the system selects a line for dialing if programmed to do this. When the set is ringing and the user goes off–hook, the system selects the line to answer. The user can also select a specific line to place or answer a call.

When the user goes off–hook to dial, the system checks for an available line appearance. The system looks at the prime line first. If it is idle then it is selected. Otherwise the rest of the lines on the set are scanned for the first personal line that is idle or a single appearance key or multicall line that is idle and is not an in-only line. If such a line is found then it is selected for dialing. Direct trunk select, CO line and private trunk lines are never automatically selected and key and multicall lines that have more than one appearance (in the system) are never selected. If a line is selected then the user is put dialing on that line. If there are no available lines then the user is prompted to select a line to dial on. For add features to control line selection on key sets, see *Key Sets - Line Preference.*

If the user initiated dialing by dialing a digit or pressing a speedcall key, the same line selection process is done as with going off–hook. When the system or the user selects a line, the system then automatically initiates speedcall dialing for the SPEEDCALL key initially pressed.

This automatic selection can be disabled using COS Option 604 (*PBX SUPERSET* Telephone - Automatic Outgoing Line). If disabled, the user must always manually select a line when originating a call. Handsfree dialing on the keypad to initiate a call from idle is ignored. Note that this COS option does not apply to *SUPERSET 401+* telephones, which provide dial tone regardless of whether COS Option 604 is enabled.

When a user goes off–hook to answer a call, the system scans the set from the prime line up the set, to find a line that is actually ringing the set. The system selects the first line found and answers the call on that line.

If this feature is not selected, handsfree dialing is prevented. Also, when a speedcall key is pressed, the set will wait for a line select key before dialing. The set will also wait for a line key selection after (a) the SPEAKER key is pressed, or (b) when the set goes off-hook.

Conditions

The following conditions apply to this feature:

- The line selection mode is entered when a line is put on hold on a *SUPERSET* telephone; see *Hold.*
- If the user presses a speedcall key and has to select a line to use, if a direct trunk select or private trunk line is selected then the speedcall key is ignored.
- Line selection after pressing the REDIAL softkey is treated the same as pressing a speedcall key.

Programming Enable COS Option 604 for *SUPERSET* telephones to have the system select a line for outgoing calls.

Operation None.

Line Types and Appearances

Description *SUPERSET* telephones are equipped with keys that can be used as line select keys to provide additional lines to the telephones, appearances of other telephones, and direct access to trunk lines.

SUPERSET telephones have two components then - the telephone and the lines on the telephone. The telephone and the lines are not always busy at the same time. The system sometimes has to distinguish between the *SUPERSET* telephone and the lines on the telephone. One or more lines (except the prime line) may be in use, but the telephone itself is still idle and available for a call.

Lines have root devices, which can be telephones, *SUPERSET* telephones, or logical lines.

Every *SUPERSET* telephone must have a Prime Line (or Intercom). The PABX identifies a set by the set's prime line extension number.

- A prime line can be a multicall or key line depending upon the type of the next programmed appearance of the prime line. If an appearance of a prime line is programmed at other sets as a multicall line, then the prime line is a multicall line. If an appearance of a prime line is programmed at other sets as a key line, then the prime line is a key line.
- The prime line is always a both way and immediate ring line.

Line types:

- Personal Outgoing Line
- Key Line
- Multicall Line
- Direct Trunk Select (DTS) Line
- Private Line.

Other line types are describe under subattendant and key set feature headings.

Line Appearance Variants

1. Direction: Both Way, Incoming Only, Outgoing Only. - A line appearance can be restricted to use for originating calls only, incoming calls only, or both.

Note: The outgoing only direction for a line appearance on a set is ONLY available if the line is programmed for no ring; if programmed for delayed ring or immediate ring then the line must be either incoming only or both way.

2. Ring: No Ring, Delayed Ring, Immediate Ring. - This allows new calls to an appearance to cause a *SUPERSET* telephone to ring immediately, ring after a delay or not ring at all.

Note: The duration of the delay time is determined by the class of service of the root device (trunk, *SUPERSET* telephone, industry-standard telephone) of the line appearance. This is determined by COS Option 263, Delay Ring Timer. For logical lines, the first *SUPERSET* telephone where the line is programmed is used.

3. Secretarial: Non-Secretarial, secretarial. - This allows special interaction with the Do Not Disturb feature; see *Secretarial Line.*

Line Ringing

For key line types, there can be only one caller calling the line at a time. When the caller calls the line, all of the appearances indicate a ringing line and the *SUPERSET* telephones where the appearances appear may start to ring if the *SUPERSET* telephones are idle.

Each line on a *SUPERSET* telephone that rings can cause the *SUPERSET* telephone to ring if the set is not in use and as long as the set does not have the Auto-Answer feature activated. If the *SUPERSET* telephone is in use and off–hook or the Auto-Answer feature is enabled then each line that starts to ring causes the set to warble briefly if the set is off–hook. This new call ring can be limited with COS Option 611 (*SUPERSET* Telephone - Limited New Call Ring) so that only the first ringing line on a telephone provides the short ring.

Conditions

The following conditions apply to this feature:

- Any line of any line type can have a maximum of 16 appearances, including the prime line of a set.
- An extension or trunk can be the root of only one line type.
- See *Call Forwarding* for details on lines and forwarding.
- For some features, the programmed sequence of appearance is important; the Review feature in CDE allows the sequence to be viewed.

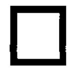
Programming

Specify the delay ring time in COS Option 263, Delay Ring Timer.

For *SUPERSET* telephones with heavy call traffic on several lines on the set, enable COS Option 611 (*SUPERSET* Telephone - Limited New Call Ring) to limit the new call ring given to the set.

Operation

None.

SYMBOL	DESCRIPTION	MEANING
	NO SYMBOL IN DISPLAY	LINE IDLE
	DISPLAY ALTERNATES BETWEEN THESE TWO SYMBOLS	INCOMING CALL
	DISPLAY SOLID SQUARE	LINE BUSY AT THIS SET
	DISPLAY SOLID CIRCLE	LINE BUSY AT ANOTHER SET
	DISPLAY ALTERNATES BETWEEN SOLID AND CLEAR	CALL ON HOLD AT THIS SET
	DISPLAY CIRCLE FLASHES ON AND OFF	CALL ON HOLD AT ANOTHER SET

CC0069

Figure 2-1 ***SUPERSET*** **Telephone Line Status Display Symbols**

Line Type: Personal O/G Line

Description This line is an outgoing-only line that allows the user to make an outgoing call without making the prime line busy to incoming calls.

Conditions The following conditions apply to this feature:

- It has no extension number associated with it.
- There can only be one per *SUPERSET* telephone.

Programming From within the Expand Set Subform of either CDE Form 09 (Station/*SUPERSET* Telephones) or CDE Form 45 (Key System Telephones), move the cursor to the TYPE field. Press the PERSONAL O/G softkey.

Operation None.

Line Type: Key Line

Description

This is an appearance of an extension number that can be an industry-standard telephone's line, a *SUPERSET* telephone's Prime Line or a logical line. If the line is in use at one set, the other appearances of the line are busy and unavailable for separate calls. See the *Line Privacy* feature description in this practice for joining established calls on key lines.

Conditions

The following conditions apply to this feature:

- Direction and ring variants can be programmed independently for each appearance.
- It can appear on several *SUPERSET* telephones.
- There can be only one appearance of any one key line on a *SUPERSET* telephone.

Programming

From within the Expand Set Subform of either CDE Form 09 (Station/*SUPERSET* Telephones) or CDE Form 45 (Key System Telephones), move the cursor to the TYPE field. Press the KEY LINE softkey.

Operation

None.

Line Type: Multicall Line

Description

A multicall line is an appearance of one extension number on two or more telephones. This extension number can be for another *SUPERSET* telephone or an industry-standard telephone. When one appearance of a multicall line is in use, the other appearances are still available to make or answer calls.

- Call direction, ring and secretarial variants can be programmed independently for each appearance.
- There can be up to 15 appearances of any one multicall line on a given *SUPERSET* telephone.

Conditions

The following conditions apply to this feature:

- Direction, ring and secretarial variants can be programmed independently for each appearance.
- It can appear on several *SUPERSET* telephones.
- There can be up to fifteen appearances of any one multicall line on a given *SUPERSET* telephone.
- The secretarial operation feature description in this practice only applies to multicall lines.
- When a multicall line is called, there can be as many callers ringing the line as there are appearances that are free to be rung. When a caller

rings the line, the first appearance of the called line is rung on all *SUPERSET* telephones where the line appears. The next caller of the line makes the next appearance on all *SUPERSET* telephones (the next level or tier of appearances) ring. On a single *SUPERSET* telephone, the appearances of a multicall line are rung in order from lowest key number to highest key number.

Programming From within the Expand Set Subform of either CDE Form 09 (Station/*SUPERSET* Telephones) or CDE Form 45 (Key System Telephones), move the cursor to the TYPE field. Press the MULTI-CALL softkey.

Operation None.

Line Type: DTS Line

Description A Direct Trunk Select (DTS) Line operates like a key line, but it directly accesses a specified CO trunk. It can be used for incoming and outgoing calls.

- The DTS trunk can appear on several *SUPERSET* telephones.
- There can be only one appearance of any one direct trunk select line on a *SUPERSET* telephone.
- Direction and ring variants can be programmed independently for each appearance. The direction variant allows control of the type of trunk call, incoming or outgoing or both.
- The user can transfer calls on this line to other extensions.
- DTS outgoing calls bypass ARS.
- DTS calls use the SMDR feature if enabled.

For further information, see *Direct Trunk Select* in this document.

Conditions See *Direct Trunk Select*.

Programming From within the Expand Set Subform of CDE Form 09 (Station/*SUPERSET* Telephones), move the cursor to the TYPE field. Press the DIR TRK ACC softkey. Note that the trunk must be programmed into a trunk group.

Operation None.

Line Type: Private Line

Description Like a DTS Line, a private line accesses a specified dedicated CO trunk directly. However, the user can only transfer established calls on this line to other *SUPERSET* telephones that have an appearance of the line, using Privacy Release (see *Privacy Enable/Privacy Release*).

Conditions The same conditions apply as for direct trunk select lines.

Programming From within the Expand Set Subform of CDE Form 09 (Station/*SUPERSET* Telephones), move the cursor to the TYPE field. Press the PRIVATE TRK softkey.

Operation None.

Lockout Alarm

Description The system locks out any set that remains off-hook and not connected to another set or trunk for more than 45 seconds. The lockout alarm feature:

- generates an audible alarm through the console
- activates the alarm relays
- displays the location of the locked out device.

When a set is locked out, if lockout alarm is enabled, all consoles warble with a long-short-long cadence. This cadence overrides other cadences that might be active. The attendant can display the time and date the lockout alarm occurred, the extension number of the device, and a message stating that the device has been off-hook too long.

Conditions The following conditions apply to this feature:

- COS Option 227 (Lockout Alarm Applies) must be enabled for the stations or sets for this feature to operate.
- COS Option 102 (Attendant Display of System Alarms) must be enabled for the alarm icon to flash when a lockout alarm occurs.
- A lockout alarm occurs 45 seconds after the station or set goes off-hook.
- The alarm totals are updated when a lockout alarm occurs.
- There can be 32 lockout alarms active at any one time; further lockout alarms are only recorded in the maintenance logs.
- The lockout alarm cadence still rings if COS Option 100 (Attendant Bell Off) is enabled.
- The lockout alarm feature operates with COS Option 701 (No Dial Tone) enabled.
- When a console is in restricted service, the lockout alarm cadence rings but the user is not able to read the alarm or turn the warbling off. The restricted access code must be re-dialed before the user can read the alarm.

- When no DTMF receivers are available, the 45 second time period before the lockout alarm occurs is increased by the time required to wait for an available receiver.

Programming

Enable COS Option 227 (Lockout Alarm Applies) in the class of service of the extensions for which the feature applies.

Enable COS Option 102 (Attendant Display of System Alarms) for those consoles which can read and cancel the lockout alarms.

Enable COS Option 108 (Attendant Audible Lockout Alarm) to cause the console to audibly ring for a lockout alarm.

Operation

Display a lockout alarm on a console by pressing the following:

- Select the FUNCTION hardkey.
- Select the APPLICATION softkey.
- Select the SHOW LOCKOUT softkey.

Logical Lines

Description

A logical line is a line on a *SUPERSET* telephone that is not an appearance of any station or other *SUPERSET* telephone. Each logical line has its own extension number. A logical line can appear on up to 16 multi-line *SUPERSET* telephones. Logical line extension numbers can be used in many places where station or *SUPERSET* telephone lines can be programmed (Call Rerouting, etc).

Conditions

The following conditions apply to this feature:

- Logical lines are either key or multicall type lines. Logical lines are not available to single-line sets.
- The *SUPERSET* telephone where the first appearance of the logical line resides is used for COS, COR, and tenant checks.
- Logical lines may be viewed by using CDE Form 09 (Station/*SUPERSET* Telephones) review procedures.
- Logical lines are created implicitly when a vacant extension number is programmed in for a line key.

Programming

Refer to CDE Form 09 (Station/*SUPERSET* Telephones).

Maintenance

Description The *SX-200* ML PABX provides extensive maintenance coverage. All types of peripheral hardware are periodically tested automatically by the system. Maintenance users may also test individual circuits on demand. The system also maintains a fault / event history log (maintenance log).

Conditions None.

Programming Feature Access Code 18 (Maintenance Functions - Test Line) allows a maintenance telephone to be used for testing - see the *General Maintenance Information Practice*, for further information.

Operation Refer to the *General Maintenance Information Practice*.

Manual Line (Dial 0 Hotline)

Description When a manual line extension goes offhook it is routed directly to the extension's dial 0 routing point. The extension can still receive calls.

Conditions The following conditions apply to this feature:

- The extension does not receive dial tone, but does receive ringback tone.
- If there is no Dial 0 routing point for the current night service then reorder tone is given to the extension.
- Priority Dial 0 applies.
- The call is made as a reroute to the Dial 0 point.
- COS Option 228 (Manual Line) and COS Option 300 (Automatic Callback) are mutually exclusive.

Programming Enable COS Option 228 (Manual Line) for the extension.

Enter appropriate answer point (e.g., an Attendant LDN) in Station Dial 0 Routing in CDE Form 19 (Call Rerouting Table) for the extension's tenant.

Operation To originate a call, lift the handset; the dial 0 point rings.

Messaging - Advisory

Description

This feature allows *SUPERSET 420* and *SUPERSET 430* telephone users to display a short message on display sets and consoles that call their set. The message replaces the time and date display on the sets where it is activated; see *Attendant Setup Of Set Advisory Messages.* *SUPERSET 430* telephone users can also program messages to be displayed. The system provides the following system-wide messages:

Message Number	Default Message
01	IN A MEETING
02	OUT OF TOWN
03	ON VACATION
04	OUT ON A CALL
05	OUT TO LUNCH
06	GONE FOR DAY
07	GONE HOME
08	IN TOMORROW
09 through 15	(BLANK)

Optionally, set users can be permitted to change these messages.

Note: Any changes to a message is applied system-wide; the system has only one set of 15 messages.

Conditions

The following conditions apply to this feature:

- This feature applies only to *SUPERSET 420* and *SUPERSET 430* telephones.
- A message currently in use cannot be altered.
- The message is not displayed by the telephone that set it when it is in Night Service; however, another telephone (or console) calling that telephone receives the message on its display.

Programming

To permit set users to create or change messages, enable COS Option 605 (*SUPERSET* Telephone - Message Program) in the set's COS.

Operation

Operation varies depending upon the type of set as described below.

***SUPERSET 420* Telephones:**

To activate an advisory message:

- Press SUPERKEY.
- Press the NO softkey until ADVISORY? appears in the display.
- Press the YES softkey. Message #1, IN A MEETING, appears in the display.

- Press the TURNON softkey to select the IN A MEETING message or,
  Press the NEXT softkey until the desired message is displayed. You can also display a specific advisory message by dialing its 2-digit number (see preceding table).
- When the desired message is displayed, press the TURNON softkey. The selected message is displayed on your set.

To deactivate an advisory message:

- Press SUPERKEY.
- Press the NO softkey until ADVISORY? appears in the display.
- Press the YES softkey. The message that is currently programmed at your set appears in the display.
- Press the TURNOFF softkey. The message is turned off and your set displays the date and time.

***SUPERSET 430* Telephones:**

To activate an advisory message:

- Press the MESSAGING softkey.
- Press the ADVISORY softkey.
- If displayed message is inappropriate, press the NEXT MSG softkey repeatedly to cycle through the repertoire of messages. If the message number is known, press the SHOW MSG NO. softkey, dial the message number (01 through 15) and press the ENTER softkey.
- Press the TURN MSG ON softkey. The selected message is now displayed on the LCD. Any *SUPERSET 420* telephone, *SUPERSET 430* telephone, or console calling the set receives the message on its display.

To deactivate an advisory message:

- Press the MESSAGING softkey.
- Press the ADVISORY softkey.
- Press the TURN MSG OFF softkey.

To program an advisory message:

- Press the MESSAGING softkey.
- Press the ADVISORY softkey.
- Press the SHOW MSG NO. softkey or the Next softkey. Dial an unused message number, unless an existing message is to be changed.
- Press the CREATE MSG softkey. Dial in the message as follows:

Letters are displayed on the LCD as they are dialed on the telephone keypad. The first press of any of these keys displays the first letter that appears on its key cap. The second press of the same key changes the display to the second letter and so on. When all the letters associated with a key have been displayed, the number is displayed. Further presses cycle through the letters again. When the desired letter is displayed, enter it by pressing the → softkey or by entering the next letter, if it is on a different key. (The → key is also used to enter spaces). Follow the same procedure to find and enter the other letters in the name.

Telephone keypad key caps 1, *, 0, and # are not marked with all the characters they can enter. They are as follows:

Key	Characters
1	! ? % 1
*	' - *
0	@ & $ 0
#	., / #

When the message is complete, press the SAVE softkey. The message is now saved.

Messaging - Call Me Back

Description

A set user calling a busy or unanswered set can leave a message for the party to return the call. The Message Waiting indication can be:

- A flashing lamp on the set, flashing at 0.5 seconds on, 3.5 seconds off (if equipped),
- An indication on the set's display (if equipped),
- Ringing at the set with a distinctive ringing pattern.

The Message Waiting indication continues until the set user reads the message. Messages can be read at any time (i.e., when the set is idle or during a call).

On *SUPERSET 420* and *SUPERSET 430* telephones, the display provides the time of the call, and the caller's extension number and name (if programmed).

Optionally, the system can be programmed to record each occurrence of Message Waiting on the system printer. See *Hotel/Motel*.

Messages can be left at industry-standard and *SUPERSET* telephones.

Conditions

The following conditions apply to this feature:

- Use the *Message - Call Me Back* feature if you know that the person you are trying to contact is out of the office. Use the *Callback* feature if you know that the person you are trying to contact is somewhere in the office.
- Only one message exists between any two parties in the system at anytime. Leaving a message from telephone A to telephone B cancels the message left from telephone B to telephone A.
- The message waiting indication is not active while the set is in use.
- When sending a message using a feature access code, if the destination set cannot receive messages, reorder tone is returned.

- Messages cannot be left for members of a recording group or an Automated Attendant recording group.
- A message is canceled automatically if the sender and receiver have a telephone conversation before the message is read or the receiver leaves a message for the sender. Note that this does not apply to messages from the console. These must be canceled at the Console.
- The system cancels the message automatically after 24 hours if System Option 7 (Cancel 24-hour Message Waiting) is enabled; otherwise, it is not canceled by the system.
- 250 messages can be active at a time.
- If COS Option 231 - Message Waiting Setup - Bell is enabled in the industry-standard or *SUPERSET* telephone's COS:
  - Message Waiting indication is three cycles of 3.5 impulse per second (ips) ringing. This is in addition to the visual indicators on *SUPERSET* telephones.
  - The set receives ringing:
    1) 10 seconds after the message is sent if the set was idle, or in do not disturb, or idle in auto answer mode, at the moment when the message was sent,
    2) each time it returns to idle (10 seconds after),
    3) and every 20 minutes while the set is idle until the message is read, canceled or acted upon.

Programming

To allow industry-standard telephones in a COS to receive messages, enable COS Option 231, Message Waiting Setup - Bell or COS Option 232, Message Waiting Setup - Lamp if the industry-standard telephone is equipped with a message waiting lamp.

Note: These two COS Options are mutually exclusive. Neither of them is required for messaging on *SUPERSET* telephones.

To allow users of industry-standard telephones to send messages, enable COS Option 259 (Message Sending) in their COS. Enable COS Option 259 (Message Sending) in the COS of *SUPERSET* telephones to allow users to send messages using the MESSAGE key.

Note: COS Option 259 enables and disables both methods of sending messages.

Industry-standard telephones need access codes to send or answer messages. In CDE Form 02 (Feature Access Codes), assign access codes to Feature 41 (Send Message) and Feature 42 (Call Message Sender of Oldest Message).

Enable COS Option 231 (Message Waiting Setup - Bell) to allow *SUPERSET 410*, *SUPERSET 420*, and *SUPERSET 430* telephones to warble when a message is sent.

In CDE form 04 (System Options/System Timers), enable System Option 07 (Cancel 24-hour Message Waiting) if messages are to be canceled by the system.

Operation

Industry-standard Telephones:

To send a message signal to a busy or unanswered set:

- Dial the Send Message access code.
- Dial 1 to activate the message waiting signal.
- Dial the extension number.
- Dial tone is returned. Re-order tone is returned if the other telephone cannot receive messages.

To cancel a message signal previously sent:

- Dial the Send Message access code.
- Dial 2 to cancel the message waiting signal.
- Dial the extension number.
- Dial tone is returned.

To answer a message:

- Go off–hook.
- Dial the Call Message Sender of Oldest Message access code.
- Message sender is called. Re-order tone is returned if there are no messages waiting.

To clear the oldest message at your set:

- Go off-hook.
- Dial the Send Message access code.
- Dial 3 to clear the message.
- Dial tone is returned.

***SUPERSET 401+* and *SUPERSET 410* Telephones:**

To send a message waiting signal:

- While receiving busy tone or ringback, press the MESSAGE key if the Message LED is on steady.
- Dial tone is returned, or,
- Follow the procedure given for industry-standard telephones. This is convenient when leaving a message without calling the message recipient.

The message LED flashes when an idle telephone has a message waiting for it.

To answer a message waiting signal:

- Lift the handset and press the MESSAGE key (*SUPERSET 401+* telephones)
  or,
  Press the MESSAGE key (*SUPERSET 410* telephones).
- The message sender is called.

To cancel a message waiting signal at your set:

- Press the MESSAGE key. You automatically ring the extension of the person that left the message signal.
- If the extension is busy or there is no answer, press the MESSAGE key again to cancel the message waiting signal at your set, and send a message signal back to the station that sent the original signal.

***SUPERSET 420* Telephones:**

To send a message waiting signal:

- Press the MESSAGE key if you receive busy or no answer when you call a station (your message lamp must be on steady). MESSAGE SENT appears briefly in the display and the call is terminated,
  or,
  Follow the procedure given for industry-standard telephones. This is convenient when leaving a message without calling the message recipient.

The message LED flashes whenever the telephone is idle and has a message waiting for it.

To respond to a message waiting signal at your set:

- Press the MESSAGE key if your Message Lamp is flashing.

  The extension number of the caller, and the time that the message signal was sent appears in the display.
  or
  The name of the caller appears in the display. Press the MORE softkey to display the extension number of the caller, and the time that the message signal was sent.
- Press the CALL[1] softkey to ring the extension. This softkey appears only when the telephone can dial the call (e.g., idle or listening to dial tone),
  or,
  Press the ERASE softkey to delete the message signal.
- Press SUPERKEY to return to the date and time display.

***SUPERSET 430* Telephones:**

To send a message:

- While receiving busy tone or ringback, press the LEAVE A MSG softkey. This softkey appears only if the other telephone is able to receive messages.
- The top line of the display briefly shows a confirmation and the call is terminated, or,
- Follow the procedure given for industry-standard telephones.

1. This softkey will not appear on Key System telephones which have no free internal lines.

The message LED flashes whenever the telephone is idle and has a message waiting for it. Also, the second line of the LCD shows on the right the number of messages waiting; e.g., < 1 MSG >.

To read the messages:

- If engaged in a call, press the SUPERKEY.
- Press the MESSAGING softkey; the top line of the LCD shows the number of messages waiting.
- Press the READ MSG softkey. The LCD second line shows the extension number and the time of the call.
- Press the CALL[1] softkey to call the party that left the message. This softkey appears only when the telephone can dial the call (e.g., idle or listening to dial tone), or,
- Press the ERASE softkey to cancel the message.

If there are more messages, the NEXT MSG softkey appears. To read:

- Press the NEXT MSG softkey; the next message is displayed. Follow the procedure above for each message.
- When there are no more messages, press the SUPERKEY to exit.

To view messages sent:

SUPERSET 430 telephones also allow you to view the messages that you have sent. To view messages that you have sent:

- Press MESSAGING softkey.
- Press MSGS I SENT softkey.
    - For each message, the set displays the name, extension number, time and date that the message was sent.
    - A softkey also appears that allows you to cancel the message.

Meter Pulse Collection

Description

Meter pulses are often used to calculate the cost of outgoing trunk calls thus allowing the call to be charged back to the originator. The system can be set up to detect and collect certain types of meter pulses sent to a trunk circuit during outgoing calls; these are then recorded in the trunk's SMDR reports. Types of meter pulses which can be detected by the PABX without additional hardware include:

- Tip-Ring reversals
- XT lead signaling (Analog CO Trunk)
- M&MM lead signaling (Digital LS/GS Trunk).

1. This softkey will not appear on Key System telephones which have no free internal lines.

Refer to the *Installation Information Practice*, for location of correct leads and proper interface to the sending equipment.

Other types of meter pulses common in the telephone industry include 50Hz, 12 kHz, and 16 kHz type pulses. Detection of these types requires the addition of an external interface which converts these pulses to a ground signal which is then applied to the XT Lead for the analog CO Trunks, or to the M or MM lead for the Digital LS/GS trunks (for Digital LS/GS trunks, -48 V dc must be applied to the other lead so that when the ground is applied to the M or MM lead, current flows through the circuit and gets detected as a pulse).

This feature is associated with the Message Registration feature. See *Hotel/Motel Features* for additional information.

Conditions

The following conditions apply to this feature:

- The PABX can only detect and collect the types of meter pulses identified above.
- The trunk must provide answer supervision. This is counted as the first meter pulse.
- Meter pulses are not recorded for ACD agents if System Option 44 (ACD Reports) is enabled.
- The system can record a maximum of 65535 pulses.
- Pulses are always recorded regardless of what is happening to the trunk (hold, talking etc.).

Programming

In the COS for the trunk, enable COS Option 247 (SMDR - Record Meter Pulses) in CDE Form 3 (COS Define).

Enable System Option 23 (Message Register Count Additional Supervisions) in CDE Form 4 (System Options/System Timers).

Enable Option (Far-end Gives Answer Supervision) in CDE Form 13 (Trunk Circuit Descriptors).

Operation

As meter pulses are received, they are collected by the PABX and reported in the trunk's SMDR record.

MILINK Data Module

Description

The *MILINK* data module is a limited distance, asynchronous, digital dataset that converts the data signals of RS-232 serial devices to high-speed digital signals. It allows an RS-232 device, such as a computer or display terminal, to be connected to the PABX through a *SUPERSET 410*, *SUPERSET 420,* or *SUPERSET 430* telephone. The 2-pair cable that connects the *SUPERSET* telephone to the switch carries both voice and data signals, allowing a user to use the RS-232 device and the *SUPERSET* telephone simultaneously.

The *MILINK* data module is positioned beside or under the user's *SUPERSET 410*, *SUPERSET 420*, or *SUPERSET 430* telephone, and connects to a modular jack located on the base of the telephone. The RS-232 device connects to a DB-25 pin connector on the *MILINK* data module. A 9 Vac wall-mount power supply provides power to the *MILINK* data module.

There are three indicator lamps and two buttons on the front plate:

- POWER indicator lamp - on steady when the *MILINK* data module is receiving power; flashes when synchronization is not achieved.
- READY (RxD) indicator lamp - on steady when a call is established; flashes when the *MILINK* data module is receiving data; off when the *MILINK* data module is idle.
- DEVICE (TxD) indicator lamp -on steady when the attached RS-232 device is powered up and ready to transmit; flashes when the *MILINK* data module is transmitting data.
- ATTENTION button - used to signal the system that the user is ready to originate a DTRX call.
- DISCONNECT button - signals the system to disconnect the data session.

Conditions

The following conditions apply to this feature:

- The *MILINK* data module connects to *SUPERSET 410*, *SUPERSET 420*, and *SUPERSET 430* telephones.

Programming

Refer to the *Customer Data Entry Practice*, for programming.

Operation

Refer to the data features described in this guide for operating instructions.

MITEL Application Interface (MAI)

Description

The MITEL Application Interface (MAI) software package allows Mitel computer-based applications to access the system features. MAI is used in conjunction with an external host computer connected via an RS-232 link to a DATASET 2100 device. The DATASET device is connected to a Digital Line Card port on the PABX via a single twisted pair.

This purchased option is enabled in Form 4.

Conditions

The protocol is based on the following communications standards: CCITT X.409, CCITT X.410 and LAPB (CCITT X.25).

Only one MAI port can be programmed on the system.

Programming

The following steps are required:

- Program a Digital Line Card into Form 01 (System Configuration).
- Enable System Option 105 (Mitel Application Interface) in Form 04 (System Options and Timers).

- You must program hunt groups that are used with MAI applications as Recall Appearance Back to Originating Set (RABTOS) type hunt groups. For example, if you want to program several Call Center Manager (CCM) Attendants into a hunt group, you must specify the hunt group as a RABTOS hunt group in CDE Form 17 (Hunt Groups).
- A data circuit descriptor must be set up for the MAI port. This is done in Form 11 (Data Circuit Descriptors). Only three fields require modification:
  - DATASET 2100: Operating Mode SYNCHRONOUS
  - SYNC: Rate Adaption Scheme X.31
  - SYNC: Clock Source SYSTEM
  - Ensure the Session Inactivity Timer in the corresponding Data Circuit Descriptor is set to 0
- The descriptor is then assigned to the circuit in Form 12 (Data Assignment) in the CDN field.
- Within Form 34 (Directed I/O) enter the extension number of the DATASET, press the MAI softkey and the AUTOPRINT softkey.

Operation Operation is dependent upon the application.

Moving Stations and *SUPERSET* Telephones

Description This feature allows extensions to be moved easily from one circuit to another. Previous programming for the extension, such as name, COR, COS, etc. is preserved and moved with the extension.

Conditions The device must be idle when the move is attempted and the new location must have nothing programmed.

Programming Refer to the *Customer Data Entry Practice*, for full details of programming.

Operation Refer to the *Customer Data Entry Practice*, for full details of operation.

Multi-Attendant Positions

Description The system can handle multiple attendant consoles with unique hold slots for each attendant. Incoming trunk calls can be programmed to appear at all consoles, or specific console(s). Similarly, extension Dial 0 calls, Priority Dial 0 calls, Intercept to Attendant calls, etc., can be programmed to appear at all consoles, or at a specific console(s).

Any console in a particular tenant group can switch that tenant group to Night Service or to Day Service; see *Attendant Night/Day Switching.*

See *Tenanting, Recall,* and *Console LDN Keys;* see also *Attendant Transparent Multi-console Operation.*

Conditions

The following conditions apply to this feature:

- When a call appears at more than one console, the first console to answer is connected to the call; other consoles stop being alerted for this call.
- There is a maximum of eleven consoles per system.

Programming

Assign consoles via CDE Form 07 (Console Assignments).

Operation

All operations are identical for all attendant consoles in the same COS and tenant group.

Music-on-Hold (MOH)

Description

A customer-provided music source can be connected to the system via a Music-on-Hold/Pager module on the Universal Card or via a DNIC Music-on-Hold/Pager Unit connected to a Digital Line Card. This music source can then be used for Campon, Hold, UCD, ACD, and other features.

See *Campon* and *Hold.*

Refer to the *Installation Information Practice*, for wiring details.

Conditions

The following conditions apply to this feature:

- On the DNIC Music-on-Hold/Pager unit, the music source should be between 10 and 100 mV rms.
- On the Universal Card, the music source should be between 50 and 500 mV rms.
- Input to the system is 600 ohms ac transformer coupled. A dc voltage should not be applied to this input.
- Only one source of music may be programmed for the system.

Programming

Program Music-on-Hold in CDE Form 18 (Miscellaneous System Ports) for both methods.

If the DNIC MOH/Pager unit is being used, program the following as well:

- Program a Digital Line Card in Form 1.
- Program the DNIC device type as "DMP" (DNIC Music/Pager) in Form 9.

Operation

None.

Names

Description

The system programmer can assign names to extensions, classes of service, tenants, trunks, trunk groups, ACD paths, ACD positions, ACD agent groups, and hunt groups. *SUPERSET 420* or *SUPERSET 430* telephone users can program their name from their set.

Conditions

The following conditions apply to this feature:

- Trunk, trunk group, ACD path, ACD agent group, tenant, and class of service names can be up to 8 characters long.
- Extension and ACD position names can be up to 10 characters long; hunt group names can be up to 12 characters long.
- Hunt groups must have an access code programmed before a hunt group name can be added.
- The programmer cannot enter names from the attendant console because it has no alphabetic keyboard.
- Tenant and COS names appear only on consoles. Set and trunk names appear on consoles, *SUPERSET 420* and *SUPERSET 430* telephones.
- Mixed case is only allowed for hunt groups.
- Programming the name of a *SUPERSET* telephone in CDE overwrites the name programmed at the *SUPERSET* telephone. Also, programming the name at the *SUPERSET* telephone overwrites the name in CDE.
- The characters \, ~, |, {,} are invalid except for hunt group names.
- Trunk groups cannot include single or double quotation marks.
- Hunt group names cannot start with a digit (0 through 9), # or * or have a blank or a - in them.
- Analog Networking information may replace trunk or trunk group names on displays; see *Analog Networking.*

Programming

Refer to the following forms in the *Customer Data Entry Practice*, for further information:

Name Type	CDE Form
ACD Path Names	Form 41 (ACD Paths)
ACD Position Names	Form 39 (ACD Agent Groups) and Form 40 (ACD Supervisors)
Class of Service Names	Form 03 (Class of Service Define)
Hunt Group Names	Form 17 (Hunt Groups)
Set Names	Form 09 (Station/*SUPERSET* Telephones)
Trunk Names	Form 15 (Dial-In Trunks) or CDE Form 14 (Non-Dial-In Trunks)
Trunk Group Names	Form 16 (Trunk Groups)
Tenant Names	Form 19 (Call Rerouting Table)

Features Description

Operation

SUPERSET 420 Telephones:

To program a new name or change an existing name:

- Press SUPERKEY.
- Press the NO softkey until NAME? appears in the display.
- Press the YES softkey
  - If a name isn't currently programmed, ENTER NAME appears in the display.
  - If a name is programmed, the name is displayed. Press the CHANGE softkey. ENTER NAME appears in the display.

Enter the name using the telephone keypad. Above each key are printed its associated letters, e.g., the "2" key has the letters "abc" above it. To choose the first letter, press the key once; to choose the second letter, press the key twice, etc. When the last associated letter appears, a subsequent press displays the key number. Further key presses cycle through the letters again. When the desired letter is displayed, press the → softkey to enter it. To correct an erroneous entry, use the ← softkey.

Telephone keys 1, *, 0 and # are not marked with all the characters they can enter. They are as follows:

Key	Characters
1	! ? % 1
*	' - *
0	@ & $ 0
#	. , / #

- When the name is complete, press the SAVE softkey.

To clear an existing name:

- Press SUPERKEY.
- Press the NO softkey until NAME? appears in the display.
- Press the YES softkey. The name currently programmed for the set is displayed.
- Press the ERASE softkey. NAME ERASED appears briefly in the display and then shows the date and time.

SUPERSET 430 Telephones:

To program a name:

- Press the SUPERKEY.
- Press the YOUR NAME softkey. The display changes to ENTER NAME:

Letters are displayed on the LCD as they are dialed on the telephone keypad. The first press of any of these keys displays the first letter that appears on its key cap. The second press of the same key changes the display to the second letter and so on. When all the letters associated with a key have been displayed, the number is displayed. Further presses cycle through the letters again. When the desired letter is displayed, enter it by pressing the → softkey or by entering the next letter, if it is on a different key. (The → key is also used to enter spaces). Follow the same procedure to find and enter the other letters in the name. If an error is made, press the ← softkey to back up and change a letter.

Telephone keys 1, *, 0 and # are not marked with all the characters they can enter. They are as follows:

Key	Characters
1	! ? % 1
*	' - *
0	@ & $ 0
#	., / #

- When the name is complete, press the SAVE softkey.

Never a Consultee

Description This feature protects an extension from being dialed or retrieved by extensions that have a consultation hold in progress.

Conditions The following conditions apply to this feature:

- The feature is checked when a caller is retrieved from being held by an extension or by the console.

Programming Enable COS Option 233 (Never a Consulted) for the extension.

Operation An extension establishes a call. The extension flashes, putting the call on consultation hold, and dials a third party that has the feature enabled. The call to the third party is blocked and the calling extension receives reorder tone.

Never a Forwardee

Description This feature prevents an extension or console from having any calls forwarded to it by another extension. Extensions are prevented from setting up forwarding to extensions or consoles with the feature enabled.

Conditions The following conditions apply to this feature:

- Calls directed to the extension via hunting or speedcalls are not affected by the selection of this feature.
- If an extension attempts to forward a call to an extension with this feature enabled (the feature was enabled after the forwarding was setup), the forwarding is ignored.

Programming Enable COS Option 234 (Never a Forwardee) for the extension or console.

Operation None.

New Call Ring

Description When a *SUPERSET* telephone is busy, and a new call attempts to ring the set, a single burst of ringing will alert the user that another call is waiting.

Conditions The tone is not heard if the *SUPERSET* telephone is using the speaker in handsfree mode.

Programming None.

Operation None.

Night Bells

Description This feature allows incoming and internal calls to be directed to common alerting devices. The call can be answered from the attendant console or from an extension with TAFAS Access; see *Trunk Answer From Any Station (TAFAS).*

The system provides a contact closure which operates the alerting device.

Night Bells are activated by relays located on a Universal Card Receiver/Relay module or on the DNIC Music-on-Hold/Pager Unit; refer to the *Installation Information Practice*.

The extension number assigned to the night bell can be used as an answer point or alternate answer point for most features in the system.

Night bells can be integrated with the paging system on the DNIC Music-on-Hold/Pager Unit. A warble tone is available on the paging system when a night bell is active. The night bell waits for an idle pager.

Conditions The following conditions apply to this feature:

- Up to 25 night bells can be programmed.
- See *Attendant Default Call Positions* for a description of the console NIGHTBELL softkey.
- More than one caller can ring a night bell at a time - it is always available to be called and no caller will find it busy.
- Extensions can dial a night bell directly.
- The night bell itself has no tenant. Tenant checks are done using the tenant of the night bell's caller.

- Standard ringing cadence only is used for night bells.
- Calls can be transferred to night bells.
- Recall features do not operate when ringing a night bell.

Programming Assign each Night Bell in CDE Form 18 (Miscellaneous System Ports).

If the DNIC Music-on-Hold/Pager Unit is being used, program the following as well:

- Program a Digital Line Card in Form 1.
- Program the DNIC device type as "DMP" (DNIC Music/Pager) in Form 9.
- Program a pager number in Form 18 with each night bell.

Operation See *Trunk Answer From Any Station (TAFAS).*

Night/Day Switching

Description *SUPERSET 420* or *SUPERSET 430* telephone users can put the system (or particular tenant group or groups) into DAY service or one of two different night service modes, NIGHT1 or NIGHT2. In Night Service, *SUPERSET 420* or *SUPERSET 430* telephones display NIGHT 1 SERVICE or NIGHT 2 SERVICE as appropriate. Also see *Night Services, Tenanting,* and *Attendant Night/Day Service Switching.*

Conditions The following conditions apply to this feature:

- This feature is available at *SUPERSET 420* and *SUPERSET 430* telephones only.
- Tenanting restrictions can be applied; see *Tenanting.* Refer to the *Tenanting Practice.*

Programming Enable COS Option 609 (*SUPERSET* Telephone - Night Service Switching) in the set's class of service.

Operation Operation varies depending upon the type of telephone as described below.

***SUPERSET 420* Telephones:**

- Press SUPERKEY.
- Press the NO softkey until NIGHT SERVICE? appears in the display.
- Press the YES softkey. The current service mode is displayed: DAY SERVICE, NIGHT 1, or NIGHT 2.
- Press the CHANGE softkey.
- If the mode shown is the one desired, press the YES softkey. If not, press the NO softkey and the other alternative is displayed; then press the YES softkey. To leave the system in its current mode, press SUPERKEY.
- Press SUPERKEY.

***SUPERSET 430* Telephones:**

- Press the SUPERKEY.
- Press the MORE softkey until the NIGHT ANSWER softkey appears.
- Press the NIGHT ANSWER softkey. The top display line indicates which mode is currently active (e.g., CURRENTLY DAY SERVICE). Softkeys appear for the two alternatives.
- Press the softkey for the desired mode; or, to leave the system in the current mode, press the BACKUP softkey or the SUPERKEY.

Night Services

Description The PABX has three different service modes: DAY, NIGHT1, or NIGHT2. When the PABX or tenant group is in night service mode, incoming trunk calls and calls to the attendant may be rerouted to specified extensions or be caused to activate common alerting devices (Night Bells).

Conditions The following conditions apply to this feature:

- This feature is available on a per tenant basis.
- Some features, such as Alternate Recall Point, Attendant Call Forward-No Answer, and Call Rerouting, operate differently under NIGHT1 and NIGHT2.

Programming CDE Form 14 (Non-Dial-In Trunks) and CDE Form 19 (Call Rerouting Table) programming is directly affected by Night Services.

Operation See *Attendant Night/Day Switching, Night/Day Switching, Call Rerouting, Trunk Operation - Non-dial-in CO, Attendant Access (Dial 0).*

Night Services Flexibility

Description This option allows the attendant to change the night service assignment of non-dial-in trunks. The system allows full flexibility of trunk assignment.

Conditions The following conditions apply to this feature:

- The console must be able to connect to the trunk (see *Device Interconnection Control).*
- CDE is prevented from accessing Form 14 (Non-dial-In Trunks) when the console is updating the night point of a trunk. Similarly, the console is prevented from changing the night service while CDE is accessing Form 14.
- The rules for the assigned access codes are the same as in CDE.

Programming Enable COS Option 104 (Attendant Flexible Night Service Setup) for the Console.

Operation Change the night service assignment of non-dial-in trunks from the console, as follows:

- Press FUNCTION.
- Press ATT. FUNCTION.
- Press MORE.
- Press FLEX NIGHT.

The display prompts for trunk number.

- Enter trunk number.
- Press either NIGHT1 or NIGHT2.
- Enter new destination.
- Press SET.

The trunk is now routed to the new destination when the PABX is in Night 1 or Night 2 (as programmed above).

Node Identification

Description The node identification feature works with the analog networking feature to provide consistent dialing of extension numbers throughout a network of *SX-200* PABXs.

For any extension, the node identification digits plus the extension number uniquely identifies the extension from all others on the network. The extension can be reached by dialing the same string of digits from any node in the network.

For the use of the node identification code in analog networking, see *Analog Networking* in this section.

Conditions The following conditions apply to this feature:

- The node identification code must match the ARS leading digits specified in the other nodes in the network to reach that node.
- Dial tone is not returned after the node identification code is dialed.
- Any device may access the node identification code.
- Within any node, it is not necessary to dial the Node Identification digits to access extensions within that node.

Programming In CDE Form 02 (Feature Access Codes) assign an access code to Feature 34 (Node ID).

Operation Dial the Node Identification code and then the extension number.

Non-Busy Extensions

Description

An extension with the non-busy extension feature enabled never appears busy to the PABX. If a new call is directed at a non busy extension that is already in a call, the PABX automatically overrides the existing call. After a warning tone, the new caller joins the conversation.

Conditions

The following conditions apply to this feature:

- The Override Security and Data Security features are ignored for the parties in the established call.
- The feature does not work for members of recording hunt groups.
- The feature is ignored when the console calls the set.
- The same conditions for override and key lines apply.
- This feature conflicts with any feature requiring a switch–hook flash.
- The non-busy extension can be in conversation with no more than four other parties. Additional callers receive busy tone.
- Only calls dialed directly to the set override; Recalls, Call Reroutes, Callbacks, ignore the feature.
- A call transferred to a non-busy extension does not override a conversation in progress.

The override is disallowed under the following conditions:

- if the caller has a consultation hold in progress and any party in the call has a consultation hold in progress.
- if a console is in the call already.
- if the established call is a three party call that is on consultation hold.
- if one of the parties in the call has the call on hold.
- if the set is overriding another extension or had called another non-busy extension.

- The feature is not available to CO trunks when they originate.
- If the override is not permitted, then normal busy destination processing results.
- No connection checks are done when the caller is added into the call.
- Once on a call with a non-busy extension, regardless of how the call was established, no party in the call except the non-busy extension can flash out of the call or put it on hold.

Programming

Enable COS Option 243 (Non-Busy Extension) for the extension.

Operation

A new incoming call joins in to an established call automatically (with a burst of warning tone).

Numbering Plan Flexibility (Conflict Dialing)

Description The numbering plan used within the system is completely flexible. The user may select any combination of 1-, 2-, 3-, 4- and 5-digit numbers.

Also see *Conflict Dialing.*

Conditions Leading digits of extension numbers must not match feature access codes except codes 20 (Call Back Busy), and 31 (Executive Busy Override).

Programming Assign the required extension numbers.

Operation None.

Off-Premise Extension (OPS)

Description Industry-standard telephones not in the immediate vicinity of the PABX can be directly connected to the PABX without the use of special trunks, using the 6-circuit OPS Line Card (Off-Premises). Refer to the *Engineering Information Practice*.

Conditions The following conditions apply to this feature:

- If the extension is located more than 2 kilometers away, it may be necessary to add a compromise balance network into the circuit. COS Option 402 (Long Loop - Off-Premise Extensions Only) does this.
- An adjustment is made to the Loss/Gain settings for the line using COS Option 402 (Long Loop - Off-Premise Extensions Only).
- There is no equivalent feature for *SUPERSET* telephones.
- COS Option 402 (Long Loop - Off-Premises Extension Only) applies to OPS Line Card circuits only.

Programming Program the OPS Line Card in CDE Form 01 (System Configuration).

If necessary, enable COS Option 402 (Long Loop - Off-Premise Extensions Only) for the extension.

Operation None.

Originate Only Extensions

Description This feature allows an extension or dataset to originate calls. The extension can only receive calls that are forwarded from another extension. The system treats calls dialed to originate only extensions as illegal numbers.

Conditions The following conditions apply to this feature:

- An extension with this COS option may receive calls via Call Forwarding (unless COS Option 234 (Never a Forwardee) is selected in its COS).
- Only calls directly dialed to an Originate Only extension are invalid.
- COS Option 235 (Originate Only) is mutually exclusive with COS Option 300 (Automatic Callback), COS Option 241 (Receive Only), and COS Option 243 (Non-busy Extension).

Programming Enable COS Option 235 (Originate Only) for the extension.

Operation None.

Overlap Outpulsing

Description Overlap outpulsing occurs when the PABX begins dialing on a trunk before the user has dialed all digits in the destination's telephone number. By default, the ARS package outpulses digits as soon as the trunk seizure is acknowledged. This provides a shorter total dialing time, especially on non-DTMF trunks. This feature can be turned off, forcing the ARS package to collect all dialed digits before outpulsing the resulting digit string on the outgoing trunk. Refer to the *Automatic Route Selection and Toll Control Practice* for more information.

Conditions None.

Programming Enable System Option 26 (No Overlap Outpulsing) to disable the feature.

Operation None.

Override (Intrude)

Description This feature allows a user who encounters a busy extension to enter the conversation. Before override voice contact is established, the overriding party and both parties in the original conversation receive a warning tone. The tone is repeated at regular intervals while the overriding party is connected to the existing call. Display *SUPERSET* telephones display the name and/or extension number of the overriding party.

For override of a telephone with Do Not Disturb set; see *Do Not Disturb.*

Conditions

The following conditions apply to this feature:

- If there is a call-me-back message outstanding between the overriding party and the overridden party, it is canceled. This is not done if the console overrides.
- The overriding party cannot manipulate the original connection in any way.
- If the overridden extension puts the call on consultation hold or goes on–hook, the intruding extension is dropped and receives reorder tone. Display telephones will read DISCONNECTED.
- If *SUPERSET* telephones involved in a conversation have multicall appearances, an override only occurs if all appearances are busy. Otherwise, the calling extension receives ringing and the idle multicall appearance rings if programmed to do so.
- The warning tone is given to all internal parties in the call being overridden.
- If the overridden call collapses to a two party call with just the overriding and the overridden party, then the override condition is cleared and a normal two-party call continues.
- If a key line is overridden, if the root extension of the line is in the call on the line then that telephone is overridden. If the extension is not in the call on the line then the first "I-busy" *SUPERSET* telephone using one of the key line appearances is overridden.
- If a logical key line is overridden then the first "I-busy" *SUPERSET* telephone using one of the line appearances is overridden.
- An extension can only be overridden when talking to a party or in a conference call.
- If the caller is prevented from overriding when the override is attempted, the caller receives reorder tone.
- If the extension called becomes idle while the caller is listening to busy tone, when the caller attempts to override, an override is not done. Instead, a new call attempt is made to the extension as if the extension was dialed again.

The user cannot override:

- a telephone on hold,
- an industry-standard telephone, *SUPERSET* telephone, DISA trunk, or TIE trunk, with COS Option 216, Data Security or COS Option 238, Override Security in its COS,
- an extension with a call on consultation hold and the extension does not have the non-busy extension feature enabled (see *Non-busy Extensions),*
- a consultee,
- a busy hunt group,
- a logical multicall line,
- an industry-standard telephone key line that is holding a party (but is not holding the party on the key line),
- a conference call on consultation hold,
- a 4-party call with one party flashed out of the conference,

- a call that has another party in the call with the Non-busy Extension feature enabled,
- a call with the attendant in it,
- a call with five parties in it,
- an extension that is overriding another extension,
- a member of a recording hunt group.

Programming

To allow extensions in a COS to override, enable COS Option 500 (Override) in that COS.

Industry-standard telephones, *SUPERSET 401+*, and *SUPERSET 410* telephones need an access code to perform Override. Assign an access code to Feature 31(Executive Busy Override). This access code must be a single digit.

To provide feature key activation of Override from a *SUPERSET 410* or a *SUPERSET 420* telephone, program an OVERRIDE (INTRUDE) feature key. (See *Feature Keys*.)

Operation

Operation varies depending upon the type of set as described below.

Industry-standard and *SUPERSET 401+* Telephones:

- While receiving busy tone, dial the Executive Busy Override access code. After the warning tone, you are connected to the call.

***SUPERSET 410* and *SUPERSET 420* Telephones:**

- While receiving busy tone, press the INTRUDE feature key. After the warning tone, you are connected to the call.

***SUPERSET 430* Telephones:**

- While receiving busy tone, press the INTRUDE softkey; this softkey appears only when override is permitted.
- After a warning tone, you are connected to the call.

Override Security

Description

This option provides an extension, DISA trunk, or Dial-in Tie trunk with security against Override (Intrude); see *Attendant Busy Override* and *Override (Intrude)*.

Conditions

The following conditions apply to this feature:

- The feature is ignored by the non-busy extension feature.
- The feature also provides an extension with security against having Do Not Disturb overridden; see *Do Not Disturb.*

Programming

Enable COS Option 238 (Override Security) for the extension or trunk.

Operation

None.

PA Paging

Description

An extension, Tie trunk, or DISA trunk can be permitted to access the paging equipment by dialing the required access code. Access may be restricted to any of the nine zones depending upon the access code dialed. If an extension tries to access busy paging equipment, busy tone is returned.

Paging equipment may be connected to an *SX-200* ML *via* a Music-on-Hold/Pager module on the Universal Card or via a DNIC Music-on-Hold/Pager Unit. Up to nine paging zones, with separate or simultaneous access, can be provided.

Also see *Attendant Paging Access.*

Conditions

The following conditions apply to this feature:

- A short burst of tone is provided when paging equipment is successfully accessed.
- Campon or Callback may not be activated on busy paging equipment.
- Paging amplifiers and loudspeakers are customer-provided equipment.
- The attendant can override any extension using paging equipment to make an announcement. The extension receives busy tone when it is removed from the pager.
- If System Option 03 (Single Paging Amplifier) is enabled, only one user at a time can access all paging in the system and the attendant cannot override the parties on the pager.
- Accessing the pager operates a relay on the pager module which may be used for controlling an external paging amplifier.
- *SUPERSET* telephone users cannot access paging while in handsfree mode.
- If access to all zones is attempted and at least 1 zone is busy then no paging is done and busy tone is returned.
- Access is to a single zone or simultaneously to all zones programmed in the device's COS.
- The PAGE softkey is not available to *SUPERSET 430* telephones with the HEADSET OPERATION feature unless there is a consultation hold in progress.
- For other *SUPERSET* devices in HEADSET OPERATION, the *SUPERSET* must first enter dialing state (by pressing a line key, and getting dial tone) before the PA PAGE feature key will be presented.

Programming

Program a Music-on-Hold module in CDE Form 18 (Miscellaneous System Ports).

If the DNIC Music-on-Hold/Pager Unit is being used, program the unit as follows:

- Program a DNIC Module in Form 1.
- Program the DNIC device type as "DMP" (DNIC Music/Pager) in Form 9.

Enable System Option 03, Single Paging Amplifier, if the PABX has only one paging amplifier output.

Enable one or more of the following COS Options for the extension in CDE Form 3 (COS Options) as shown in the following table.

COS Option Number	Description
303	Paging Zone 1 Access
304	Paging Zone 2 Access
305	Paging Zone 3 Access
306	Paging Zone 4 Access
307	Paging Zone 5 Access
308	Paging Zone 6 Access
309	Paging Zone 7 Access
310	Paging Zone 8 Access
311	Paging Zone 9 Access
312	Paging Default (0 to 9) (0 Gives All Enabled Zones)

For access to the default zone, assign an access code to Feature12 (Paging Access to Default Zone) in CDE Form 02 (Feature Access Codes).

Assign an access code to Feature 13 (Paging Access to Specific Zones) for access to zones other than the default zones.

To provide convenient access to this feature from a *SUPERSET 410 or SUPERSET 420* telephone, program a PA PAGE feature key (see *Feature Keys*).

Operation

Operation varies depending upon the device as described below.

Industry-standard and *SUPERSET 401+* Telephones:

- Lift the handset.
- Dial the appropriate paging access code.
- If access code is for specific zone paging, dial the zone number (0 through 9).
- A tone is returned. Make announcement.
- Hang up.

***SUPERSET 410* and *SUPERSET 420* Telephones:**

Note: If a PA PAGE feature key is not programmed for the set, follow the Industry-standard telephone operation.

- Go off-hook.
- Press and hold down the PA PAGE feature key for default zone access.
- Wait for a short burst of tone.
- Make the required announcement.
- Hang up.

SUPERSET 430 **Telephones:**

- Go off-hook.
- Press and hold down the PAGER softkey for default zone access.
- Wait for a short burst of tone.
- Make the required announcement.
- Hang up.

Parallel Connection of Industry-standard Telephones

Description A maximum of three industry-standard telephones equipped with bells can be connected (hard-wired) together on one ONS line.

Conditions The following conditions apply to this feature:

- When one industry-standard telephone is in use, if any other of the industry-standard telephones goes off–hook, it then joins in the call (without proper conferencing facilities).
- All of the industry-standard telephones ring when the extension number is called.
- If the telephones are equipped with Message Waiting lamps, further restrictions may apply.

Programming None.

Operation None.

Pickup Groups

Description Extensions can be programmed into pickup groups, permitting users to answer calls on any other extension within their particular group; see *Pickup - Local and Directed.*

Conditions The following conditions apply to this feature:

- A maximum of 50 Pickup groups are permitted per system, with a maximum of 50 extensions permitted per group.
- Calls are picked up in the group in order of the extensions in the pickup group. The search for a ringing extension starts with the first extension in the group and ends with the last in the group.
- An extension can only be in one pickup group at a time.

Programming Assign the desired extensions to the appropriate pickup groups via CDE Form 10 (Pickup Groups).

Operation See *Pickup - Local And Directed.*

Pickup - Local and Directed

Description

A telephone can be assigned to a pickup group, and can answer any ringing telephone within that group. This is Local Pickup.

Directed Call Pickup allows an extension user to answer any ringing telephone within the system.

See *Pickup Groups.*

Conditions

The following conditions apply to this feature:

- Local Pickup operates only for extensions within the same pickup group.
- For ringing *SUPERSET* telephones, the ringing line is scanned for from the prime line up the *SUPERSET* telephone until a line that can be picked up is found.
- For Directed Call Pickup, the scan is done on the extension specified and not on the line appearances of that extension (unless the *SUPERSET* telephone has multicall line appearances of itself).
- For pickup groups, the scan for a ringing extension always starts with the first pickup group member (terminal scanning) each time a pickup is attempted. A ringing extension is determined to be ringing based on the scan on the extension for a ringing line that can be picked up.
- Reorder tone is returned if there is no call to pick up (or it is already picked up).
- The following call types cannot be picked up:
  - callbacks,
  - wakeup/reminders,
  - calls to members of recording hunt groups,
  - calls ringing back an extension (Station Transfer Security),
  - Direct Trunk Select or Private Trunk lines,
  - silently ringing line appearances or delay ring line appearances that have not begun to ring,
  - console calls when a consultation hold is in progress,
  - callers that cannot connect to the party picking up the call (see *Device Interconnection Control),*
  - the party picking up the call and the ringing extension cannot connect,
  - the *SUPERSET* telephone is Auto-Answering the call.

Programming

Program Pickup Groups in CDE Form 10 (Pickup Groups).

Assign an access code in Form 02 to Feature 08 (Dial Call Pickup).

Assign an access code in Form 02 to Feature 09 (Directed Call Pickup).

To provide convenient access to this feature from a *SUPERSET 410* telephone, program a PICKUP feature key (see Feature Keys).

To permit sets in a COS to pick up calls outside their Pickup Group, enable, in Form 03, COS Option 218 (Directed Call Pickup) in that COS.

Features Description

Operation

Local Pickup - Industry-standard and *SUPERSET 401+* Telephones:

When a telephone in the same pickup group rings:

- Lift the handset - dial tone is returned.
- Dial the Call Pickup code - the call is connected.

Local Pickup - *SUPERSET 410* Telephones:

When a telephone in the same pickup group rings:

- Lift the handset - dial tone returned.
- Press the PICKUP feature key - the call is connected.

Local Pickup - *SUPERSET 420* Telephones:

When a telephone in the same pickup group rings:

- Lift the handset - dial tone returned.
- Press the PICKUP softkey - the call is connected.

Local Pickup - *SUPERSET 430* Telephones:

When a telephone in the same pickup group rings:

- Lift the handset - dial tone returned.
- Press the PICKUP CALL softkey - the call is connected.

Directed Pickup - All Telephone Types

When a telephone outside the pickup group rings:

- Go off–hook - dial tone is returned.
- Dial the Directed Call Pickup code.
- Dial the extension number of the ringing telephone - the call is connected.

Power Fail Transfer (PFT)

Description

In the event of a common control or power failure which would cause a major loss of call processing, preselected CO trunks are automatically switched to designated extensions. When normal system operation is restored, calls on the transfer circuits remain in effect until they are terminated.

Generally, the circumstances which cause a power fail transfer are:

- The main control card stops functioning.
- The link between the main controller and the equipment bay stops functioning (the system is cut over into PFT mode).
- On power-up, the system bay fails to initialize properly (the system is cut over into PFT mode). PFT follows a Critical Alarm.
- Commercial power failure with no PABX backup power source (UPS).
- Bay Power Supply failure.

Conditions

The following conditions apply to this feature:

- If a transfer takes place, any existing calls on the transferred trunks are dropped.
- If trunks are rotary dial only, DTMF sets may not be used for dialing.
- *SUPERSET* telephones and consoles cannot be Power Fail Transfer extensions.

Programming

None.

Operation

None.

Printer / Terminal Support

Description

This feature allows the routing of printouts to the system printer port, to any data port, or to the printer port on the *SUPERCONSOLE 1000* Attendant Console. If no new port is specified through CDE, printouts default to the system printer port. All printer ports are RS-232C interface. Printout types include:

- Traffic measurement
- SMDR (Trunk, Data, ACD)
- CDE
- Hotel/Motel system printouts
- PMS interface port
- ACD reports
- Maintenance logs.

Customer programming printouts may be directed to any or all of seven user-defined printers. A maximum of six DNIC-based printer ports can be defined; the remaining port is the system printer port.

CDE forms may be printed individually or collectively.

Certain CDE forms require the user to specify which sub-forms are to be printed (in a "from - to" format).

Conditions

The following conditions apply to this feature:

- Printouts may be sent to more than one printer at a time. The time required to print is determined by the slowest printer.
- Not more than five printouts may be directed to one printer.
- Some printouts can be guaranteed. If there is a printer failure, the information is preserved and printed when the printer is ready again.
- If the printer runs out of paper and can send flow control information, the system suspends printing until the printer is ready again.
- If the printer or dataset is powered down, the system waits until the printer or dataset is ready again.
- Printers must have an RS-232C interface.

- A functioning printer must always be connected to the port assigned for SMDR printouts. If the printer fails or is disconnected, outgoing trunk calls are disabled as soon as the internal storage buffer is full if 'No Overwrite' (COS Option 702) is enabled in the COS for the trunk. The system's internal SMDR record buffer holds 200 SMDR records; this provides sufficient time for printer maintenance.
- The PMS printer port must be dedicated to PMS only; it cannot be shared with any other application.
- The *SUPERCONSOLE 1000* Attendant Console printer port limitations are described in the *Peripherals Devices Practice.*

Programming For a printer connected via a dataset, define an appropriate Data Circuit Descriptor in CDE Form 11 (Data Circuit Descriptor). Program the dataset in Form 12 (Data Assignment) using the new Data.

Complete programming of CDE Form 34 (Directed I/O); specify the printer ports available, and then direct each printout to its associated printer.

After assigning the *SUPERCONSOLE 1000* Attendant Console in CDE Form 07 (Console Assignments), program subcircuit 2 of the console's Bay/Slot/Circuit location as a DSCONS in CDE Form 12 (Data Assignment), if the console port is to be a printer port.

Note: Printer connected to a printer port must have parameters set up as shown in the *RS-232 Maintenance Terminal Practice.*

Operation The system checks CDE Form 34 (Directed I/O) whenever it generates a printout, to determine to which printer port it should route the printout.

Priority Dial 0

Description Priority Dial 0 is a second class of Dial 0 call, with its own separate DAY/NIGHT routing points. This feature can be used to provide an alternate Dial 0 routing for extensions in the system.

Conditions The following conditions apply to this feature:

- Priority Dial 0 calls can be routed to the same type of answering points as Dial 0 calls.
- Wherever Dial 0 routing is used in the system, Priority Dial 0 is checked.
- Priority Dial 0 calls may be routed to the same point as Dial 0 calls or they may be routed to an LDN key which distinguishes Dial 0 and Priority Dial 0 calls.

Programming Enable COS Option 239 (Priority Dial 0) for the required extension.

Program an Access Code (usually 0) in CDE Form 02 (Feature Access Codes) for Feature 11 (Extension General Attendant Access).

Operation Dial the access code.

Privacy Enable/Privacy Release

Description

A *SUPERSET 410, SUPERSET 420,* or *SUPERSET 430* telephone may have line appearances of key, direct trunk select, CO line and private trunk lines that are shared with other sets. When privacy is enabled, while a conversation is in progress on a line, other sets with an appearance of that line are denied access. See *Line Privacy.* The user of the line can, however, use the Privacy Release feature to allow the other sets to join the conversation.

Conditions

The following conditions apply to this feature:

- Privacy is effective only against other appearances of the line; it has no effect on override.
- Privacy Release is available only on *SUPERSET 410, SUPERSET 420,* and *SUPERSET 430* telephones.

Programming

To allow Privacy Release operation at a *SUPERSET 410* or *SUPERSET 420* telephone, program a PRIVACY RELEASE feature key (see *Feature Keys*).

Operation

***SUPERSET 410* and *SUPERSET 420* Telephones:**

- During an established call, press the PRIVACY RELEASE feature key. The LCD shows the status of privacy on the line as follows:
  - LCD clear indicates Private.
  - LCD dark indicates Not Private.
- Another telephone user with an appearance of the same line can now enter the conversation by pressing the appropriate Line Appearance key.
- Reestablish privacy by pressing the key again.

***SUPERSET 430* Telephones:**

- During an established call, press the REL PRIVACY softkey. Another telephone user with an appearance of the same line can now enter the conversation by pressing the appropriate Line Select key.
- Reestablish privacy by pressing the MAKE PRIVATE softkey.

Programmable Key Module (PKM)

Description

A Programmable Key Module (PKM) provides *SUPERSET 410, SUPERSET 420,* and *SUPERSET 430* telephones with 30 additional personal keys. You can program these personal keys as

- speed call keys,
- feature keys,
- key line appearances,

- personal outgoing line keys,
- key system appearances,
- multicall line appearances,
- co line keys, or
- busy lamp field/direct trunk select keys.

The keys are arranged in two vertical rows on the module. Beside each key is a Line Status Display that indicates the status of the key. The flash rates for the Line Status Displays on the PKM are identical to those on the *SUPERSET 410*, *SUPERSET 420,* and *SUPERSET 430* telephones.

Up to three PKM modules connect directly to the set.

Each *SUPERSET 410*, *SUPERSET 420,* and *SUPERSET 430* telephone has one *MILINK* port. Each PKM has two *MILINK* ports. The first PKM connects to the *MILINK* port located on the base of the telephone. The other two PKMs are connected in series (i.e., daisy-chained together) to the first. You can also connect one *MILINK* Data Module anywhere in the chain of PKMs. For installation instructions refer to the *Peripheral Devices Practice.*

SUPERSET 410, *SUPERSET 420,* and *SUPERSET 430* telephone users can program any PKM personal key that isn't assigned to a line appearance. Users can program personal keys as feature keys from their set. Refer to *Feature Keys* for instructions.

If you print Form 09 (Station/*SUPERSET* telephones) or Form 45 (Key System Telephones) from Form 32 (Customer Data Print), any PKM key line appearances are printed following the associated set key line appearances. If you have more than one PKM programmed with key line appearances, the keys are printed in the order of the address settings. If a PKM is not programmed with any key line appearances, "UNPROGRAMMED PKM" is printed following the PKM's address setting.

Conditions

- A *SUPERSET 410* with three PKMs provides up to 96 key line appearances. A *SUPERSET 420* or *SUPERSET 430* with three PKMs programmed provides up to 102 key line appearances.
- You can connect a *MILINK* Data Module to the *MILINK* port on one of the telephone's PKMs (one *MILINK* Data Module per telephone set).
- The PKM has rocker dip switches located on the base of the set. You must set these switches to identify the address (1, 2, or 3) of the PKM. Each PKM must have a different address. For information on how to set the PKM addresses, refer to the *Peripheral Devices Practice.*

Programming

To program a telephone or key system telephone to support a PKM:

- Display CDE Form 09 (Stations/*SUPERSET* Telephones) or CDE Form 45 (Key System Telephones).
- Move the cursor to the bay/slot/circuit number of the *SUPERSET 410*, *SUPERSET 420*, or *SUPERSET 430* telephone. An asterisk (*) appears to the left of the set type (410, 420, 430) if PKMs are programmed for the set.

- Select the EXPAND PKM softkey. "ENTER PKM NUMBER (1-3):" appears in the command line.
- Type the address setting (1, 2, or 3) of the next PKM that you can add.
- Select the ENTER softkey to display the Expand PKM Set Subform. The Expand PKM Set Subform allows you to program the functions of the 30 available PKM keys. Keys 31 and 32 in the form are listed as RESERVED since the PKM only has 30 keys.

To program the PKM keys:

- Move the cursor to the desired key. Figure 2-2 shows how the keys are numbered on a PKM.
- Move the cursor to the TYPE column (use the TAB key if necessary).
- Select the key type using the softkeys on the Expand PKM Set Subform.
- Press the SAVE softkey.

  Key numbers are preceded by 1, 2 or 3. The 1, 2 or 3 indicates the order in which you associated the PKM with the set (e.g., the keys belonging to the first PKM that you associated with the set will be preceded by a "1"). The 1, 2 or 3 is not related to the PKM address settings.

  To program the PKM keys of a *SUPERSET* telephone refer to "Expand PKM Set Subform for Form 09" in the *Customer Data Entry Practice* for definitions of the available key types. If you are programming the PKM keys of a Key System Telephone, refer to "Expand PKM Set Subform for Form 45" in the *Customer Data Entry Practice*.
- Users can also program any PKM personal key that isn't a line appearance as a feature key from their sets (refer to *Feature Keys* for instructions).

When deleting a set that has PKMs:

- You cannot delete a set from CDE Form 09 (Stations/*SUPERSET* Telephones) or from CDE Form 45 (Key System Telephones) until you delete all of the set's PKM keys. For each PKM assigned to the set, you must access the Expand PKM Set Subform and delete the PKM keys.

To delete a PKM:

- Delete all of the keys from Form 09, Expand PKM subform.
- Press the DELETE softkey.

Operation

For instructions on using the features associated with PKM personal keys, refer to the specific feature.

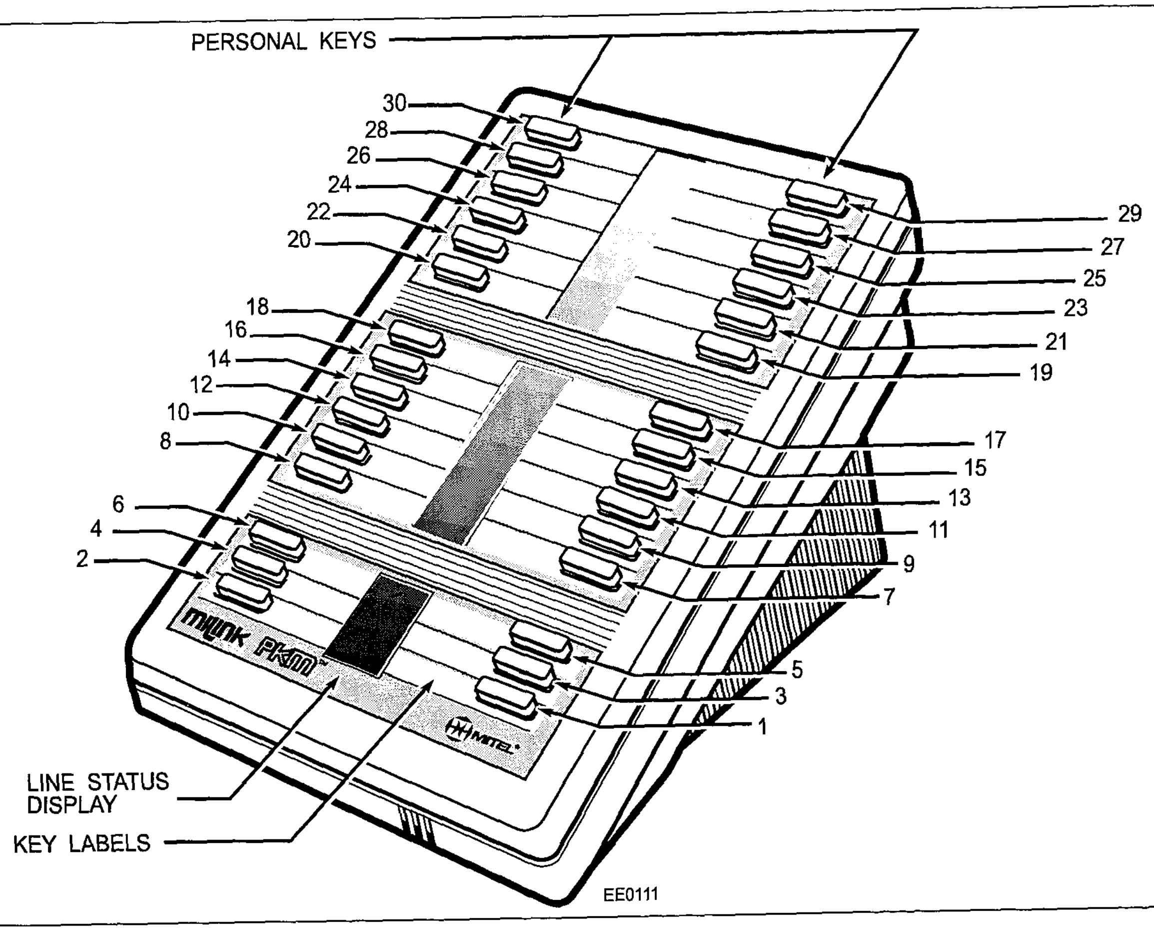


Figure 2-2 PKM Key Numbering

RAD Support

Description

Recorded Announcement Devices (RAD) are supported in the system as recording hunt groups. These special hunt groups have features and restrictions on them that allow efficient use of the recording resources. Recording hunt groups are used in ACD, UCD, Hotel/Motel Wakeup, Automatic Attendant Overflow and Automated Attendant.

For ACD, Attendant Automatic Overflow, and Automated Attendant, more than one caller at a time (a listen only conference) can listen to a recording in the recording hunt group. For UCD and Hotel/Motel Wakeup, only one caller at a time can listen to a recording in the recording hunt group.

For some applications, there is support for various kinds of recording devices and support for various recording device failures. These features only apply when the members of the group are accessed through the group - they do not operate when an individual member of the recording hunt group is called directly by dialing its individual extension number.

When a recording is playing for ACD, Automatic Attendant Overflow, and Automated Attendant, the Recording Message Length Timer (programmed on the recording group) applies. When the timer expires and the recording is still playing, the system terminates the recording. For some devices this acts as a safety timer for recording devices that fail to terminate their recording. For tape based recording devices, this allows the system to not play the dead time at the end of the tape.

When an error occurs, a maintenance log is created and the failed recording device is put into the Do Not Disturb state. There are two error conditions handled for members of recording groups:

- When a RAD does not answer a call within 20 seconds, the ringing is canceled and the recording is put into Do Not Disturb.
- When the system ends the recording, due to the message length timer or the caller hanging up, then the Recording Failure To Hangup Timer (in the recording device's COS) is started (not for UCD or Hotel/Motel Wakeup recordings). If the recording fails to hang up during this time then the recording is put into Do Not Disturb.

See *Attendant Automatic Overflow, Automated Attendant, Uniform Call Distribution (UCD)* and *Reminder.* Also see *Automatic Call Distribution* and refer to the *ACD TELEMARKETER Application Package Practice.*

Conditions

The following conditions apply to this feature:

- Hunt group overflow does not apply to recording hunt groups.
- Industry-standard telephones only can be members of recording hunt groups.
- There is no limit to the number of simultaneous listeners to a recording.
- Recalls never occur to recordings that are not directly dialed.
- Trunks (all types) are answered before listening to a recording.

The following restrictions apply to members of recording hunt groups:

- they cannot have line appearances of them programmed on any *SUPERSET* telephone in the system
- they cannot originate calls (regardless of any options in the system)
- a flash is always ignored
- they cannot have call me back messages left at them
- they cannot be overridden
- campon warning tone is never applied
- they cannot be an alternate music source port in ACD
- calls ringing them cannot be picked up using the Directed Pickup feature
- the Last Party Clear Dial Tone feature does not operate for recording devices
- standard ringing is always applied when ringing.

Programming

For recording hunt groups used in Automatic Call Distribution, Attendant Automatic Overflow, and Automated Attendant, select a time for COS

Option 404 (Recording Failure to Hangup Timer) in the COS of the recording hunt group members.

Since some RADs won't answer calls that use discriminating ringing (non-standard ringing), ensure that COS Option 809 (Standard Ring Applies) is enabled in the COS of the incoming trunks.

See *CDE Form 17 (Hunt Groups)* for details on programming the hunt group itself.

Operation Not applicable.

Recall

Description The recall feature ensures that calls do not remain unanswered or on hold for an extended period. Any call that has been extended by a console, or an external call that has been extended by an extension to another party, recalls the console or extension if the call is not answered or remains on hold at the end of a timeout period.

The recall feature also works for outgoing external calls. When a trunk is seized, the calling party becomes the recall point. If the trunk is transferred somewhere in the system, recall is by default to the party that made the call.

Setting Up the Recall Point to consoles

When a console answers a call, regardless of where the call came from, that caller's recall point is always set to be that console. The call recalls back to the console (at the Recall call position) unless some other recall feature has been added.

Performing an unsupervised transfer of a call to another party in the system does not change the recall point of the transferred party. For a supervised transfer, the recall point of the transferred party (if it is an internal party) is cleared, which prevents any recall back to the console. If the console is transferring a non-serial trunk to a subattendant extension, the recall point is set to be the subattendant.

A console also becomes a trunk's recall point if the console ever becomes alone in a conversation with the trunk. This could occur under three circumstances:

- an extension does a supervised transfer of a trunk to a console, and hangs up while the trunk is still softheld.
- an extension does a supervised transfer of a trunk to a console, the three are conferenced by the console, the transferring extension then hangs up.
- a console transfers a trunk to another console, and hangs up while the trunk is softheld.

Setting Up the Recall Point to extensions

Extensions only affect the recall point of trunk calls when the trunk is answered (this includes answering calls via features such as Auto-Answer, directed pickup or TAFAS). The recall point is only set to the answering extension if the trunk call is not a serial trunk and the trunk has not had a recall point set up already. If the extension is a subattendant, and the trunk is not a serial call trunk, then the trunk recall point is changed even if it is already set-up.

When any party performs a Hold Pickup Access (see *Attendant Paged Hold Access*) from an Attendant Hold Slot (Feature Access Code Number 16) it becomes the new recall point for the picked up party, provided that the picked up party was not a serial call trunk.

The recall feature works for outgoing external calls as well. When a trunk is seized, the recall point is set to be the calling party. If the trunk is transferred somewhere in the system, recall is by default back to the party that made the call unless the recall point is altered.

When A Recall Is Done

Recall is done when ringing a device, camping on to a device, or being held by a device.

When ringing another device in the system and forwarding is done, then the Recall No Answer Timer is started for the device and when the timer expires, a recall is attempted. If calling an LDN and the Attendant Calls Forwarded On No Answer feature is active then the timer for that feature is started instead; see *Attendant Calls Forwarded On No Answer.*

When camping on to another device, the attendant campon recall timer is started. When the timer expires, a recall is attempted.

When a call is held by an industry-standard telephone after the call hold Recall timer timeout, (COS Option 254), a recall is attempted if the industry-standard telephone is not idle.

Recall Processing

When a recall is attempted, the following decisions are made:

- if ringing a console, LDN, or Night Bell, then no recall at all is done.
- if the caller is a CO or DISA trunk and the Non Dial-in Alternate Recall Point is defined and the trunk is not already calling that point then a reroute is done to that point; see *Alternate Trunk Recall.*
- if a DID or Tie trunk is ringing a party, then DID/Tie Trunk Routing On No Answer is checked. This is only done if the recall point is not set up for the caller; see DID/*Dial-in/Tie Intercepts.*
- if the recall point has not been set up, then the Final Ringback Timer is started if the caller is recalling from ringing. Otherwise recall is attempted again after 10 seconds.
- if the recall point is set up and is not busy, then that point is called.

- if the recall point is set up and is busy, and if recalling from ringing, then the Recall No Answer Timer is started again. Otherwise, recall is attempted again after 10 seconds.

Conditions

The following conditions apply to this feature:

- Ringing an extension from a hold timeout for *SUPERSET* telephones or industry-standard telephones that are idle or have Do Not Disturb activated is handled as a recall after a ring no-answer timeout.
- Unlike a reroute point, a busy recall point does not have Recalls camp onto it. If the recall point has Do Not Disturb activated then no recalls to the point are done.
- A *SUPERSET* telephone with the Subattendant - Basic Function feature has less stringent tests for being available to handle a recall; see *Subattendant.*
- The serial call feature is similar to the basic recall in that it sets up the recall point for the serial trunk; see *Attendant Serial Call.*
- Recall occurs from campon after the Attendant-Timed Recall (CAMPON) time. The timer value is taken from the caller's recall point's COS. If no recall point has been setup then the timer value is taken from the caller's COS. If the timer value is 0 in the caller's COS then the final ringback timer is started and no recall is done.
- For a recall from campon to the console or an LDN key, the recalling party is not removed from campon. If the busy destination becomes available before the recall is answered at the console then the recall stops and the recalling party rings the now available party.
- For recall from ringing, the Attendant-Timed Recall (NO ANSWER) timer value is taken from the caller's recall point's COS. If no recall point has been set up then the timer value is taken from the caller's COS. If the timer value is 0 in the caller's COS then the Final Ringback Timer is started and no recall is done.
- When a recall timer is needed for the recall point, if the recall point is an LDN key then the COS of the console with the lowest bay/slot/circuit PLID where the LDN is programmed is used.
- No recall is done when a device is receiving busy tone and it does not camp on to the busy device.
- No recall of any kind is done for Direct Trunk Select and private trunks that are ringing in.
- Listening to recordings has no effect on recall for Uniform Call Distribution, Automated Attendant, Attendant Automatic Overflow, and Automatic Call Distribution callers.
- When a reroute is performed for UCD Busy Agent timeout, the recall point for the waiting caller is cleared.
- Whenever an internal caller talks to another device, the recall point of the internal caller is cleared.
- Enabling repeated campon beeps for a trunk prevents recall from campon; see *Campon Warning Tone.*
- The Auto-Answer feature is ignored for recalls directly back to a *SUPERSET* telephone.

- See *Attendant* and *Subattendant Hold Positions* and *Hold* for details on Hold Timeout handling.
- A console never recalls to any other device.
- A call recalls to extensions and consoles through a recall point. Recall to other device types must use features available through the Call Rerouting table.
- When a *SUPERSET* telephone answers a call on a non-prime line, the recall point for the caller, if it is altered, is set to the answering *SUPERSET* telephone and not to the line that was answered.
- Recalls to a console recall call position are directed to the Recall call position of a group of consoles if the Transparent Multi-Console Operation feature is used.
- Subattendant Recalls are not affected by Transparent Multi-Console Operation feature.

Programming Select values for COS Options 117 (Attendant-Timed Recall - CAMPON), 115 (Attendant-Timed Recall (NO ANSWER) and System Option 51 (Final Ring Timeout) for the recall point, console, or extension, or in the caller's COS (see Conditions to determine which COS is used); see *Attendant-timed Recall.*

Operation None.

Receive Only Extensions

Description An industry-standard telephone with this class of service (COS) option, can receive calls but cannot originate calls. The industry-standard telephone may, however, originate calls and select features specified in its COS after having received a call, and placed the call on hold by flashing.

Conditions The following conditions apply to this feature:

- If the station tries to originate a call, the attempted call is ignored.
- If used in conjunction with the Flash Disable feature, ALL types of call origination are blocked.
- See *Never A Forwardee* and *Callbacks.*
- COS Options 241 (Receive Only) and 400 (Contact Monitor) are mutually exclusive.

Programming Enable COS Option 241 (Receive Only) for the extension.

Operation None.

Reminder

Description Extension users can program their telephones to provide a single ringing burst at a particular time within a 24-hour period. This feature acts as a reminder for appointments, meetings, etc.

Conditions Do Not Disturb has no effect on the Reminder feature.

You can only program an extension with one reminder at a time, i.e., you cannot program an extension with multiple reminders.

A reminder will occur each day at the programmed time until it is acknowledged.

Programming Enable COS Option 202 (Alarm Call) in the extension's COS.

To allow industry-standard, *SUPERSET 401+*, and *SUPERSET 410* telephones to use this feature, enable System Option 11 (Automatic Wakeup), and assign an access code to Feature 32 (Automatic Wakeup).

Operation Operation varies depending upon the type of set as described below.

Industry-standard, *SUPERSET 401+*, and *SUPERSET 410* Telephones:

To set or modify the current reminder setting:

- Obtain dial tone.
- Dial the Automatic Wakeup code.
- Dial in the wakeup time. The time is specified by dialing the hour and minutes in 24-hour time format.
- Reorder tone is returned for an invalid time; dial tone is returned for a valid time.

To cancel a reminder:

- Obtain dial tone.
- Dial the Automatic Wakeup code, followed by four 9's.
- Replace the handset. The reminder is canceled.

***SUPERSET 420* Telephones:**

To set up a reminder:

- Press SUPERKEY.
- Press the NO softkey until TIMED REMINDER? appears in the display.
- Press the YES softkey. ENTER TIME HH:MM appears.
- Dial the desired time in a 12-hour format (e.g., 01:45). In case of error, use the ← key.
- Specify AM or PM by pressing the AM/PM softkey.
- Press the SAVE softkey. REMINDER SAVED appears briefly in the display.

To display the current reminder setting:

- Press SUPERKEY.
- Press the NO softkey until TIMED REMINDER? appears in the display.
- Press the YES softkey. The current reminder time is displayed.
- Press SUPERKEY to return to normal display

To cancel a reminder before it occurs:

- Press SUPERKEY.
- Press the NO softkey until TIMED REMINDER? appears in the display.
- Press the YES softkey. The current reminder time is displayed.
- Press the CANCEL softkey. REMINDER CANCEL appears briefly in the display.

To change a reminder before it occurs:

- Press SUPERKEY.
- Press the NO softkey until TIMED REMINDER? appears in the display.
- Press the YES softkey. The current reminder time is displayed.
- Press the CHANGE softkey. ENTER TIME HH:MM appears.
- Dial the desired time in a 12-hour format (e.g. 01:45). In case of error, use the ← key.
- Specify AM or PM by pressing the AM/PM softkey.
- Press the SAVE softkey. REMINDER SAVED appears briefly in the display.

To acknowledge an alarm that has occurred:

- You hear a short burst of ringing. REMINDER EXPIRED is shown in the display.
- Press the CONFIRM softkey to acknowledge the alarm. ACKNOWLEDGED appears briefly in the display.

***SUPERSET 430* Telephones:**

To set up a reminder:

- Press the SUPERKEY.
- Press the REMINDER softkey.
- Dial the desired time in a 12-hour format (e.g. 01:45). In case of error, use the ← key.
- Specify AM or PM by pressing the AM/PM softkey.
- Press the SAVE softkey.

To acknowledge an alarm that has occurred:

- You hear one short burst of ringing and a REMINDER EXPIRED is shown in the display.
- Press the flashing ACKNOWLEDGE softkey to acknowledge the alarm. ACKNOWLEDGED appears briefly in the display.

To display the current reminder setting:

- Press the SUPERKEY.
- Find and press the REMINDER softkey; the current reminder time is displayed.
- Press the SUPERKEY to return to normal display.

To cancel a reminder before it occurs:

- Press the SUPERKEY.
- Press the REMINDER softkey. The current reminder time is displayed.
- Press the CANCEL softkey.

Resale Package

Description

The Resale Package is a method of offering the system's Automatic Route Selection (ARS) "Least Cost Routing" facilities to external users requiring low cost Long Distance calling, much like the offerings of other Common Carriers.

DISA trunks are installed for external access to the system. The external user dials up one of the DISA trunks, enters a verified account code, and dials the desired external number. The Direct to ARS feature can be used to route the caller directly to ARS.

This feature is actually a specialized application of the Automatic Route Selection, Toll Control, and Verified Account Code features.

Conditions

The following conditions apply to this feature:

- Verified Account Codes can be activated and/or deactivated for problem accounts.
- See *Automatic Route Selection* and *Verified Account Codes.*

Programming

Program DISA trunks in CDE Forms 01 (System Configuration), 13 (Trunk Circuit Descriptors), and 15 (Dial-In Trunks) - see *Trunk Operation - Direct Inward System Access (DISA).*

Program verified account codes in CDE Form 33 (Account Code Entry); also see *Verified Account Codes (Special DISA).*

Operation

Dial into system via DISA trunk.

Enter a verified account code.

Make call as required.

Ringer Control

Description

You can adjust the ringer in a *SUPERSET* telephone or attendant console.

Conditions

- Attendant console users can adjust the ringer volume.
- *SUPERSET 401+, SUPERSET 420,* and *SUPERSET 430* telephone users can adjust both the ringer volume and the ringer pitch.

Programming

None.

Operation

To adjust ringer volume:

SUPERSET 401+ **and** ***SUPERSET 410*** **Telephones:**

- While the set is ringing, press the VOL ↑ and VOL ↓ keys to increase and decrease the ringer volume.

SUPERSET 420 **Telephones:**

- Press SUPERKEY.
- Press the NO softkey until RINGER ADJUST? appears in the display.
- Press the YES softkey.
- ADJUST PITCH? appears in the display.
- Press the NO softkey.
- ADJUST VOLUME? appears in the display.
- Press the YES softkey.
- Press the VOL ↑ and VOL ↓ keys repeatedly to increase and decrease the ringer volume.
- Press SUPERKEY to exit.

SUPERSET 430 **Telephones:**

- Press SUPERKEY.
- Press the MORE softkey.
- Press the RING ADJUST softkey.
- Press the RINGER VOL softkey.
- Press the VOL ↑ and VOL ↓ keys repeatedly to increase and decrease the ringer volume.
- Press SUPERKEY to exit.

To adjust ringer pitch:

SUPERSET 401+ **and** ***SUPERSET 410*** **Telephones:**

- Lift the handset - dial tone is returned.
- Dial the Tone Demonstration Feature access code.
- Dial the code for Ringer Pitch Adjust (33 - see Tone Demonstration).
- Press the VOL ↑ and VOL ↓ keys repeatedly to increase and decrease the pitch.
- Replace the handset.

***SUPERSET 420* telephones:**

- Press SUPERKEY.
- Press the NO softkey until RINGER ADJUST? appears in the display.
- Press the YES softkey.
- ADJUST PITCH? appears in the display.
- Press the YES softkey.
- Press the VOL ↑ and VOL ↓ keys repeatedly to increase and decrease the pitch.
- Press SUPERKEY to exit.

***SUPERSET 430* telephones:**

- Press SUPERKEY.
- Press the MORE softkey.
- Press the RING ADJUST softkey.
- Press the RINGER PITCH softkey.
- Press the VOL ↑ and VOL ↓ keys repeatedly to increase and decrease the pitch.
- Press SUPERKEY.

Attendant Console:

- While the console is ringing, press the VOL ↑ and VOL ↓ keys to increase and decrease the ringer volume.

Ringing-Discriminating

Description

This feature provides two different ringing cadences to allow a user to distinguish between internal calls (standard ringing) and incoming trunk, attendant, or subattendant calls (discriminating ringing). The system can also be programmed to provide discriminating ringing for all calls.

Standard ringing	1 seconds on, 3 seconds off
Discriminating ringing (1)	400 msec on, 200 msec off, 400 msec on, 3 seconds off.

Conditions

The following conditions apply to this feature:

- Discriminating Ringing does not apply to Night Bells or Recorded Announcement Devices.
- The feature applies to both the prime and additional lines on a *SUPERSET* telephone.
- In an unsupervised transfer situation, the ringing cadence of the called party is not changed after the transfer is done.

Programming

To provide discriminating ringing on incoming trunk and attendant calls:

Enable System Option 17 (Discriminating Ringing).
Disable System Option 18 (Discriminating Ringing Always).

To provide discriminating ringing for all calls:

Enable System Option 18 (Discriminating Ringing Always).

If System Options 17 and 18 are not selected, standard ringing is provided for all calls.

To provide only standard ringing for specific trunks or consoles calling (while still providing discriminating ringing to others), enable COS Option 809 (Standard Ring Applies) for the specific trunks or consoles.

To provide only standard ringing for specific stations (regardless of the ringing cadence determined by the trunk/console caller), enable COS Option 809 (Standard Ring Applies) for the specific station. This option, useful for fax machines or modems that require a standard ringing cadence, is available with software loads F41.0 and above.

Operation None.

Ringing Plan

Description The PABX provides the North American ringing plan, used with the tone plan and rotary dial pulse-to-digit conversion features to enable the system to be used in the North American marketplace. The ringing plan is stored in the database on the PCMCIA Flash card. Refer to the *Engineering Information Practice*, for ringing plan information.

Conditions None.

Programming None.

Operation None.

Ringing Time-Out (Final Ringback)

Description A call to an extension can ring for 1 to 30 minutes before the call is dropped (default ringing time is 1 minute).

Conditions This is only done when forwarding and all recall possibilities have been exhausted; see *Forwarding* and *Recall.*

Programming Enable System Option 51 (Final Ring Timeout) and set it to the desired time-out period.

Operation None.

Satellite PABX

Description An *SX-200* ML PABX can be installed as a satellite PABX. In this configuration, the system has no direct connection to the serving central office for incoming traffic. The satellite PABX has no directory number and receives all its incoming calls over tie trunks from another PABX. Enabling the satellite PABX system option automatically adjusts any required settings for the loss and level plan.

Conditions Some gain settings for the loss and level plan must be adjusted before the PABX can operate as a satellite PABX. Once the Satellite PABX System Option is enabled, these gain adjustments are done automatically.

Programming Enable System Option 31 (Satellite PBX).

Refer to the *Engineering Information Practice* for loss and level plans.

Operation None.

Secretarial Line

Description A *SUPERSET* telephone programmed with a secretarial Multicall line appearance of another extension can override the Do Not Disturb feature on the second set. The second telephone has Do Not Disturb active and the first telephone answers the calls. The secretary can override the second telephone's Do Not Disturb at any time by making the call on one of the Multicall line appearances of the second telephone's prime line. If it is important to contact the second telephone, the first telephone can ring the second telephone, despite the Do Not Disturb feature.

See *Line Types and Appearances* for information on multicall appearances.

Conditions The following conditions apply to this feature:

- The calling *SUPERSET* telephone has to make the call on the line appearance for the feature to operate.
- See *Do Not Disturb* for the interaction of Do Not Disturb with multicall line appearances.
- *SUPERSET 410, SUPERSET 420,* and *SUPERSET 430* telephones can be secretarial sets; the other telephone can be of any type.
- *SUPERSET 401+* telephones cannot be secretarial sets.

Programming Program the secretarial set with at least one multicall appearance, secretarial variant, of the other extension, in the secretarial set's subform of CDE Form 09 (Station/*SUPERSET* Telephones).

Operation The other extension has Do Not Disturb enabled and there is an incoming call. At the secretarial set:

- The multicall appearance rings.
- Answer the call; the call needs to be transferred to the other extension.
- Press the TRANS/CONF key and dial the other extension's number.
- The other extension rings. Consult privately and/or hang up to transfer the call.

Speaker Volume Control

Description *SUPERSET* users may control the volume of the set's speaker.

Conditions Not available on *SUPERSET 401+* telephones.

Programming None.

Operation To control speaker volume:

***SUPERSET 410, SUPERSET 420,* and *SUPERSET 430* Telephones:**

- While set is idle, or during a handsfree conversation, press the VOL ↑ and VOL ↓ keys repeatedly to increase or decrease the speaker volume.

Speedcall

Description This feature allows a *SUPERSET* telephone user to save frequently dialed telephone numbers and to access these numbers by pressing a single key. Only unassigned Line Select keys can be used to save Speedcall numbers.

Access codes for features such as Directed Call Pickup, Remote Call Hold Retrieve, and Call Forwarding may be programmed into Speedcall numbers. Forwarding to a speedcall number can be programmed; see *Call Forwarding.*

Conditions The following conditions apply to this feature:

- The quantity of speedcall numbers available to a *SUPERSET* telephone user is dependent on the number of lines programmed to appear at the set. Lines which are not programmed default to speedcall keys.
- Speedcall numbers can be made private so that they are not displayed when reviewed or used (similar to private abbreviated dial numbers).
- Only unassigned Line Select keys may be programmed with speedcall numbers.

- Except for in the *3 and ** codes described below, the asterisk (*) may not appear in speedcall numbers.
- *3 and ** are ignored in the digit string if ARS leading digits were dialed manually before the Speedcall was selected.
- The same set of speedcall numbers can be programmed for multiple extensions by using key templates; see *Guest Room SUPERSET Key Programming* in Form 37 in the *Customer Data Entry Practice.*
- A speedcall key can be used whenever the *SUPERSET* telephone is idle or when dialing if the manually dialed digits are ARS leading digits. The Speedcall digits will be appended to the ARS digit string; otherwise, the manually dialed digits will be replaced by the Speedcall digits.
- If ARS leading digits appear in the speedcall string, a *3nn is interpreted as insert manual digits.
- When speedcalls are used in ARS, there is a minimum display update of 5 seconds to allow the user to view the number.
- See *Line Selection* for operation when the prime line is not free for the call and another line has not been selected.
- Private numbers are not displayed in SMDR reports.
- A speedcall number may contain an abbreviated dial number.
- COS Option 807 - (SMDR - Display Private Speedcall) must be enabled to display private numbers in SMDR reports.
- Personal speedcall numbers cannot be saved if a Copy Database or Verify Database operation is in progress.
- Personal speed call numbers are not available to logged in ACD agents or supervisors. Once logged in, the set reflects the information programmed for that Agent/Supervisor in Form 38.

Programming

All programming is done at the telephone.

Operation

Operation varies depending upon the type of set as described below.

Note: The following codes can be inserted into a stored number:

*3nn - wait for nn manually dialed digits (nn can range from 01 to 14)

e.g., the general number for external directory assistance is 9+1+(area code)+555-1212; the area code is to be dialed manually. The number to be stored would be 91*3035551212.

** - dial an asterisk

***SUPERSET 401+* Telephones:**

To program or change a speedcall number:

- Expose the Speedcall programming button by removing the set faceplate.
- With the handset on–hook, press the Speedcall programming button.
- Press the desired speedcall key.

- Dial the number to be stored including the outside access code and area code, if necessary. Up to 40 digits may be saved. A 3 second pause can be added to the digit string by pressing the red HOLD key instead of a digit.
- Press Speedcall programming button; the speedcall number is now saved.

To dial a call using a speedcall key:

- Pick up the handset, then press the desired speedcall key.

SUPERSET 410 **Telephones:**

To program, change or clear a speedcall number:

- With the handset on–hook, press SUPERKEY.
- Press a personal speedcall key. (For instructions on how to program a personal key as a speedcall key, refer to *Feature Keys*.)
- Dial the number to be stored or press the REDIAL key to save the last number dialed. If no entry is made, the current entry is cleared.
- Press SUPERKEY; the speedcall number is now saved.

To dial a call using a speedcall key:

- Press the programmed speedcall key with the telephone on–hook; if Immediate Line Select is enabled, or,
- Pick up the handset or press the SPEAKER key, then press the speedcall key.

To set up a speedcall key as a feature access code:

- With the handset on–hook, press SUPERKEY.
- Press a personal speedcall key. (For instructions on how to program a personal key as a speedcall key, refer to *Feature Keys*.)
- Dial 00.
- Dial the feature access code and the extension number of the applicable set.
- Press SUPERKEY.

SUPERSET 420 **Telephones:**

To program, change or clear a speedcall number:

- Press SUPERKEY.
- Press the NO softkey until PERSONAL KEYS? appears in the display.
- Press a personal key that isn't assigned to a line.
- Press the CHANGE softkey. SPEED CALL? appears in the display.
- Press the YES softkey.
- Dial the number to be stored or press the REDIAL key to save the last number dialed. If no entry is made, the current entry is cleared.
- Press the SAVE softkey.
- Press SUPERKEY; the speedcall number is now saved.

To dial a call using a speedcall key:

- Press the programmed speedcall key with the telephone on–hook; if Immediate Line Select is enabled, or,
- Pick up the handset or press the SPEAKER key, then press the speedcall key.

SUPERSET 430 **Telephones:**

To check saved numbers:

- Press the SUPERKEY.
- Find and press the DISPLAY KEYS softkey (use the MORE softkey).
- Press the required line appearance key. The speedcall number currently saved is displayed on the LCD.

To program a speedcall number:

- Press the SUPERKEY.
- Press the MORE softkey until the FEATURE KEY softkey appears.
- Press the FEATURE KEY softkey.
- Press an available personal key.
- Press the CHANGE softkey.
- Press the SPEED CALL softkey.
- Dial the number to be stored. (If the redial number is to be saved, press the REDIAL softkey.)
- If desired, press the MAKE PRIVATE softkey to prevent the DISPLAY KEYS function from revealing the number.
- Check the entry on the LCD. If correct, press the SAVE softkey. To correct an error before saving the entry, use the ← softkey to backspace and clear the incorrect entry, or press the BACKUP softkey to cancel the entire entry. Then re-enter the correct digits.

To dial a call using a speedcall key:

- Press the programmed speedcall key with the telephone on–hook; if Immediate Line Select is enabled, or,
- Pick up the handset or press the SPEAKER key, then press the speedcall key.

Split

Description

Split allows a *SUPERSET* telephone user, engaged in a conference call, to split the call between the conferees. Once active, swapping can take place between the calls, or the conference can be reestablished.

Conditions

The following conditions apply to this feature:

- This feature is available only on *SUPERSET 420* and *SUPERSET 430* telephones.

- A 3-party conference must be active.
- Any one of the three parties may split the call.
- The party splitting the call must press the CONF feature key to re–establish 3-way conversation.
- There cannot be a console in the call.
- There can be no parties in the call holding the call.
- The call cannot be on consultation hold.
- There cannot be an extension with the Non-Busy Extension feature enabled.
- The other two parties cannot be Direct Trunk Select Line trunks.
- There cannot be a *SUPERSET* telephone in the call using the Add Held feature at the time.
- There can be no industry-standard telephone in the call with the Broker's Call feature enabled.

Programming None.

Operation ***SUPERSET 420*** **and** ***SUPERSET 430*** **Telephones:**

- A 3-party conference is established.
- Press the SPLIT softkey. The LCD indicates which party is connected. The other party is on hold.
- To converse with the other party press the TRADE CALLS softkey.
- To re-establish the conference, press the CONFERENCE softkey.

Station Message Detail Recording (SMDR)

Description Station Message Detail Recording (SMDR) allows data to be collected for each outgoing and incoming trunk call. This data can be output to a printer or a data recording device for subsequent processing.

Refer to the *Station Message Detail Recording Practice.*

Conditions None.

Programming The following system options apply:

- 06 - Analog Networking SMDR
- 08 - Five Digit SMDR
- 28 - SMDR - Indicate Long Calls
- 39 - Data SMDR - Indicate Long Calls
- 44 - ACD Reports
- 49 - Pseudo Answer Supervision Timer.

The following COS options apply:

- 246 - SMDR - Extended Record

- 247 - SMDR - Record Meter Pulses
- 700 - SMDR - Does Not Apply
- 702 - SMDR - Overwrite Buffer
- 803 - SMDR - Drop Calls < n Digits
- 804 - SMDR - Drop Incomplete Outgoing Calls
- 806 - SMDR - Record Incoming Calls
- 807 - SMDR - Display Private Speedcall
- 814 - SMDR - Record ANI/DNIS/CLASS
- 906 - Data SMDR - Does Not Apply
- 907 - Data SMDR - Extended Record
- 908 - Data SMDR - Overwrite Buffer

Ensure outgoing SMDR is enabled for the trunk groups in CDE Form 16 (Trunk Groups).

Refer to the *Station Message Detail Recording Practice* for details.

Operation None.

Subattendant - Basic Function

Description This feature provides a *SUPERSET* telephone with enhanced recall and call queuing capabilities, allowing the set to be used as a subattendant position.

Any calls that are handled by the subattendant will recall the subattendant instead of the attendant. Recalls to the subattendant ring the set's prime line.

A *SUPERSET* telephone is normally considered to be busy when the set and/or the prime line appearances are busy. A *SUPERSET* telephone that is programmed as a subattendant is considered busy only if the prime line and all of the line appearances of the prime line are busy. The state of the telephone itself is not checked. Therefore, a subattendant telephone can receive as many calls as there are multicall appearances of its prime line on the telephone.

The *Night/Day Switching* feature can be used to allow the user to select DAY, NIGHT1, or NIGHT2 service for the system.

A subattendant position telephone can also be used as the alternate trunk recall point - see *Trunk Recall.*

Note: This feature is not related to, and does not interact with Subattendant - Enhanced Function features.

Conditions

The following conditions apply to this feature:

- The set to be used as an enhanced answering position should be programmed in its own COS.
- The *SUPERSET* telephone should have at least one multicall line appearance of its prime.
- Line appearance checking is done for re-routes, forwarding, and recalls.
- There can be as many calls ringing at the enhanced answering position set as there are multicall appearances of its prime line on the telephone.
- A message set up on a telephone by the subattendant is automatically canceled when that telephone calls, and is answered by the subattendant that set up the message.
- An enhanced subattendant (COS option 606) appears as an external device when calling a destination that has Call Forwarding Internal/External Split enabled.

Programming

Enable COS Option 606 *(SUPERSET* Telephone - Enhanced Answering Position in the class of service form of the subattendant telephone.

Enable COS Option 262 *(Ignore Forward Busy with Free Appearance)* in the class of service form of the subattendant telephone to have Call Forward Busy ignored when there are any free line appearances of the set.

Program the unused line appearance keys of the enhanced answering position telephone as a multicall appearance of its prime line. Incoming calls that are waiting to be answered queue up on these keys.

Enable COS Option 262 (Ignore Forward Busy with Free Appearance) in the set's COS to have call forward busy ignored when there are any free line appearances of the set.

Operation

None.

Subattendant - Enhanced Functions

Description

The enhanced subattendant functions allow *SUPERSET 430* telephones to be used as a subattendant position for multi-tenanting, consoleless operations (but can be used in conjunction with a console) where call clearing of twenty five calls per hour is considered to be busy. The functions of the *SUPERSET 430* telephone remain unchanged; the Enhanced Subattendant functions are an addition.

The subattendant position can perform functions for extension users; e.g., setup and cancel call forwarding, advisory messages, and toggle do not disturb on and off. The enhanced subattendant functions include:

- System Abbreviated Dial Programming
- Station Advisory Message Programming

- Station Call Forward Setup and Cancel
- Calls Waiting Indication
- Hold Positions
- Listed Directory Number (LDN) Keys
- Enhanced Zone Paging
- Station Do Not Disturb Setup and Cancel
- System Time and Date Setup
- Enhanced Recalling Capabilities

Discriminating ringing is available for calls from the subattendant position (See Ringing - Discriminating).

Note: The Subattendant - Enhanced Functions feature is not related to, and does not interact with, the Subattendant - Basic Function feature.

Conditions

The following conditions apply to this feature:

The subattendant can program an extension call forwarding or do not disturb function without enabling the extension call forwarding COS options.

Programming

Enable COS Option 125 (Attendant Multi-New Call Tone) to provide an audible alert for calls waiting when the attendant is engaged in a call.

Set the time interval for audible alerts in COS Option 404 (Recording Failure to Hangup Timer). The default for COS Option 404 is 30 seconds.

In either CDE Form 09 (Station/*SUPERSET* Telephones) or CDE Form 45 (Key System Telephones):

- Press the DEVICE TYPE softkey to set or change the device type. Use the SUB ATT softkey to set/change it to subattendant. The subattendant device type must be set before any other fields. If the device is already programmed, it must be deleted and re-entered.
- Press the EXPAND SET softkey to get to the Expand Set Subform - (subattendant keys are defined here) - refer to the specific subattendant feature for further information.

Operation

Refer to the specific subattendant feature.

Subattendant - Abbreviated Dial Programming

Description

The Subattendant - Abbreviated Dial Programming feature allows the subattendant to program system abbreviated dial numbers from the subattendant telephone set. The subattendant has the option of making abbreviated dial numbers confidential.

Conditions

The following conditions apply:

- This feature is available only to *SUPERSET 430* telephones that have been programmed with enhanced subattendant functionality (see *Subattendant - Enhanced Functions*).

- Only one device can program system abbreviated dial numbers at a time.

Programming The following steps are required:

- In Form 3, enable COS Option 111 (Attendant Abbreviated Dial Programming) in the subattendant COS.
- For the display of confidential abbreviated dial numbers, enable COS Option 110 (Attendant Abbreviated Dial Confidential Number Display).

Operation To program an abbreviated dial number:

- Press the SUPERKEY.
- Press the MORE softkey.
- Press the FEATURE KEY softkey.
- Press the SYSTEM softkey.
- Enter an Abbreviated Dial index number.
- Press the ENTER softkey.
- Enter the new number (see *Abbreviated Dial* for details on special character entry).
- Press the MAKE PRIVATE softkey (if desired).
- Press the SAVE softkey.
- Press the SUPERKEY.

Subattendant - Advisory Message Setup

Description There are eight default advisory messages and seven programmable advisory messages that the subattendant may setup on behalf of another extension. The subattendant can read another extension's currently displayed advisory message, or read through the available advisory messages and choose one for display on the set, or program one for display.

Conditions The following conditions apply:

- This feature is available only to *SUPERSET 430* telephones that have been programmed with enhanced subattendant functionality (see *Subattendant - Enhanced Functions*).
- Messages can only be setup for *SUPERSET* telephones.

Programming Enable COS Option 150 - Sub-attendant Station Setup Advisory Message in the subattendant COS.

Operation To remotely set up an advisory message on a *SUPERSET* telephone; from the subattendant position:

- Press SUPERKEY.
- Press the STATIONS softkey.
- Enter the extension number of the applicable *SUPERSET* telephone.

- Press the ADVISORY MSG softkey.
- Find the desired message - use the NEXT MSG (PREVIOUS MSG) and/or SHOW MSG NO. softkeys to move through the list of existing messages. Stop at the desired message.
- Press the TURN MSG ON softkey, or
- Press the CREATE MSG softkey to create a custom message.
- Press the TURN MSG ON softkey.

To cancel an advisory message on a *SUPERSET* telephone:

- Press SUPERKEY.
- Press the STATIONS softkey.
- Enter the extension number of the applicable *SUPERSET* telephone.
- Press the ADVISORY MSG softkey.
  - The currently displayed message will be displayed
- Press the TURN MSG OFF softkey.

Subattendant - Call Forward Setup and Cancel

Description

The Subattendant - Call Forward Setup and Cancel feature allows the subattendant to setup, review, and cancel call forwarding for any extension. The extension for which the subattendant sets up forwarding need not have any of the call forwarding features in its COS. The subattendant may also set up call forwarding from the extension to the subattendant.

All forwarding types can be setup or canceled in this function, whether or not forwarding types have been previously defined for either the subattendant or the affected extension.

Standard call forwarding routes all calls to the same destination. Split Call Forwarding is also available. Split Call Forwarding allows the subattendant to route internal and external calls to different destinations (see Call Forwarding - Internal/External Split).

Conditions

The following conditions apply:

- For general conditions see *Call Forwarding.*
- This feature is available only to *SUPERSET 430* telephones that have been programmed with enhanced subattendant functionality (see *Subattendant - Enhanced Functions*).

Programming

The following programming is required:

- Enable COS Option 123 (Attendant Call Forward Setup and Cancel) in the subattendant COS.
- If Split Call Forwarding is desired, enable COS Option 260 (Internal/External Split Call Forward) for the extension.

Operation

To set up call forwarding for an extension from the subattendant telephone:

- Press SUPERKEY.
- Press the STATIONS softkey.
- Enter the extension number of the extension being setup.
- Press the FORWARDING softkey.
- Press the CHANGE softkey.
- Press the appropriate softkey for desired type of forwarding (see note 1).
- Enter the number of the extension that you want the calls forwarded to.

OR

- Press the CURRENT NO. softkey if the number displayed is the number that you want the calls forwarded to.
- Press the SAVE/ON or SAVE/OFF softkey.
- Press SUPERKEY to exit or you may continue and setup the other forwarding type (if Split Forwarding is enabled).

To cancel call forwarding for an extension from the subattendant telephone:

- Press SUPERKEY.
- Press the STATIONS softkey.
- Enter the extension number of the extension having forwarding canceled.
- Press the FORWARDING softkey.
- Press the TURN FWD OFF softkey.
- Press SUPERKEY to exit.

To set up Call Forwarding - Internal/External Split for an extension from the subattendant telephone:

- Press SUPERKEY.
- Press the STATIONS softkey.
- Enter the number of the extension being set up.
- Press the FORWARDING softkey.
- Press the Change softkey.
- Press the appropriate softkey for desired type of forwarding (see note 1).
- Press either the Internal or the External softkey (see note 2).
- Enter the number of the extension that you want the calls forwarded to.

OR

- Press the CURRENT NO. softkey if the number displayed is the number that you want the calls forwarded to.
- Press the SAVE/ON or SAVE/OFF softkey.
- Press SUPERKEY to exit.

OR

- Continue and set up the other forwarding type.

- Press SUPERKEY to exit.

To cancel Call Forwarding - Internal/External Split for an extension from the subattendant telephone:

- Press SUPERKEY.
- Press the STATIONS softkey.
- Enter the number of the extension that you want to cancel forwarding from.
- Press the FORWARDING softkey.
- Press the INT FWD OFF or the EXT FWD OFF softkey.
- Press SUPERKEY to exit.

OR

- Continue and turn off the other call forwarding type.
- Press SUPERKEY to exit.

To review call forwarding for an extension from the subattendant telephone:

- Press SUPERKEY.
- Press the STATIONS softkey.
- Enter the extension number of extension being set up.
- Press the FORWARDING softkey.
- Press SUPERKEY when done.

Note: 1. For programming details see *Call Forwarding*.

2. The INTERNAL/EXTERNAL softkeys will be shown only if COS Option 260 (Internal / External Split Call Forward) is enabled. See *Call Forwarding*.

Subattendant - Calls Waiting Indication

Description

The Calls Waiting indicator appears in all call processing states and is displayed in the subattendant C/W (Calls Waiting) area of the display. The Calls Waiting indicator appears in the top right corner of the display, directly below the area where the Forwarding flag appears. The Call Waiting flag takes precedence over the Message Waiting and Mic On flags when clashes occur.

The subattendant may have calls from outside trunks and extensions queued that are waiting to be answered. The number shown by the Calls Waiting indicator, is the total number of calls in the queue. This includes only calls ringing LDN's (or the RECALL key) that appear on the subattendant telephone and any calls ringing the night bell.

Each new call ringing the subattendant position increments the indicator. The indicator decrements each time a caller hangs up.

Conditions

The following conditions apply to this feature:

- This feature is available only to *SUPERSET 430* telephones that have been programmed with enhanced subattendant functionality (see *Subattendant - Enhanced Functions*).
- If there are no calls in the Calls Waiting queue, there is nothing in the CW area of the LCD display and either the Msg Waiting or Mic On flag may occur.
- The maximum number of calls waiting that can be displayed at the subattendant is 99 (the actual queue maximum is 200).
- Only night bell callers allowed to connect to the subattendant as specified in Form 5, Tenant Interconnection, will be counted on the Subattendant C/W.

Programming None.

Operation None.

Subattendant - Date and Time Setup

Description

When the subattendant position is idle, it continually displays the time and date (day, month, year) on the LCD display. The system time and/or date can be changed using the subattendant SUPERKEY and softkeys.

The time may be displayed in 12-hour or 24-hour format depending on the system feature settings.

Conditions

The following conditions apply:

- This feature is available only to *SUPERSET 430* telephones that have been programmed with enhanced subattendant functionality (see *Subattendant - Enhanced Functions*).
- A time/date change may cause some traffic measurements to be lost, and can also affect Automatic Call Distribution (ACD) Reports. Care must be taken when setting the COS option for this feature.

Programming

Enable COS Option 122 - Attendant Setup Time/Date, for 24 -hour clock; disable for 12-hour clock.

Operation

To set the time from a subattendant telephone:

- Press SUPERKEY.
- Press the MORE softkey three times.
- Press the TIME softkey.
- Enter the time in the correct format.
- Press the AM or PM softkeys (if necessary).
- Press the SAVE softkey.

To set the date:

- Press SUPERKEY.
- Press the MORE softkey three times.
- Press the DATE softkey.
- Enter the date in the indicated format.
- Press the SAVE softkey.

Subattendant - Hold Positions

Description

This feature provides the subattendant with up to three hold position keys. When enabled, the hold position keys permit the subattendant to answer other Listed Directory Numbers (LDN) or the Prime (Intercom) line without having to release current calls on the subattendants telephone first. It is recommended that at least one hold position be programmed, otherwise the subattendant will have to release incoming calls on the prime line of the subattendant telephone before being able to answer other incoming LDN or Prime line calls.

The subattendant can transfer a current incoming call to one of the hold positions by selecting the corresponding hold position key. The call is then placed in the hold position, releasing the prime line for the subattendant to receive another incoming call.

The indicator for the hold position to which the call has been transferred flashes, indicating to the subattendant that a call is currently occupying that hold position. The ADD HELD softkey appears when a call is transferred to a hold position.

Should all programmed hold positions at a subattendant position become occupied, incoming calls on the subattendant prime line may be placed on hold by selecting the red HOLD key. The red HOLD key places the incoming call on a hard hold on the subattendant prime line, but since the line is occupied, subsequent LDN calls and Recalls cannot be answered until the call on hard hold is released. The subattendant may select a held party (via the HOLD key) while dialing on the Prime (Intercom) Line.

Recall

COS Option 116 - Attendant-Timed Recall - Hold timer applies to calls placed in subattendant hold positions. If this timer times-out for a call being held in one of the hold positions, the indicator associated with the hold position flashes and a new call ring is given, indicating to the subattendant that the hold position is recalling (at this point the Call Hold Notify timer is started).

COS Option 681 - Key Set/Sub Att - Call Hold Notify Timer also applies to calls placed in subattendant hold positions. This timer starts after the expiration of the Timer Recall Hold timer. Every time this timer expires, a new call ring is given, and the timer is restarted. This will continue until the call is retrieved from hold. If this is not programmed there will be no notifications and the caller will remain on hold until retrieved. This should no be confused with COS Option 254 - Call Hold Recall Timer, which is used in conjunction with hard held lines on the subattendant set, rather than the hold positions.

Add Held Prompt

An ADD HELD prompt softkey is provided to permit the subattendant to pickup calls currently in hold positions and add them to an existing conversation on one of the set's active lines to create a 3-party conference.

Conditions

The following conditions apply to this feature:

- This feature is available only to *SUPERSET 430* telephones that have been programmed with enhanced subattendant functionality (see *Subattendant - Enhanced Functions*).
- Each subattendant position can be programmed to provide up to three hold positions.

Programming

To program hold positions:

- Enter the Expand Set Subform of CDE Form 09 (Station/*SUPERSET* Telephones) or CDE Form 45 (Key System Telephones), and program one or more keys as hold positions via the HOLD POS softkey.
  - Note that the given telephone must be a subattendant device type to be offered the HOLD POS softkey.

Related programming:

- Set an appropriate value for COS Option 116 - Attendant-Timed Recall (Hold) timer in the subattendant COS.
- Set an appropriate value for COS Option 681 - Key Set/Sub Att - Call Hold Notify Timer in the subattendant COS.

Operation

To place a call on hold using a subattendant hold position key:

- Press a free hold position key.

  The associated line appearance indicator flashes, and the ADD HELD softkey is enabled.

To retrieve call from subattendant hold position key:

- Press the desired hold position key.

  The associated line appearance indicator changes from a flashing state to an inactive (off) state, while the prime line appearance indicator indicates an active (on) line condition. The held call is transferred to prime line.

To add a held call to current conversation:

- In Off-hook state select the ADD HELD softkey, or while in handsfree mode, press the ADD HELD softkey after choosing a line.
- Press the desired hold position key.

  The held call is transferred to current conversation and associated line appearance indicator changes from a flashing state to an inactive (off) state.

Subattendant - LDN Keys

Description

Each subattendant can have up to three keys programmed as listed directory number (LDN) keys. The LDN keys appear on the subattendant telephone line keys. The LDNs may be programmed to appear on other subattendant telephones or attendant consoles to permit greater call handling flexibility. When this occurs, the COS and tenant of the subattendant LDN is taken from the subattendant or console with the lowest bay, slot and circuit on which the LDN is programmed.

The LDN keys and the RECALL key (see *Subattendant - Recall)* act as call queueing indicators. Unlike line keys; they cannot be selected to dial on and conversations cannot be held on them. When a subattendant LDN call is answered, the call is automatically connected to the prime line (or intercom line) of the subattendant telephone.

Each LDN position can be programmed as the answer point for a trunk or reroute destination for a particular type of call. To ensure that the prime line is free to answer any LDN calls, the subattendant prime line cannot be programmed to appear on other devices.

The subattendant can answer a LDN call three ways:

- by going offhook; the longest waiting LDN call is then automatically connected to the subattendant's prime line, or
- by pressing the SPEAKER key; the longest waiting LDN call is then automatically connected to the subattendant prime line, or
- by selecting the LDN key directly.

LDN keys with "ring type" set to NO RING are not automatically selected when the subattendant goes offhook (or selects the SPEAKER key), therefore must be answered manually by selecting the No Ring LDN key.

Once answered by the subattendant, an LDN call is treated as though it were a regular call received on a *SUPERSET 430* telephone, with the exception of serial calls.

Conditions

The following conditions apply to this feature:

- This feature is available only to *SUPERSET 430* telephones that have been programmed with enhanced subattendant functionality (see *Subattendant - Enhanced Functions*).
- Subattendant LDNs and recall calls will override the subattendant DND settings and will ring the DND subattendant telephone on the LDN or RECALL key.
- Station and *SUPERSET* telephones and consoles cannot directly dial an LDN.
- There is a maximum of three LDN keys per subattendant position.
- For multi-appearance LDN keys, the console (or subattendant position) with the lowest physical location number (bay #/ slot #/ circuit #) is always the owner of an LDN.

Programming

To program LDN keys, enter the Expand Set Subform of CDE Form 09 (Station/*SUPERSET* Telephones) or CDE Form 45 (Key System Telephones), and:

- Program one or more keys as LDN Keys via the LDN softkey.
- Select a ring type for each LDN.
- Enter an access code in the EXT NUM field for each LDN - this may be a unique number, or an existing LDN number.
- Enter a label for each LDN.

Note: If a subattendant desires to always automatically clear down the current call when another line, LDN, recall or hold position is selected, enable COS Option 601 (*SUPERSET* - Auto Hold Disable) in the subattendant COS.

Operation

The subattendant can selectively answer any incoming call type by pressing the appropriate LDN softkey, or calls may be answered on a first-come first-served basis by using the SPEAKER key (handsfree operation) or by going offhook.

Subattendant - Paged Hold Access

Description

The subattendant can place an incoming call on hold, page for the called party, then inform the called party of the digits to dial, the called party can then pick up the incoming call directly from the subattendant hold position. When the subattendant accesses a PA Pager with a call on hold position, the Hold Pickup access code is displayed along with the subattendant identifier code. The subattendant would then instruct the paged party to call those digits followed by the hold position number to pick up their call. Also see *PA Paging* in this document.

Conditions

The following conditions apply to this feature:

- This feature is available only to *SUPERSET 430* telephones that have been programmed with enhanced subattendant functionality (see *Subattendant - Enhanced Functions*).

- Consoles, subattendants, industry-standard telephones, *SUPERSET* telephones, DISA trunks and TIE trunks can pickup the held calls.
- The party picking up the call must be able to connect to the held party; see *Device Interconnection Control.*
- An extension cannot pick up the held party if the extension has a consultation hold in progress and the held party has COS Option 233 (Never A Consultee) enabled.
- A station or *SUPERSET* telephone (with a consultation hold in progress) cannot pick up the held trunk if the station or *SUPERSET* telephone has COS Option 214, (Cannot dial a trunk after flashing) enabled.
- A station or *SUPERSET* telephone in a conference with a trunk on consultation hold cannot pick up the held trunk if the station or *SUPERSET* telephone has COS Option 215, (Cannot Dial a Trunk if Holding or in Conf With a Trunk) enabled.

Programming

The following programming is required:

- Assign an access code to Feature 16 (Hold Pickup Access - Attendant Hold Slots).
- Enable COS Option 225 (Hold Pickup - Attendant Page Access) in the COS of the device from which the pickup call is made.

Operation

If paging the default paging zone:

- Place the incoming calling party on hold by using one of the subattendant hold positions.
- Press the PAGE softkey - when the subattendant accesses the PA Paging function, the subattendant telephone display indicates:
  - the feature access code, assigned to Feature 16,
  - followed by two digits (11 to 99) that identify the subattendant,
  - followed by the hold position number.
- Page the desired party, specifying the displayed number (the last number being the hold position number); e.g., 677112 (where 677 is the feature access code, 11 is the subattendant identifier, and 2 is the hold position number).

When the paged party dials the subattendant Hold Pickup Access code number, the paged and held parties are connected.

Note: If the paged party does not call, the held party recalls the subattendant automatically after the hold recall timeout; see *Subattendant - Hold Positions* in this document.

If paging a zone other than the default zone:

- Place the incoming calling party on hold by using one of the subattendant hold positions.
- Dial the 'Paging Access To Specific Zones' access code, followed by the desired zone.

- Page the desired party, specifying the Subattendant Hold Pickup access code and the subattendant Identifier, followed by the hold position number.

When the paged party dials the Subattendant Hold Pickup access code number, the paged and held parties are connected.

Note: If the paged party does not call, the held party recalls the subattendant automatically after the hold recall timeout; see *Subattendant - Hold Positions* in this document.

Subattendant - Recall

Description

The recall feature ensures that calls do not remain unanswered or on hold for an unlimited period of time. Any call that has been extended by a subattendant, recalls the subattendant position if the call is not answered or remains on hold at the end of a timeout period.

The LDN keys (see *Subattendant - LDN Keys)* and the RECALL key act as call queueing indicators. Unlike line keys; they cannot be selected to dial on and conversations cannot be held on them. When a subattendant recall is answered, the call is automatically connected to the prime line (or intercom line) of the subattendant telephone.

To ensure that the prime line is free to answer any recall calls, the subattendant prime line cannot be programmed to appear on other devices. To avoid recall calls tieing up the prime line of the subattendant it is important to program the RECALL key. Recall calls to the subattendant will then be queued on the RECALL key.

The subattendant can answer a recall call three ways:

- by going offhook; the longest waiting recall call is then automatically connected to the subattendant prime line, or
- by selecting the SPEAKER key; the longest waiting recall call is then automatically connected to the subattendant prime line, or
- by selecting the RECALL key directly.

For further information on the recall feature, refer to *Recall* in this document.

Conditions

The following conditions apply to this feature:

- This feature is available only to *SUPERSET 430* telephones that have been programmed with enhanced subattendant functionality (see *Subattendant - Enhanced Functions*).
- Subattendant recall calls will override the Subattendant DND settings and will ring the DND subattendant telephone on the RECALL key.
- Station and *SUPERSET* telephones and consoles cannot directly dial a RECALL key.

- There is a maximum of one RECALL key per subattendant position.
- Also see *Conditions* under *Recall.*

Programming

To program a Recall key, enter the Expand Set Subform of Form 09 (Station/*SUPERSET* Telephones) or Form 45 (Key System Telephones), and:

- Program a RECALL key via the RECALL softkey.

Also see *Programming*, under *Recall* in this document.

Operation

When a call recalls to the subattendant position, press the RECALL key, go off-hook, or press the SPEAKER ON/OFF key (handsfree mode).

Subattendant - Station DND Setup

Description

The subattendant can set up or cancel Do Not Disturb (DND) for an extension by selecting the DO NOT DISTURB softkey under the subattendant stations function.

Selection of the DO NOT DISTURB softkey causes the extension information on the subattendant display to be updated to indicate that the extension status has been changed to Do Not Disturb if the station was not already Do Not Disturb or if the device was already in the Do Not Disturb state when the softkey was selected, the Do Not Disturb indicator is turned off. The corresponding set will also have its DND indicator updated. See *Do Not Disturb* in this document.

Conditions

This feature is available only to *SUPERSET 430* telephones that have been programmed with enhanced subattendant functionality (see *Subattendant - Enhanced Functions*).

Programming

Enable COS Option 121 - Attendant Station Do Not Disturb in the subattendant COS.

Operation

To remotely set up Do Not Disturb on an extension from the subattendant position set:

- Press SUPERKEY.
- Press the STATIONS softkey.
- Enter extension number.
- Press the DO NOT DISTURB softkey.
  - In the top right hand portion of the display, the extension's current state indication changes to DND.
- Press SUPERKEY to exit.

To cancel Do Not Disturb on an extension:

- Press SUPERKEY.
- Press the STATIONS softkey.
- Enter extension number.
- Press the DO NOT DISTURB softkey.
- Press SUPERKEY to exit.
  - In the top right hand portion of the display, the DND indication changes to the current state of the extension.

SUPERSET 401+ Telephone

Description The *SUPERSET 401+* telephone is a single-line digital telephone that employs DNIC line transmission technology. It provides a **Flash** key, a **Message** key with associated indicator, a red **Hold** key, and keys for adjusting the ringer and handset receiver volume. The set also has keys designed for use as Speed call keys or feature access key.

Refer to the *Peripherals Devices Practice.*

Conditions None.

Programming Refer to the *Customer Data Entry Practice*; program *SUPERSET 401+* telephones in CDE Form 09 (Station/*SUPERSET* Telephones); select their COS Options in CDE Form 03 (COS Define).

Operation Refer to the specific feature.

SUPERSET 410 Telephone

Description The *SUPERSET 410* telephone is a multi-line digital telephone that employs DNIC line transmission technology. The *SUPERSET 410* telephone has 6 programmable keys, each with an associated LCD indicator. These keys can be programmed as speedcall keys, feature access keys or line appearances. The lowest key must be the prime line appearance. In addition, there is a **Message** key and a **Microphone** key, a red **Hold** key, and 7 other fixed function keys. The **Message** key and **Microphone** key both have an associated indicator lamp.

You can use up to three Programmable Key Modules (PKMs) with this set. The PKMs are daisy-chained together and connect to *a MILINK* port on the base of the set. You can also connect one *MILINK* Data Module to this set.

Refer to the *Peripherals Devices Practice.*

Conditions None.

Programming Refer to the *Customer Data Entry Practice*; program *SUPERSET 410* telephones in CDE Form 09 (Station/*SUPERSET* Telephones); select their COS Options in CDE Form 03 (COS Define).

Operation Refer to the specific feature.

SUPERSET 420 Telephone

Description The *SUPERSET 420* telephone is a multi-line digital telephone that employs DNIC line transmission technology. The *SUPERSET 420* telephone has 12 programmable keys with associated LCD indicators. These keys can be programmed as speedcall keys, feature access keys or line appearances. The lowest key must be the prime line appearance.

Directly below the 12 programmable keys is a 2 x16 alphanumeric display and three softkeys. The three softkeys allow set users to select command prompts that appear in the display. The set also provides a **Message** key and **Microphone** key with indicator lamps, and 8 fixed function keys without indicator lamps.

You can use up to three Programmable Key Modules (PKMs) with this set. The PKMs are daisy-chained together and connect to *a MILINK* port on the base of the set. You can also connect one *MILINK* Data Module to this set.

Refer to the *Peripherals Devices Practice*.

Conditions None.

Programming Refer to the *Customer Data Entry Practice*; program *SUPERSET 420* telephones in CDE Form 09 (Station/*SUPERSET* Telephones); select their COS Options in CDE Form 03 (COS Define).

Operation Refer to the specific feature.

SUPERSET 430 Telephone

Description The *SUPERSET 430* telephone is a multi-line digital telephone that employs DNIC line transmission technology. The *SUPERSET 430* telephone has 12 programmable keys with associated LCD indicators. These keys can be programmed as speedcall keys or line appearances.

Directly below the 12 programmable keys is a 4 x 40 alphanumeric display and six softkeys. The softkeys allow set users to select command prompts that appear in the display. The set also provides a **Message** key and **Microphone** key with indicator lamps, and five fixed function keys without indicator lamps.

You can use up to three Programmable Key Modules (PKMs) with this set. The PKMs are daisy-chained together and connect to *a MILINK* port on the base of the set. You can also connect one *MILINK* Data Module to this set.

Refer to the *Peripherals Devices Practice.*

Conditions

None.

Programming

Refer to the *Customer Data Entry Practice*; program *SUPERSET 430* telephones in CDE Form 09 (Station/*SUPERSET* Telephones); select their COS Options in CDE Form 03 (COS Define).

Operation

Refer to the specific feature.

SUPERSET **LCD Display**

Description

The *SUPERSET 420* and *SUPERSET 430* telephones are equipped with a liquid crystal display (LCD). The display (not to be mistaken for the LCD line appearance indicators) indicates the date and time of day, along with softkey names for the set's softkeys. A redial number is displayed, if applicable. Also, when a *SUPERSET 420* or *SUPERSET 430* telephone accesses a trunk and establishes a call, the duration of the call is displayed. For other information displayed, details can be found under the appropriate feature heading.

Conditions

The following conditions apply to this feature:

- This feature is only available on *SUPERSET 420* and *SUPERSET 430* telephones.
- The duration shown is the time that all parties in the PABX spend talking on the trunk call. If a trunk is transferred from extension to extension, the time is not restarted.
- The duration display does not indicate the complete time that the trunk has been answered where recordings are involved. The recording answers the trunk but the call duration counting does not start until the trunk is answered by an extension.
- The duration display indicates the time up to a maximum of ten hours (0 thru 9:59). After ten hours, the duration display resets to zero then continues to count for another ten hours.
- If Analog Networking is enabled, the extension number of the party calling from the other node is displayed.
- If the trunk has a name, the trunk name is displayed. If the trunk has no name programmed, the group name of the trunk is displayed. For *SUPERSET 420* and *SUPERSET 430* telephones, the digits dialed are displayed instead of the name for outgoing trunk calls.

Programming None.

Operation None.

Swap (Trade Calls)

Description This feature permits a *SUPERSET* telephone user to alternate conversation between two calls. Call Swap places one call on hold while conversation continues with the other call. This feature is similar to the Broker's Call feature available on industry-standard telephones.

Conditions The following conditions apply to this feature:

- Swap operates only on *SUPERSET* telephones.
- There must be a party in conversation and a party on consultation hold for Swap to operate.
- A swap cannot be done when talking to the console or a non-busy extension.
- The SWAP/TRADE CALLS feature key on a *SUPERSET 410* telephone, performs a Swap Campon if a camped on caller is present and may be swapped in. Only one of the Swap or Swap Campon functions is available at the telephone at one time; see *Swap Campon.*
- If there is another party in the call with a consultation hold then the Swap cannot be done.

Programming For feature key activation of Swap from a *SUPERSET 410* telephone, program a SWAP/TRADE CALLS feature key (see *Feature Keys*).

Operation Operation varies depending upon the type of set as follows:

***SUPERSET 401+* Telephones:**

There is a party on hold and a party in conversation (e.g., the hold key has been pressed and a new party called); proceed as follows:

- Press the HOLD key repeatedly to alternately place one call on hold and connect to the other call.

***SUPERSET 410* Telephones:**

There is a party on consultation hold and a party in conversation (e.g., the TRANS/CONF key has been pressed and a new party called); proceed as follows:

- Press the SWAP/TRADE CALLS feature key repeatedly to alternately place one call on hold and connect to the other call.

***SUPERSET 420* and *SUPERSET 430* Telephones:**

There is a party on consultation hold and a party in conversation (e.g., TRANS/CONF has been pressed and a new party called, or a conference has been split). Proceed as follows:

- Press the SPLIT softkey to break up the three party conference. The party in conversation is held, the held party is connected to the set.
- Press the TRADE CALLS softkey repeatedly to alternately place one call on hold and connect to the other call.

Swap Campon

Description

This feature allows the user of a telephone to place the current call on hold and converse with a camped-on party. The telephone user can alternate between the two calls as required, form a 3-party conference, or release the telephone from the call, leaving the other two parties connected. This also applies to members of hunt groups. The first extension in the hunt group that does not have Do Not Disturb activated and is logged in (UCD agent hunt groups) is able to swap in the first waiting caller on the hunt group.

Normally the *SUPERSET 410* telephone Swap operation is a "swap calls" operation; but it becomes a "swap campon" operation when there is a party camped on, and a Swap Campon is possible.

Conditions

The following conditions apply to this feature:

- For *SUPERSET 410* telephones, this feature is accessed by a SWAP/TRADE CALLS feature key.
- This feature is accessed by a switch-hook flash for industry-standard telephones and *SUPERSET 401+* telephones.
- The feature is only available when talking in a call.
- When a call camps on to a key line then all *SUPERSET* telephones using an appearance of the line are able to use the feature.
- The caller that is swapped in is the first waiting caller; see *Campon.*
- The feature is unavailable when:
  - a consultation hold is already in progress,
  - overriding a call or another party in the call is being overridden,
  - a console is in the current call,
  - another party has the current call on consultation hold,
  - a party has the call on hold,
  - the call is ringing back a party in the call (Station Transfer Security),
  - a Non-busy Extension is in the call,
  - the current call is a conference call with a CO line in the call.
- For industry-standard telephones, flashing for a waiting call takes priority over the flash features *Flash for Attendant,* and *Flash for Dial Tone.*
- For industry-standard telephone, COS Option 223 Flash Disable is ignored for this feature.

Programming

For industry-standard telephones enable COS Option 205 (flash for waiting call) and enable System Option 38 (switch-hook flash).

For feature key activation of Swap from a *SUPERSET 410* telephone, program a SWAP/TRADE CALLS feature key (see *Feature Keys*).

Operation

Industry-standard and *SUPERSET 401+* Telephones:

A call is in progress at the extension; a second caller camps on;

- Flash the switch-hook or press the FLASH key (*SUPERSET 401+* telephones). The current call is put on hold. The telephone is connected to the second caller.
- To return to the original conversation, flash switch-hook or press the FLASH key if the telephone is a broker.
- To form a conference, flash the switch-hook or press the FLASH key if the telephone is not a broker.
- To exit from the 3-party conference, while leaving the two other parties connected, hangup.

***SUPERSET 410* Telephones:**

A call is in progress at the extension; a second caller camps on;

- Press the SWAP/TRADE CALLS feature key. The current call is put on hold: the telephone is connected to the second caller.
- To return to the original conversation, press the SWAP/TRADE CALLS feature key.
- To form a conference, press the TRANS/CONF key.
- To exit from the 3-party conference, while leaving the two other parties connected, hangup.

***SUPERSET 420* Telephones:**

A call is in progress at the set; a second caller camps on; the TRADE softkey appears:

- Press the TRADE softkey. The current call is put on hold. The telephone is connected to the second caller.
- To return to the original conversation, press the TRADE softkey again,

  OR
- Press the TRANS/CONF key to form a conference. To exit from the conference, while leaving the two other parties connected, press the CANCEL key.

***SUPERSET 430* Telephones:**

A call is in progress at the set; a second caller camps on; the CALL WAITING softkey appears:

- Press the CALL WAITING softkey. The current call is put on hold; the telephone is connected to the second caller.

- To return to the original conversation, press the TRADE CALLS softkey, or,
- To form a conference, press the CONFERENCE softkey. To exit from the conference call, while leaving the two other parties connected, press the RELEASE ME softkey.

System Identifier

Description A unique 1- to 3-digit identifier may be assigned to the system. It appears on traffic measurement and SMDR reports to identify the system when central polling equipment is used for traffic measurement, trunk SMDR, ACD, DATA SMDR, ACD SMDR and analog networking.

Conditions The system identifier is programmed only from the attendant console.

Programming Refer to the *Peripherals Devices Practice.*

Operation None.

Tandem Operation

Description *SX-200* PABXs support two PABXs connected in tandem using tie trunks to connect the two systems together. See the analog networking, satellite PABX, and resale package features in this section. See *Analog Networking, Satellite PABX* and *Resale Package.*

Conditions The following conditions apply to this feature:

- Appropriate automatic route selection (ARS) must be provided.
- When a supervision is received from an outgoing trunk, if the calling party on the trunk is an incoming DID or Tie trunk then the supervision is passed on to the incoming DID or Tie trunk (the incoming trunk is given another answer supervision).
- Device interconnection must be set up appropriately to ensure that valid trunk connections are allowed. Device interconnection must be checked carefully when using loop start CO or DISA trunks to make outbound calls; see *Device Interconnection Control.*

Programming None.

Operation None.

Tenanting

Description

Economy of scale makes sharing PABX services practical. Using the tenanting features, up to 25 small businesses, or departments of a larger business, can share the services of a *SX-200* PABX. Logically, the PABX is divided into 25 separate PABXs, each providing its tenant with customized features and services.

Consoles, night bells, CO trunks, and dial-in trunks can either be shared between tenants or allocated individually to each tenant. Switching to night service can be done centrally, or by an individual tenant. Calls through the PABX can be blocked, so tenants can only call each other on CO trunks.

Unanswered or after-hours calls can be answered by a “landlord” console. Tenants can gain additional flexibility by using *SUPERSET 430* telephones as subattendant positions. The main console position could be handled by a *SUPERSET 430* telephone, using line keys to receive tenant recalls.

Data calls may be assigned to separate tenants, as required, to control access to data devices.

Refer to the *Tenanting Practice.*

Conditions

The following conditions apply to this feature:

- A maximum of 25 Tenant groups may be programmed.
- If tenanting is used to segregate departments, CDE Form 06 (Tenant Night Switching Assignment) must be programmed accordingly.
- Tenants may have names associated with them.

Programming

Assign consoles to the desired tenant groups via CDE Form 07 (Console Assignments).

Assign extensions and *SUPERSET* telephones to the desired tenant groups via CDE Form 09 (Station/*SUPERSET* Telephones).

Assign trunks to the desired tenant groups via CDE Forms 14 (Non-Dial-In Trunks) and 15 (Dial-In Trunks).

Assign a tenant number to ACD Paths via Form 41 (ACD Path Form).

Assign the desired tenant name, day and night service numbers in CDE Form19 (Call Rerouting Table).

Program CDE Form 06 (Tenant Night Switching Control) if tenanting is used to segregate departments.

Operation

None.

Toll Control

Description The toll control feature forms part of the automatic route selection (ARS) feature. It allows the PABX to restrict external calls placed by designated groups of extensions. This may mean denying all outside calls, denying calls to specific locations, denying calls over expensive routes, or any combination of these.

See *Automatic Route Selection (ARS)* and *Class of Restriction (COR)* in this section; note that this is not key system toll control.

Conditions Refer to the *Automatic Route Selection and Toll Control Practice.*

Programming Refer to the *Automatic Route Selection and Toll Control Practice.*

Operation None.

Tone Demonstration

Description The tone demonstration feature familiarizes users with the tones the system generates. This feature also allows *SUPERSET 401+*, *SUPERSET 410*, *SUPERSET 420,* and *SUPERSET 430* users to adjust ringer volume and pitch.

Conditions The following conditions apply to this feature:

- Refer to *Ringer Control* for instructions on how to adjust ringer volume and pitch.
- If no new tone is selected after a while then the tone demonstration feature is canceled and regular dial tone to allow normal dialing is returned to the extension.

Programming Assign an access code to Feature 27 (Tone Demonstration).

Operation Dial the Tone Demonstration feature access code plus the code for the tone to be played. The codes, the tones, and the main uses of each tone in the system are as follows:

- Codes 1 through 10, 16, 22 and 25 are unassigned

11 - Dial Tone

This tone is heard when dialing can start, for internal callers as well as for external callers. ARS dial tone, interrupted dial tone and transfer dial tone use this basic tone.

12 - Transfer Dial Tone

This tone is heard when dialing can start and the extension has a party on consultation hold; see *Transfer Dial Tone.* The tone is the same as interrupted dial tone.

13 - Busy Tone

This tone is heard when:

- a busy device is reached and campon to the device is not allowed (device type, COS options, etc.)
- during campon to a busy device
- devices are busy when a feature is accessed such as paging.

14 - Special Busy Tone

This tone is heard by the calling party when a busy device is reached and campon to the device is allowed. Only some types of external callers receive this tone; see *Campon.*

15 - Standard Ringback Tone

This tone is heard by the calling party when a device is being rung. This is supplied to internal and external callers.

17 - Reorder Tone

This tone is heard when an illegal operation is attempted or when an invalid number is dialed (or no number is dialed at all); see *Illegal Number Intercept* and *Vacant Number Intercept* also. This is the same tone as busy tone; however, the cadence is different.

18 - Conference Tone

This tone can be heard when the attendant is in a conference; see *Attendant Conference.*

19 - Call Waiting (Campon) Tone

This tone is heard by a busy extension when another extension camps onto the busy extension; see *Campon Warning Tone.*

20 - Intrusion (Override) Tone

This tone is heard by a device overriding a call; see *Override (Intrude).*

21 - Interrupted Dial Tone

If call forwarding or Do Not Disturb is activated at a set, a user hears interrupted dial tone followed by regular dial tone when the user goes off-hook. Interrupted dial tone is regular dial tone broken by short intervals of silence.

23 - Waiting Tone (Silence or Music-on-Hold)

This is the tone supplied to devices on hold or consultation hold and to devices camped on to busy devices, see *Music-on-Hold* and *Campon.* Industry-standard telephones do not hear this tone.

24 - Paging Tone

This tone is heard when the pager is successfully accessed; see *Paging.*

26 - Trunk Camp-on Double Beep Tone

This tone is heard by an extension when an external call or the attendant camps onto the extension; see *Campon Warning Tone.*

27 - ARS Warning Tone

This tone is heard when an expensive route is accessed in ARS; see *Expensive Route Warning.*

28 - ARS Dial Tone

This tone is heard if dial tone is supplied from ARS during an outbound call. Refer to the *Automatic Route Selection and Toll Control Practice.*

29 - Override Warning Tone

This tone is heard by parties in a call while a conversation is being overridden; see *Override (Intrude).*

30 - Privacy Release Tone

This tone is heard when an extension enters an existing call on a key, direct trunk select, or private trunk line; see *Privacy Enable/Privacy Release.*

31 - Auto-Answer Call End Tone

This tone is heard when a *SUPERSET* telephone that has the auto-answer feature enabled ends a call; see *Auto-Answer.*

32 - Attendant Error Tone

This tone is heard when the attendant attempts an illegal operation.

33 - Ringer Pitch Adjust

Dialing this code starts the *SUPERSET 401+, SUPERSET 410, SUPERSET 420,* or *SUPERSET 430* telephone ringing, and enables the ringer pitch adjust mode for the volume keys on the telephone. Refer to the *Peripherals Devices Practice.*

Tone Plans

Description

Numerous tones can be generated when dialing telephone numbers in North America. To accommodate these, the system loads the North American tone plan from the PCMCIA Flash card.

Refer to *Call Progress Tones*, in the *Engineering Information Practice*, for details of North American tone plans.

Conditions

None.

Programming

Tone plans are selected via System Option 60 (Tone Plan) in CDE Form 04 (System Options and Timers).

Operation

None.

Traffic Measurement

Description

Traffic measurements can be made on *SX-200* ML PABX systems, and the results printed through a printer port (see *Directed Input/Output)*. The types of measurements made include the following:

- DTMF receiver activity
- DTRX calls
- Console activity
- Feature activity
- Guest room activity
- Hunt group activity
- Line and trunk activity
- PCM channel activity
- Trunk activity
- Trunk group activity.

Information is accumulated during a user-programmed time period, and is then available for output. Programming is done from the maintenance terminal or from the attendant console.

Refer to the *Traffic Measurement Practice.*

Conditions

None.

Programming

Refer to the *Traffic Measurement Practice.*

Operation

None.

Transfer

Description

This feature allows a telephone user, on an established call, to put the call on consultation hold, dial a third party, and transfer the second party to the third party. The transfer can be done before the third party answers, after the third party answers, or if the third party is busy (Transfer Into Campon).

A *SUPERSET* telephone user can consult privately with each party using the Swap feature (see *Swap*) and then form a conference or perform a transfer.

The user of an industry-standard telephone can consult privately with the third party only once prior to transferring or conferencing.

Conditions

The following conditions apply to this feature:

- When an extension with a consultation hold in progress calls the console and the console answers, the console has the caller as the DEST party and the party held by the caller as the SOURCE party. The calling party no longer has the third party on consultation hold - it is on hold by the console.
- When a party is transferred to a third party while the third party is ringing, the display of the third party is updated to show the identity of the new caller.
- The *SUPERSET 420* and *SUPERSET 430* telephones provide visual indications on the display that a consultation hold is in progress. *SUPERSET 430* telephones identify the call on consultation hold.
- For transfers to campon, see *Campon.*
- Device Interconnection checking is done between the party being transferred and the party being transferred to. An exception is when a direct trunk select trunk line is being transferred.
- See *Transfer Security (Recall)* for situations where the transferring party accidentally hangs-up or attempts an illegal transfer.
- Private trunk lines cannot be transferred.
- Conferences can be put on consultation hold.
- Single parties on consultation hold receive system music.
- Calls may not be transferred to the paging circuit or to a conference.
- COS Options 223 (Flash Disable), 400 (Contact Monitor), 228 (Manual Line), 224 (Flash for Attendant) prevent an extension from putting a party on consultation hold.
- COS Option 214 (Cannot Dial a Trunk After Flashing) and COS Option 215 (Cannot Dial a Trunk if Holding or Conf with One) prevent trunks from being accessed as the third party under certain circumstances.
- An extension with COS Option 233 (Never a Consultee) may not be consulted.
- See *Transfer Dial Tone* for special dial tone on transfer.

- A call cannot be put on consultation hold if:
  - the party is an attendant,
  - the call has an incoming trunk in it and the Flash on Incoming Trunk feature is enabled (the same applies for outgoing trunks),
  - the call is on a private trunk line,
  - there is a party in the call using the ADD HELD feature enabled,
  - a party in the call is holding the call,
  - an extension in the call has the Non-Busy Extension feature,
  - the call is ringing back one of the parties in the call (Station Transfer Security),
  - another party in the call is being overridden,
  - the extension is overriding a party in the call,
  - there are five parties in the call,
  - the call is already on consultation hold,
  - the call is a conference and the extension has the Broker's Call or Transfer With Privacy feature,
  - the call is a conference and the Flash In Conference option is disabled,
  - the called party in a member of a voice mail hunt group.

Programming

See *Flash Control* for restrictions on consultation hold.

See *Conference* for programming to allow conferences of more than three parties.

Operation

Industry-standard Telephones:

On an established call:

- Flash the switch-hook - dial tone is returned; the current party (second party) is on consultation hold and hears music, if provided.
- Dial the number of the third party.
- If desired, wait for the third party to answer (if the number is busy or does not answer, flash the switch-hook to return to the held call).
- If desired, consult privately with the third party.

***SUPERSET 401+* Telephones:**

On an established call:

- Press the FLASH key - transfer dial tone is returned, the current party (second party) is on consultation hold and hears music, if provided.
- Dial the number of the third party.
- Replace the handset,
  or,
  If desired, wait for the third party to answer and provide an introduction. If you receive busy tone, or if the called party doesn't answer, hang up. You will be recalled by the original party.

***SUPERSET 410* and *SUPERSET 420* Telephones:**

On an established call:

- Press the TRANS/CONF key - transfer dial tone is returned, the current party (second party) is on Consultation Hold and hears music, if provided.
- Dial the number of the third party.
- Replace the handset,
  or,
  If desired, wait for the third party to answer and provide an introduction (if the number is busy or does not answer, press the CANCEL key to return to the held call).
- If desired, switch between parties by pressing the SWAP feature key. The conversation with each party is private.
- To connect both parties: hang up, or press the RELEASE softkey (if provided).

***SUPERSET 430* Telephones:**

On an established call:

- Press the TRANS/CONF softkey - transfer dial tone is returned. The current party (second party) is on consultation hold and hears music, if provided.
- Dial the number of the third party.
- If desired, wait for the third party to answer. (If the number is busy or does not answer, press the BACK TO HELD softkey to return to the held party. If another call is desired without returning to the held party (e.g., wrong number), use the NEW CALL softkey).
- If desired, consult privately with this party.
- If desired, switch between parties by pressing the TRADE CALLS softkey. The conversation with each party is private.
- To connect both parties: hang up, or press the RELEASE ME softkey.

Transfer Dial Tone

Description

The transfer dial tone feature supplies a tone to indicate that an extension has a call on consultation hold. Transfer dial tone is returned when an extension places an established call on hold to consult with another party or to transfer the call. Transfer dial tone is 350/440 Hz, three bursts of 100 ms on, 100 ms off, followed by continuous tone. Regular dial tone is 350/440 Hz continuous tone.

Conditions

The following conditions apply to this feature:

- The console is not affected.
- COS Option 701 (No Dial Tone) must be disabled.
- The *SUPERSET 420* and *SUPERSET 430* telephones have indicators on the display to show that there is a consultation hold in progress.

- This is the same tone used for discriminating dial tone.

Programming Enable COS Option 251, Transfer Dial Tone for the extension.

Operation Establish a call.

Place call on consultation hold. Transfer dial tone is returned to indicate consultation hold.

Transfer Security (Recall)

Description This feature is designed to prevent the dropping of mishandled calls. If an extension, during transfer, hangs up before completing dialing, or if the transfer is not allowed, the call that was placed on hold by the original extension flashing, automatically calls back to that extension. This also applies to conference calls.

Conditions The following conditions apply to this feature:

- See *Inhibit Trunk Ring-me-back During Dialing* for an exception for this feature.
- *SUPERSET* telephones with displays indicate the special ring-me-back calls.
- The ring back only affects the party being rung back - other appearances of the line do not ring.
- The features available to extensions ringing an extension (callback, messaging) are available to the extension ringing back.
- The parties in a conference ringing back do not hear any tones.
- There is no time limit on the conference ringing back.
- Forwarding - no answer and recall operate for a single caller ringing back. All types of forwarding, and recall are ignored for a conference ringing back.
- The Trunk Recall Partial Inhibit feature blocks this feature in a particular situation.
- The Directed Call Pickup feature has no effect on calls ringing back.
- Other appearances of the extension being rung back cannot answer the call.
- An attempted flash to go back to a consultation hold could be interpreted by the system (if it is too long) as a hang up attempt. The industry-standard telephone user could then possibly find itself reconnected to the held party with no ringing heard at the station.
- If an illegal transfer is attempted then the transferring extension is rung back.

Programming None.

Operation None.

Trunk Answer From Any Station (TAFAS)

Description

This feature allows the user to answer incoming calls appearing at common alerting devices (night bells). The user can answer calls for a single tenant or for all tenants in the system. The answering extension can then invoke any feature associated with the incoming call that is normally available at that extension. TAFAS can also be used to answer certain calls which ring at the console during the day.

Conditions

The following conditions apply to this feature:

- Tie trunks, industry-standard telephones, *SUPERSET* telephones, and consoles can access the feature.
- The NIGHT BELL softkey appears at the console only if the appropriate COS options are enabled for the console and a call of the correct tenant is ringing the night bell.
- Extensions cannot have a consultation hold in progress when attempting to answer the TAFAS call.
- Night Bells have no tenant assignment; the tenant used is always that of the caller.
- Device interconnection control checks must pass between the caller and the device answering the call.
- All calls ringing all consoles, LDN keys, or night bells in the system are available, depending upon the tenant checks made.
- Recall is handled as if the caller had directly dialed or originated to the extension and the extension had answered; see *Recall.*

Programming

To restrict access to the device's tenant only, assign an Access Code to Feature 15, TAFAS - Local Tenant and enable COS Option 249, TAFAS Access - Tenant for this device.

To allow access to all tenants, assign an access code to Feature 14, TAFAS - Any) and enable COS Option 248, TAFAS Access - Any for this device.

If desired, enable COS Option 250, TAFAS Access During Day Service; this permits set users to answer calls ringing the console during the day.

Operation

Operation varies depending upon the device type as described below.

Industry-standard and *SUPERSET 401+* Telephones:

To answer a TAFAS call:

- When the alerting device is heard, go off-hook.
- When dial tone is heard, dial the appropriate TAFAS code.

The incoming call is now connected to the extension.

***SUPERSET 410* and *SUPERSET 420* Telephones:**

To answer a TAFAS call (in the local Tenant group) or any TAFAS call:

- When the alerting device is heard, go off-hook.
- Press the NIGHT ANSWER feature key.

The incoming call is now connected to the extension.

***SUPERSET 430* Telephones:**

To answer a TAFAS call (in the local Tenant group) or any TAFAS call:

- When the alerting device is heard, go off-hook.
- Press the NIGHT ANSWER softkey.

The incoming call is now connected to the extension.

Attendant Console:

To answer a TAFAS call at the console:

- When you hear the alerting device, press the NIGHT BELL softkey.

The incoming call is now connected to the console.

Trunk Circuit Descriptor Options

Description Trunk circuit descriptors specify the programmable hardware parameters of each trunk circuit. The system supports a maximum of 25 different trunk circuit descriptors. Each trunk in the system must have a trunk circuit descriptor number with an associated set of selected options.

Conditions None.

Programming Select Trunk Circuit Descriptors in CDE Form 13 -Trunk Circuit Descriptors. If the system is to provide pseudo answer supervision, set an appropriate time for the time-out period via System Option 49 - Pseudo Answer Supervision Timer. See the *Customer Data Entry Practice.*

Operation None.

Trunk Dial Tone Detection

Description

After accessing a trunk the system tries to detect dial tone on it. If dial tone is detected before time-out, the system begins sending digits. If no dial tone is detected after the time-out period and limited wait is specified, the system automatically begins sending digits. Otherwise it removes the trunk from service and indicates an alarm. Dial tone detection, without the alarm, can also occur in the middle of trunk dialing; see *Automatic Route Selection.*

Conditions

The following conditions apply to this feature:

- Enabling System Option 48 (Limited Wait For Dial Tone) disables COS Option 805 (Trunk No Dial Tone Alarm).
- The no dial tone alarm time is set at 10 seconds.
- When the trunk is removed from service, the party dialing on the trunk ends dialing and is given busy tone.
- 50 simultaneous dial tone detections can occur in the system at any one time.
- If dial tone detection resources are not available and the alarm is enabled then the trunk is removed from service. If the alarm is not enabled then dial tone is assumed to have been supplied and dialing commences after a short pause.

Programming

Set System Option 48 (Limited Wait for Dial Tone) for 1-15 seconds.

To program a trunk to wait the limited time for dial tone, select COS Option 802 (Limited Wait for Dial Tone) for the trunk.

To enable the no dial tone alarm for a trunk, enable COS Option 805 (Trunk No Dial Tone Alarm).

Operation

None.

Trunk Groups

Description

Trunk groups are defined and used in the ARS forms in CDE to control extension access to trunks, to define trunk options, and to apply features to trunk groups. For further information, refer to the *Automatic Route Selection and Toll Control Practice*, and to the *Customer Data Entry Practice.*

Conditions

The following conditions apply to this feature:

- SMDR is applied to trunks on a per trunk group basis; refer to the *Station Message Detail Recording Practice.*
- A maximum of 50 individual trunk groups are available.

- A maximum of 50 trunks are permitted in an individual trunk group.
- A trunk may be a member of only one trunk group.
- Individual trunks must be programmed before they are entered into trunk groups.
- Direct trunk select and private trunk line trunks must be in trunk groups before outgoing calls can be made on the trunk lines.
- Direct trunk select and private trunk line trunks are never hunted from a trunk group.
- A trunk is selected from a trunk group if it is idle, if it is not being seized by another party, if a no seize alarm is not pending and if the caller and the trunk can connect together.
- A trunk does not need to be in a trunk group to originate a call.
- Trunk groups can have circular or terminal hunting.

Programming Program trunks into Trunk Groups via CDE Form 16 (Trunk Groups).

Each trunk group may also be given a unique name via CDE Form16 (Trunk Groups).

System Option 49 - Pseudo Answer Supervision Timer applies to all trunks in the system.

Refer to *the Automatic Route Selection and Toll Control Practice.*

Operation None.

Trunk Operation - Direct Inward Dial (DID)

Description The DID trunk type is one of the four trunk types, independent of the hardware actually used to support the types, used in the system.

DID trunks allow incoming trunk calls to reach extensions without attendant intervention or assistance. The length of the incoming number, the number of digits to be absorbed, and a prefix digit, if required, can also be specified through CDE programming.

Calls arriving at the PABX on DID type trunks are assumed to be outside calls. Callers therefore receive different call progress tones. Call handling differs from tie and DISA trunk type calls, which are assumed to be internal calls.

For the hardware that supports the DID trunk type operation, see *Trunk Support - Direct Inward Dial (DID), Trunk Support - T1*, and *Trunk Support -E&M.*

Also see *DID/Dial-In/Tie Intercepts.*

Conditions

The following conditions apply to this feature:

- If the DID trunk sends less digits than expected, the trunk receives reorder tone from the PABX system after the inter digit time-out (15 s).
- DID trunks can dial any access code for any type of device.
- DID trunks can dial Account Codes if running software loads F41.0 and above.
- An extension with Option 226 (Inward Restriction DID) in its COS cannot receive a call directly from a DID trunk. It is treated by the system as an illegal number.
- Calls attempted using vacant or illegal numbers can be routed to answering points for completion; see *Illegal Access Intercept and Vacant Number Intercept.*
- DID trunks receive ringback tone when calling and camping onto a busy device. They receive busy tone if they reach a busy device and no busy intercept occurs (see *DID/Dial-In/Tie Intercepts*) and the trunk cannot camp on (see *Campon)*. If it can campon, it does so immediately.
- DID trunks ignore the DND feature if enabled on the called extension if there is no Do Not Disturb Routing for the called extension's tenant; also see *DID/Dial-In/Tie Intercepts.*
- DID trunks are answered before being connected to a recording device.
- If there is no DID rerouting, DID trunks ignore the DND that is enabled on the called party.
- The attempt to call the destination is done only when the specified number of digits to be received on the trunk have been received. Extra digits are ignored and no matches are done for access codes that contain less than the specified number of digits.
- Digits are absorbed before the prefix digits are inserted.
- The trunk may be programmed to ignore incoming DTMF digits, and recognize only rotary digits.
- A DTMF receiver is needed when a DID trunk originates. If there is no receiver available then the trunk waits indefinitely until a receiver is available. Digits received on the trunk are stored and processed when a receiver is available. The Traffic Measurement feature can be used to track the wait for receiver.
- If a seize error occurs on a DID trunk then the trunk is given reorder tone.
- Prefix digits are used for matching access codes only after the DID trunk has dialed at least one digit that is not absorbed and the expected number of digits have been received. The prefix digits by themselves cannot be used to match an access code.

Programming

To program a trunk to ignore incoming DTMF digits during origination of the call, enable COS Option 801 (Incoming Trunk Call Rotary) for the trunk.

Program a DID trunk in CDE Form 15 (Dial-in Trunks).

Operation

None.

Trunk Operation - Direct Inward System Access (DISA)

Description

The DISA feature allows an external caller to access the system by dialing the directory number of a special DISA trunk and then dialing a security code. After the code is dialed, the system returns dial tone to the caller, who may then access features in the DISA trunk's COS except those which require a switch-hook flash.

Optionally, the external caller can be forced to enter a special account code rather than the standard DISA security code. See *Verified Account Codes (Special DISA).*

DISA trunks can be supported on many different hardware types. See T*runk Support - T1, Trunk Support - Tie, Trunk Support - E&M.*

A trunk can be programmed as DISA at all times, or during night service only.

Conditions

The following conditions apply to this feature:

- After the trunk originates, the system waits a programmable time, and then answers the trunk; then dial tone is returned.
- After the preprogrammed wait, the system answers the trunk before dialing begins.
- The outside caller must use a DTMF telephone.
- The security code may be one to five digits in length.
- The same security code applies to all DISA calls.
- If a caller dials an invalid code, the call is dropped only after three digits have been dialed. This makes it a bit more difficult for unwanted callers to determine what the security code is. This does not apply when a verified account code is used.
- Three service modes -- DAY, NIGHT1, or NIGHT2 -- are available to tenant groups (see *Night Services*). If COS Option 810 (DISA During Night Service Only) is enabled for a trunk, the DISA feature is disabled for the trunk's tenant during DAY service mode. Therefore, during DAY service mode, an external caller can call in on the trunk without entering the DISA security code.
- Reorder tone is not returned to the caller when an invalid security code is dialed.
- A switch-hook flash is not possible on a DISA trunk.
- DISA trunks follow the illegal, do not disturb and vacant number routing as an internal industry-standard telephone would.
- A DISA trunk is given the same call progress tones as an internal extension.
- The Alternate Trunk Recall feature applies to DISA trunks.
- If a loop start CO trunk is the hardware supporting the trunk then loop start interconnection rules apply and the restrictions on loop start CO trunks applies; see *Device Interconnection Control* and *Trunk Operation - Non-dial-in.*

- If the DISA trunk caller cannot complete a call then the caller must hang up and try again.
- If there is no DTMF receiver available for dialing when the DISA trunk originates a call, the initial answer delay period is extended indefinitely until a DTMF receiver is available.
- No call progress tone is provided by the PABX to the trunk in the initial answer delay period.
- Up to seven digits can be absorbed from the trunk.
- Access to the allowed features is controlled by the COS assigned to the trunk.
- The features available to DISA trunks include:
  - account codes
  - ARS
  - console hold slot retrieve
  - paging
  - system abbreviated dial.

Programming

If the trunk is to be a DISA trunk during Night Service only, enable COS Option 810 (DISA During Night Service Only) for the trunk, and set the DISA trunk DAY service routing in CDE Form 19 (Call Rerouting Table) for the trunk's tenant.

Select options for the DISA trunk type via CDE Form 13 (Trunk Circuit Descriptors).

Select options for the specific DISA trunk via CDE Form 15 (Dial-In Trunks). Assignment of a DISA circuit descriptor to a trunk changes the trunk to a DISA trunk.

Assign a DISA access code to Feature 19 (Direct Inward System Access); see *Attendant DISA Code Setup.*

Select the DISA answer timer via System Option 54(DISA Answer Timer: 1-8 Seconds).

Enable COS Option 812 (Loop Start Trunk to ACD Path Connect) to allow loop start DISA trunks to access the ACD feature.

Operation

To access the System:

Dial the required directory number from a DTMF telephone.

The system waits the DISA answer time before answering the trunk and supplying dial tone.

Dial the DISA security code - dial tone is returned again.

Dial the required feature access code or extension.

Trunk Operation - Non-Dial-in CO

Description

CO trunks usually carry calls between the local central office and the PABX. Calls arriving on CO trunks are assumed to be outside callers. Callers therefore receive different call progress tones. Call handling differs from tie and DISA trunk type calls, which are assumed to be internal calls.

CO trunks are assigned an origination point for DAY, NIGHT 1, and NIGHT 2 service. They can optionally be assigned as a dedicated line on a *SUPERSET* telephone.

For the hardware that supports CO trunk type operation, see *Trunk Support - CO (LS/GS), Trunk Support - Tie, Trunk Support - DID, Trunk Support -E&M, and Trunk Support - T1.*

The NIGHT1 or NIGHT2 service for CO trunks can be changed directly from the Attendant console; see *Night Service Flexible.*

Also see *Direct-In Lines.*

Conditions

The following conditions apply to this feature:

- A DISA trunk may be programmed to be a CO trunk type during day service. See *Trunk Operation - Direct Inward System Access (DISA*).
- Connection checking is done between the trunk and the destination point when the trunk originates.
- If the trunk origination is blocked then the trunk stays idle and the PABX does no further processing of the call.
- Trunks are always answered before listening to a recording device.
- The processing of the origination follows the operation of call rerouting; see *Call Rerouting.*
- For trunks assigned to DTS or private trunk lines, the trunk originates to the line and ignores the NIGHT/DAY points programmed.
- The audio heard by the caller before answer depends upon the hardware/circuit descriptor connected to that trunk. For CO trunks, CO tones provided by the central office are heard; for trunks, based on dial-in hardware circuits, PABX call progress tones provided by the PABX are heard.
- The trunk receives ringback tone when it camps on to a device.
- The Device Interconnection feature can be used to prevent loop start trunk interconnection.
- If a CO trunk in a bay fails to get resources required for a call when it originates, the trunk will wait for resources to become available.
- If the trunk is routed out to an external trunk call and the CO trunk has not been answered yet, the CO trunk is answered when dialing is finished on the outgoing trunk.
- CO trunks cannot dial ARS directly; however, ARS can be accessed via system abbreviated dial.

- If no routing points are programmed in form 14 and there is no key appearance of the trunk, the trunk will not ring.

Programming Select options for the specific trunk circuits via CDE Form 14 (Non-Dial-In Trunks) for incoming calls.

Enable COS Option 812 (Loop Start Trunk to ACD Path Connect) to allow loop start trunks to access the ACD feature.

Refer to the *Automatic Route Selection and Toll Control Practice*, for the selection of options for outgoing calls.

Operation None.

Trunk Operation - Tie

Description The Tie trunk is one of four trunk types, and independent of the hardware actually used to support the types, used in the system.

Tie trunks allow incoming trunk calls to reach extensions directly, without attendant intervention or assistance. The number of digits expected from the trunk is unknown. Digit absorption and adding prefix digits can be done.

Calls coming into the PABX on Tie type trunks are assumed to be callers from inside the company, similar to DISA trunk type calls. The callers therefore receive the same call progress tones that internal callers hear and may have access to many extension features.

For the hardware that supports the Tie trunk type operation, see *Trunk Support - E&M, Trunk Support - T1, and Trunk Support - Direct Inward Dial (DID).*

See *Dial Tone Disable* for dial tone control for the Tie trunk.

Conditions The following conditions apply to this feature:

- Tie trunks have access to the following extension features:
  - account code
  - ARS
  - console hold slot retrieve
  - directed pickup
  - hold retrieve
  - paging
  - system Abbreviated Dial.
- For Tie intercept handling, see *DID/Dial-In/Tie Intercepts.*

- Tie trunks dialing extensions with Do Not Disturb enabled are handled in the same way as extensions.
- Tie trunk dialing follows the same rules as for extensions.
- Up to 7 digits can be absorbed and 2 digits can be prefixed.
- The prefix digits are used for matching access codes only after the Tie trunk has dialed at least one digit that is not absorbed. The prefix digits by themselves cannot be used to match an access code.
- The limit to the number of digits that a Tie trunk can dial is the same as for an internal extension (25). This includes the prefix digits but not the absorbed digits.
- The trunk may be programmed to ignore incoming DTMF digits, and to recognize only rotary digits.
- A DTMF receiver is needed when a Tie trunk originates. If there is no receiver available then the trunk waits indefinitely until a receiver is available. Digits received on the trunk are stored and processed when a receiver is available. The Traffic Measurement feature can be used to track the wait for a receiver.
- For incoming answer supervision when directed to ARS, see *Tandem Operation.*

Programming

To program a trunk to ignore incoming DTMF digits, enable COS Option 801 (Incoming Trunk Call Rotary) for the trunk.

Operation

None.

Trunk Recall

Description

The feature provides an alternate recall point for trunks in the system. The alternate recall point can be specified for each tenant and each NIGHT/DAY service. Under the following conditions, trunks are rerouted to the alternative call point:

- for all trunk types, when an extension with a trunk on consultation hold is listening to reorder tone and times out, the trunk is removed from consultation hold and rerouted.
- for DISA and CO trunks, when a trunk recalls from campon or ringing an extension; see *Recall.*

Conditions

The following conditions apply to this feature:

- This feature does not apply to DTS or private trunks ringing into the *SUPERSET* telephone where the line appears.
- For the reorder tone case, the reroute is done before recalling serial trunks.
- For the reorder tone case, the tenant group of the extension is used to determine the routing point.

- For the reorder tone case, if there is no point programmed for the current NIGHT/DAY service then the trunk is dropped; Serial trunks recall.
- An LDN key cannot be the routing point.
- For recall situations, the tenant of the called party is used to determine the rerouting point. When a logical line is called, the tenant of the first appearance of the line is used. When a hunt group is called, the tenant of the first member of the hunt group is used.
- The feature does not operate when calling an LDN, Night Bell, or console.

Programming In CDE Form 19 (Call Rerouting Table), program the recall points for Non-Dial-In Trunk Alternate Recall Point in the tenant of the called party.

Operation None.

Trunk Support - CO (LS/GS)

Description The system supports CO (LS/GS) trunks with the LS/GS trunk card.

For loop start CO trunks that do not provide release supervision, take caution when routing to auto-answer telephones or to UCD/ACD applications that do not provide termination for a music/recording sequence. In these cases, since the system does not disconnect the trunk, the call may stay up indefinitely.

Conditions None.

Programming Enter the trunk into the system's physical configuration table via CDE Form 01 (System Configuration).

Select options for the CO trunk type via CDE Form 13 (Trunk Circuit Descriptors).

Select options for the specific trunk circuits via CDE Form14 (Non-Dial-In Trunks) for incoming calls.

Refer to the *Automatic Route Selection and Toll Control Practice*, for the selection of options for outgoing calls.

Operation None.

Trunk Support - Direct Inward Dial (DID)

Description The following types of DID trunks are supported:

- Wink Start
- Delay Dial
- Immediate Dial.

DID trunks support Tie, CO, DID, and DISA operation.

Conditions The following conditions apply to this feature:

- Digital DID trunk cards are incoming only.
- Digital DID trunk cards do not support DISA type operation.

Programming Enter the trunk into the system's physical configuration table via CDE Form 01 (System Configuration).

Select options for the DID trunk type via CDE Form 13 (Trunk Circuit Descriptors).

Select options for the specific trunk circuits via CDE Form 15 (Dial-In Trunks).

Operation None.

Trunk Support - E&M

Description E&M trunks are supported with the E&M Trunk module on the Universal Card.

The signaling schemes supported include: type I and type V, 2-wire or 4-wire.

E&M trunks support Tie, CO, DID, and DISA operation.

Conditions None.

Programming Enter the trunk into the system's physical configuration table via CDE Form 01 (System Configuration).

Select options for the E&M trunk type via CDE Form 13 (Trunk Circuit Descriptors).

Select options for the specific trunk circuits via CDE Forms 14 (Non-Dial-In Trunks) and CDE Form 15 (Dial-In Trunks).

Operation None.

Trunk Support - T1

Description

T1 trunks are supported using T1/D4 channel associated signaling (CAS), also referred to as DS-1.

T1 trunk cards support DID, tie, CO and DISA operation on a per circuit basis. For each circuit, the circuit descriptor can be programmed through CDE to alter the signaling scheme to one of E&M, DID loop-tie, CO (loop and ground start), DISA E&M, DISA DID loop-tie, DISA CO (loop or ground start).

The PABX provides for a Stratum 3 or 4 clock source. The PABX can be used in master mode to serve as a clock source for the network, or in slave mode to use the network as its clock source. In slave mode the system prevents data losses due to clock rate differences by adjusting its internal T1 clock module to remain in phase with the incoming frame clock rate.

T1 Trunk Maintenance

The PABX automatically monitors various errors on a link and maintains a 24-hour sliding window for each link. Thresholds are programmable on a link basis for these errors. When the thresholds are exceeded, based upon the threshold type (maintenance, service, or network synchronization), the PABX generates maintenance logs, removes the link from service, or causes the link to not be used as a network synchronization source. Once the link is removed from service or is removed from network synchronization, a programmable timer starts. When this time is elapsed, if the link does not exceed a specified number of errors, the link is again made available for use. All of this activity is logged in maintenance and the cumulative error counts (24-hour basis) are available in maintenance error reports (refer to the *RS-232 Maintenance Terminal Practice*).

There are two different states for a T1 trunk that is out of service, a yellow or red alarm. A yellow alarm is the result of a link-based event. The link transmits a yellow alarm condition on the link, which signals the far end that the link is out of service. This is the action taken on the service threshold limits or the reception of a yellow alarm condition from the far end.

A red alarm is based upon loss of synchronization on the link. Maintenance logs are generated for any change in alarm status, and can be seen by using the maintenance SHOW STATUS command.

A link monitoring tool is available for troubleshooting T1 links, and for specifying a manual network synchronization source. Refer to the, *RS-232 Maintenance Terminal Practice.*

T1 Trunk Synchronization

The PABX provides for Stratum 3 or 4 clock source, and can be used in master or slave mode. There is a programmable list of links to be used in network synchronization, with each link backing up the other links, should the previous link exceed the error thresholds (Slip Rates, Bit Error Rates (BER), and Framing Losses) programmed in the link descriptors. There are three network synchronization modes in the PABX: Auto, Manual, and Freerun. Auto mode is when the PABX chooses the network synchronization source automatically. Manual mode is when a link is specifically selected as the network synchronization source (this only lasts for 24 hours, at which point the system reverts to auto or freerun mode). Freerun mode is when there is no clock source or no link to be used as a network synchronization source. The PABX changes modes based on user actions, thresholds, and events on the T1 links. Maintenance logs notify the installer of these changes.

Conditions

The following conditions apply to this feature:

- There are 12 link descriptors available.
- There can be up to 8 network synchronization sources for network synchronization.
- One link provides 24 trunks.
- If a link descriptor is not assigned (it is recommended that one is assigned) then the default values in the link descriptors will apply. A link cannot be a network synchronization source without a link descriptor.
- If the last available network synchronization source specified is removed from service then the PABX is put into free-run mode.
- When a network source link is returned to service or is again ready for use as a network synchronization source, then the system will review the list of network synchronization sources and possibly select the returned link (depending upon the current network synchronization source and the order of the links in the list).
- A link is removed from service immediately, regardless of activity on the link, when its service threshold limits are exceeded.
- The T1 Link Monitor feature is not available in maintenance until at least one circuit is programmed (CDE forms 14 or 15) on a link in the PABX.
- The T1 Clock Module clock rate is adjusted by the system if there are entries made in CDE Form 44 (Network Synchronization) and one of the links there is available. If no network synchronization source is available, then the clock rate will not be adjusted by the PABX.
- The threshold error counts are cleared when the T1 card is plugged in or any of the values in the link descriptor assigned to the link are changed in CDE Form 42 (T1 Link Descriptors). A change in link status has no effect.
- Changing the entries in CDE Form 44 (T1 Network Synchronization) will override manual mode and change the PABX mode to freerun or auto.
- The term link is synonymous with a T1 trunk card.
- The normal maintenance busy-out feature applies to a circuit on a link and not the link itself. The link remains active but the circuit(s) are disabled.

Programming

Enter the trunk into the system's physical configuration table via CDE Form 01 (System Configuration).

Specify options for the T1 link descriptors via CDE Form 42 (T1 Link Descriptors).

Specify options for the T1 trunk types via CDE Form 13 (Trunk Circuit Descriptors).

Specify options for specific trunk circuits via CDE Forms 14 (Non-dial-in Trunks) and 15 (Dial-in Trunks).

Assign link descriptors to the various T1 links via CDE Form 43 (T1Link Assignment).

If the T1 clock module is used, assign the primary and backup T1 links to be used as sources for network synchronization via CDE Form 44 (T1 Network Synchronization). The order that the links are programmed in the form gives the order in which the links are used as sources.

Operation

None.

Uniform Call Distribution (UCD)

Description

Uniform Call Distribution (UCD) concentrates incoming trunk traffic onto one or more special agent hunt groups. Trained operators (agents) answer the calls. If all agents are busy, the caller camps on and may be connected to a recording hunt group, where the caller hears recorded announcements. The caller retains his position in the queue. If the agents are still busy when the recording ends, the system connects the call to Music-on-Hold (if provided). After a pre-determined time, the unanswered call is rerouted to a designated answering point.

The call rings an agent immediately when one is available.

Agents can log in and out of the agent hunt group to control the arrival of calls from the agent hunt group.

When a caller is connected to a Recorded Announcement Device (RAD), full two-way audio is provided. This allows the caller to leave a message with the RAD device (if desired), while waiting for the UCD group.

See *RAD Support* for details on recording devices.

For night service handling, trunk calls can be routed to a recording device directly; see *Intercept To Recorded Announcement.*

Conditions

The following conditions apply to this feature:

- If there is no hold timeout point and recordings are given, then for loop start trunks that do not provide release supervision, the system does not disconnect the trunk; the trunk remains on Music-on-Hold. It is important to provide a routing point for these unanswered calls to ensure that the loop start trunk is either answered, or timed out for no answer and then released.

- Agent hunt groups are special hunt groups; hunt group conditions plus some additional features apply to them.
- Circular hunting is advised for the agent hunt group to distribute the calls in the group. There is no distribution of calls based on workload or waiting time; see *Automatic Call Distribution (ACD).*
- Campon to agent hunt groups follows the normal campon to hunt groups. The recording and reroute features are available once a caller has camped on to the agent hunt group.
- Features available for hunt groups, such as Trunk Campon Warning Tone and Swap Campon, are available to agent hunt groups.
- The recording hunt group is selected from the tenant of the first programmed member of the hunt group.
- If there is no recording programmed, then no music is provided; it is only provided after the recording.
- The system provides a recording for the caller as soon as a recording device is available.
- At least one hunt of the agent hunt group is done before the waiting timeout timer is started (regardless of the waiting timeout length).
- If the waiting timeout time is set to 0 seconds then the timeout effectively is an overflow busy point for the agent hunt group.
- A reroute is done for the waiting timeout.
- If the caller is listening to a recording at the time of the waiting timeout, the recording is terminated before the reroute is done. If the waiting timeout occurs while listening to music and if a campon is done on the reroute point then music is maintained while waiting for the busy reroute point. Otherwise it is removed and normal call progress tones provided.
- The waiting timeout is ignored if the caller is ringing an agent at the time.
- If the waiting timeout occurs while ringing a recording device then the ringing recording device stops ringing and the caller is rerouted to the timeout point.
- By default all agents are logged into an agent hunt group.
- Logging out of an agent hunt group only prevents calls from being presented to the agent set from the agent hunt group. Other features are not affected.
- An agent that is logged out will not be selected to receive campon beeps and the swap campon capability for the hunt group; see *Swap Campon And Campon Warning Tone.*
- When an agent becomes available and is rung by a waiting caller, the recording or music is removed and the caller hears ringback tone.
- UCD agent hunt groups and ACD agent groups are completely separate and different entities. The first is a collection of devices, and the second is a collection of agent id codes.
- The campon warning tone can be turned off for members of the agent hunt group; see *Campon Warning Tone.*

Programming

Enter a series of extension numbers into a hunt group in CDE Form 17 (Hunt Groups). Press the GROUP TYPE softkey, followed by the AGENT softkey.

Program a recording hunt group in CDE Form 17 (Hunt Groups); see *Recording Support.*

Enter the extension of the recording device hunt group into CDE Form 19 (Call Rerouting Table), under "UCD Recording Routing For This Tenant". The tenant number is that of the first programmed member of the agent hunt group.

Enter an On Hold Time-out period via COS Option 256, UCD Music-on-Hold Timer, for each incoming trunk.

Enter an On Hold Time-out answering point extension into CDE Form 19 (Call Rerouting Table), under "UCD Time-Out Routing For This Tenant". The tenant number is that of the first programmed member of the agent hunt group.

Assign an access code to Feature 38 (UCD Login/Logout Code) in CDE Form 02 (Feature Access Codes).

Enable COS Option 301 (Campon) for the incoming trunk to camp onto the UCD group.

Operation

To log in at an agent set:

- Dial the UCD Login/Logout Code, followed by 1.

To log out at an agent set:

- Dial the UCD Login/Logout Code, followed by 2.

Vacant Number Intercept

Description

Calls to unassigned (vacant) access codes can be routed to a given answering point for completion. This point can be an LDN position on the Attendant Console (see *Console LDN Keys*) or any valid routing point. Vacant number intercept points can be programmed to be different or the same for DAY, NIGHT1, and NIGHT2 modes of system operation.

See *DID/Dial-In/Tie Intercepts* for the same feature for DID and Tie trunks.

Conditions

The following conditions apply to this feature:

- If the required programming is not done, such calls receive reorder tone.
- Telephones, DISA trunks, and CO trunks only are routed to the answerpoint.
- See *DID/Dial-In/Tie Intercepts* for vacant number handling for DID and Tie trunks.

- If the call is routed to a console, the call is shown as an intercept call at the console.

Programming To cause all calls to vacant numbers to be routed to a specific answering point, program CDE Form 19 (Call Rerouting Table) with the desired answering point access code, in the appropriate column for the "Station Vacant Number Routing For This Tenant" call type.

Operation None.

Voice Mail - COV Port

Description Voice Mail devices may use the MITEL COV interface. For further information, refer to document *9150-953-003-NA, VX Voice Processor Installation and Repair Manual VX-200/VX-400/VX-800 Systems.*

Conditions None.

Programming Program the COV Voice Mail Port(s) with the following Class of Service Options:

COS Option 212 (Can Flash if Talking to an Incoming Trunk)

COS Option 213 (Can Flash if Talking to an Outgoing Trunk)

COS Option 216 (Data Security)

COS Option 229 (COV Voice Mail Port)

COS Option 238 (Override Security)

COS Option 259 (Message Sending)

COS Option 301 (Camp-on)

COS Option 604 (PBX *SUPERSET* Telephone-Automatic Outgoing Line)

COS Option 606(*SUPERSET* Telephone-Enhanced Answering Position)

COS Option 609 (*SUPERSET* Telephone - Night Service Switching)

Note: Disable all other options including Call Forwarding

Define Hunt Group in CDE Form 17 (Hunt Groups) by programming the following fields:

EXT NUM Enter the extension number of each voice mail port that is a member of the voice mail hunt group.

ACCESS CODE The access code is the system access number for the VX Voice Processing system. Define the access code of the voice mail hunt group by using the ACCESS CODE softkey (softkey 7).

NAME Enter the name of the voice mail hunt group. Enter a name for the voice mail hunt group by using the OPTIONS softkey (softkey 4).

Note: Do not program an overflow destination for this hunt group. This will ensure that calls can camp on to the busy group.

Operation

Refer to document 9150-953-003-NA, VX Voice Processor Installation and Repair Manual VX-200/VX-400/VX-800 Systems.

Voice Mail - ONS Port

Description

This feature integrates an *SX-200* PABX with an ONS Voice Mail system. The integration is based on the use of system abbreviated dial numbers. This eliminates several dialing steps involved in the sending and retrieving of voice mail messages.

Special codes allow the PABX operation to be customized to suit the operation of the particular voice mail system. Refer to the programming instructions for Form 31 - System Abbreviated Dial Entry, for a description of these special codes.

Conditions

The following conditions apply to this feature:

- This feature does not conflict with the *Voice Mail - COV Port* feature - both systems may be installed simultaneously.
- This feature does not conflict with other ONS voice mail systems - they may be installed simultaneously.
- The *1, *6, *9, codes are only operational with the ONS Voice Mail feature. If the abbreviated dial number dials anything but an ONS voice mail hunt group, the codes are ignored.
- The *4 and *8 codes are used with the *MITEL MAIL* Enhanced Inband Signaling.
- Callers hear ringback while the digits are pulsed to the voice mail system. Audio is not cut through until after the last pause has been executed.

Programming

Form 01 - System Configuration

Program the ONS Line Card.

Form 02 - Feature Access Codes

Assign access codes to:

- Feature 24 - Abbreviated Dial Access
- Feature 41 - Send Message

When appropriate for use with *4, assign access codes to:

- Feature 3 - Call forwarding - All Calls
- Feature 4 - Call forwarding - Internal Only
- Feature 5 - Call forwarding - External Only

Form 03 - COS Define

In ONS voice mail port COS, enable the following:

- COS Option 208 - Call Forwarding - External
- COS Option 212 - Can Flash if Talking to an Incoming Trunk
- COS Option 213 - Can Flash if Talking to an Outgoing Trunk
- COS Option 216 - Data Security
- COS Option 238 - Override Security
- COS Option 261 - ONS Voice Mail Port
- COS Option 301 - Camp-On

In Message Waiting/Pager Port COS, enable the following:

- COS Option 216 - Data Security
- COS Option 220 - Do Not Disturb
- COS Option 235 - Originate Only
- COS Option 238 - Override Security
- COS Option 259 - Message Sending
- COS Option 265 - Voice Mail system Speed Dial Index

Note: Include option 265 in the COS of the first member of the hunt group for Messaging - Call Me Back feature.

In the COS of the mailbox holder, enable the following:

- COS Option 231 - Message Waiting Setup - Bell, or
- COS Option 232 - Message Waiting Setup - Lamp
- COS Option 264 - Half Fwd NA timer for DID call when VM msg on (This is a toll saving option, allowing external DID callers who are calling in to check their messages to hang up if the cfna timer is not shortened, knowing they have no messages to retrieve.)

Form 04 - System Options/System Timers

Enable:

- System Option 21 - Incoming to Outgoing Call Forwarding
- System Option 22 - Last Party Clear Dial Tone

Form 09 - Stations/*SUPERSET* Telephones

Program ONS lines and assign appropriate COS.

Ensure Voice Mail Ports are in a separate COS from the Message Waiting Ports.

Form 14 - Non-Dial-In Trunks

If required, enter the Voice Mail port hunt group access code for DAY, Night1 and/or Night2 for the trunk to go to the Receptionist portion of the VM system.

Form 17 - Hunt Groups

Program a hunt group for the Message Waiting port.

Program a hunt group containing the ONS voice mail ports.

Note: The Message Waiting Hunt Group is NOT the same as the ONS Hunt Group.

Form 19 - Call Rerouting Table

For Message Port Tenant

- DND Intercept Routing for this tenant for Day, Night1, and Night2 will be routed to the Voice Mail hunt group access code.
- Station Illegal Number Routing for this tenant for Day, Night1, and Night2 will be routed to the Voice Mail hunt group access code.

Form 31 - System Abbreviated Dial Entry

Special codes entered in this form allow the PABX operation to be customized to suit the operation of the particular voice mail system. These special codes eliminate many of the dialing steps involved in the sending and retrieving of voice mail messages. The integration of the PABX and voice mail system is based on the abbreviated dial numbers shown below.

Code	Description
*1	5 second pause
*3 XX	Insert manual dialed digits (XX) 2 digits expected
*4	Send call forward condition to Voice Mail (See Note)
*6	Tone out caller extension number
*8	Send digits of calling party to Voice Mail (See Note)
*9	1 second pause
**	DTMF digit *
#	DTMF digit #
0-9	DTMF digits 0 - 9

Note: *4 and *8 are used with *MITEL MAIL* Enhanced Inband Signaling.

Features Description

Event Timing

The following example describes the feature's timing requirements:

Hunt Group Access Code	Pause A	Voice Mail System Operation Code	Pause B	Ext. no.	Pause C
1	2	3	4	5	6

1: Access code of the voice mail port hunt group.
2: Length of time required for the voice mail system to prepare to receive an optional operation code after answering the call.
3: Digit(s) indicating the type of operation to be performed voice mail system (record or playback message).
4: Length of time required for the voice mail system to receive a mail box number, after receiving the operation code.
5: Usually *6: For a forwarded call, *6 translates into the originally called party's extension number. For non-forwarded calls, *6 translates into the caller's extension number.

6: Not required with the *LIGHTWARE* 16 ML software load.

Set up the desired call forwarding for the telephones using this voice mail feature.

- Place all ONS voice mail ports in the same hunt group in Form 17 (Hunt Groups) - assign an access code to the group.
- If the ONS voice mail system requires dial tone on hang up to release the port, enable System Option 22 (Last Party Clear - Dial Tone).
- If trunk calls are to reach the ONS voice mail system via a forwarding system abbreviated dial number, enable System Option 21 (Incoming to Outgoing Call Forward) and COS Option 208 in the trunk COS.

For Message Forward:

- Enter an index number for message forward in the INDEX NUMBER field of Form 31 - System Abbreviated Dial Entry
- Enter the following in the DIGIT STRING field:
  - the ONS voice mail hunt group access code,
  - any combination of the *1 and *9 codes combining to result in an answer pause suitable to the voice mail system,
  - the voice mail system access code for the type of call forwarding which corresponds to the mailbox owner's call forwarding,
  - *6
  - the shortest *1 / *9 combination to ensure that the ONS voice mail port receives all of the digits.

865
4397

For Message Retrieve:

- Enter an index number for message retrieve in the INDEX NUMBER field of Form 31 - System Abbreviated Dial Entry
- Enter the following in the DIGIT STRING field:
  - the ONS Voice Mail hunt group access code,
  - any combination of the *1 and *9 codes combining to result in an answer pause suitable to the voice mail system,
  - the voice mail system access code for message retrieval,
  - *6
  - the shortest *1 / *9 combination to ensure that the ONS voice mail port receives all of the digits.

Operation

Forward message:

- Dial extension number and leave message.

Retrieve message:

- Dial the system abbreviated dial access code and the message retrieve abbreviated dial index number.

The PABX dials out feature access code (3, 4, or 5) plus the index number.

Valid index numbers are:

1 - Forward Always

2 - Forward on Busy

3 - Forward No Answer

4 - Forward Busy/No Answer

The PABX dials out the ANI digits if the call originated over a trunk. If ANI digits are not allowed or not available, then no digits are sent. When the call is internal, the extension's access code is dialed out.

Index

Features Description

106 (Attendant New Call Tone)
107 (Attendant Auto Call Fwd - No Answer)
108 (Attendant Audible Lockout Alarm)
109 (Attendant Serial Call)
110 (Attendant Abbreviated Dial Confidential Number Display)
111 (Attendant Abbrev Dial Programming)
112 (Attendant Station Busy-Out)
113 (Attendant Call Block Key)
114 (Attendant Trunk Busy-Out)
115 (Attendant Timed Recall - No Answer)
116 (Attendant-Timed Recall - Hold)
117 (Attendant Timed Recall - Campon)
118 (Attendant Call Fwd - No Answer Timer)
119 (Attendant Tone Signalling)
120 (Attendant Conference Disable)
121 (Attendant Station Do Not Disturb)
122 (Attendant Setup Time/Date)
123 (Attendant Call Fwd Setup and Cancel)
124 (Attendant Hold Position)
125 (Attendant Multi-New Call Tone)
150 (Subattendant Stn Setup Advisory Msg)
200 (Account Code Forced Entry-Ext Calls)
201 (Account Code Forced Entry-LD Calls)
202 (Alarm Call)
203 (Brokers Call)
204 (Call Block Applies)
205 (Flash For Waiting Call)
206 (Call Forwarding - Busy)
207 (Call Forwarding - DontAnswer)
208 (Call Forwarding - External)
209 (Call Forwarding - Follow Me)
210 (Call Fwd Inhibit on Dial-In Trunks)
211 (Call Hold and Retrieve Access)
212 (Can Flash if Talking to an Incoming Trunk)
213 (Can Flash if Talking to an Outgoing Trunk)
214 (Cannot Dial a Trunk After Flashing)
215 (Cannot Dial a Trunk if Holding or in Conference With a Trunk)
216 (Data Security)
217 (Direct to ARS)
218 (Directed Call Pickup)
219 (Discriminating Dial Tone)
220 (Do Not Disturb)
221 (Clear All Features)
222 (Call Fwd Inhibit On Hold Timeout)
223 (Flash Disable)
224 (Flash for Attendant)
225 (Hold Pickup - Attendant Paged Access)
226 (Inward Restriction -DID)
227 (Lockout Alarm Applies)
228 (Manual Line)
229 (COV Voice Mail Port)
230 (Message Register Overflow Alarm)
231 (Message Waiting Setup - Bell)
232 (Message Waiting Setup - Lamp)
233 (Never a Consultee)
234 (Never a Forwardee)
235 (Originate Only)
236 (Outgoing Trunk Callback)
237 (Outgoing Trunk Campon)
238 (Override Security)
239 (Priority Dial 0)
240 (Line Privacy)
241 (Receive Only)
242 (Repeated Campon Beeps)
243 (Non-Busy Extension)
244 (Room Status Applies)
245 (Abbreviated Dialing Access)
246 (SMDR - Extended Record)
247 (SMDR - Record Meter Pulses)
248 (TAFAS Access - Any)
249 (TAFAS Access - Tenant)
250 (TAFAS Access During Day)
251 (Transfer Dial Tone)
252 (Brokers Call With Transfer)
253 (Call Forward - Don't Answer Timer)
254 (Call Hold Recall Timer)
255 (Repeated Campon Beeps Timer)
256 (UCD Music on Hold Timer)
257 (Flash Over Trunk)
258 (Display Prime as Forwarder)
259 (Message Sending)
260 (Internal / External Split Call Forward)
261 (ONS Voice Mail Port)
262 (Ignore Forward Busy)
263 (Delay Ring Timer)
264 (Half Fwd NA timer for DID call when VM msg on)
265 Voice Mail System Speed Dial
300 (Automatic Callback)
301 (Campon)
302 (Flash in Conference)
303 - 312 (Paging Zone Access)
313 (CO Trunk to CO Trunk Connect)
314 (CO Trunk to Tie Trunk Connect)
315 (CO Trunk to DID Trunk Connect)
316 (Tie Trunk to Tie Trunk Connect)
317 (Tie Trunk to DID Trunk Connect)
318 (DID Trunk to DID Trunk Connect)
319 (Ext Non-CO Trunk To Trunk Connect)
320 (Transparent Multi-Console Operation)
326 (Acct Code Forced Entry-Data Internal)
327 (Acct Code Forced Entry-Data External)
328 (Acct Code Forced Entry-Data LD Calls)
400 (Contact Monitor)
401 (Call Park)
402 (Long Loop-Off Premise Ext Only)

D

Features Description

Features Description

9109-098-126-NA
Issue 1
March 1997

SX-200® ML PABX

Peripheral Devices

MITEL®

NOTICE

The information contained in this document is believed to be accurate in all respects but is not warranted by Mitel Corporation (MITEL®). The information is subject to change without notice and should not be construed in any way as a commitment by Mitel or any of its affiliates or subsidiaries. Mitel and its affiliates and subsidiaries assume no responsibility for any errors or omissions in this document. Revisions of this document or new editions of it may be issued to incorporate such changes.

NOTICE

1. If any of the warranty seals on any MITEL SUPERSWITCH® or SUPERSET™ products are broken, MITEL reserves the right to refuse to service the equipment or to void any remaining warranty on the equipment.
2. If any MITEL equipment has been modified so that it is not within manufacturer's specifications (including, but not limited to, the installation of any non-MITEL parts, components, or replacement boards), then MITEL reserves the right to: refuse to service the equipment; void any remaining warranty; remove and replace any non-MITEL parts found in the equipment; and perform whatever modifications are necessary to return the equipment to original manufacturer's specifications.
3. The cost for the labor and parts required to return the MITEL equipment to the original manufacturer's specifications will be charged to the customer in addition to the normal repair charge.

INDUSTRY CANADA NOTICE

This Class A digital apparatus meets all requirements of the Canadian Interference-Causing Equipment Regulations.

FEDERAL COMMUNICATIONS COMMISSION (FCC) NOTICE

This equipment generates, uses, and can radiate radio frequency energy and if not installed and used in accordance with the instruction manual, may cause interference to radio communications. It has been tested and found to comply with the limits for Class A Computing Device pursuant to Subpart J of Part 15 of FCC Rules, which are designed to provide reasonable protection against such interference when operated in a commercial environment. Operation of this equipment in a residential area is likely to cause interference in which case the user at his own expense will be required to take whatever measures may be required to correct the interference.

Table of Contents

List of Tables

List of Charts

List of Figures

1 Introduction

General

1.1 This Practice consolidates the peripheral device descriptions into one document.

Reason for Issue

1.2 This section is issued to provide a general description of current peripheral devices used with the SX-200® ML PABX.

Disclaimer

The following products have been manufacture discontinued by Mitel. These products are supported but not described in *SX-200* ML Practices:

- SUPERSET 3™ and SUPERSET 4® telephone sets
- SUPERSET 3DN™ and SUPERSET 4DN™ telephone sets
- DATASET 1101 data cartridge
- SUPERSET™ DSS module.

The following products and peripheral devices are not supported on the *SX-200* ML PABX and are not described in *SX-200* ML Practices:

- Modem Interconnect Panel
- DATASET 1102 Rack-mounted Dataset
- DATASET 2102 Rack-mounted Dataset
- DATACABINET 9000 data cabinet
- DATASHELF 9100 datashelf
- ISDN Node
- Fiber Interface Module (and associated products)
- Peripheral Node
- LCD Console (and Console module for Universal Card).

2 Attendant Console

Physical Description

2.1 There are two types of Attendant consoles.

- SUPERCONSOLE 1000™ Attendant Console, with an RS-232 printer port and a tilt display
- SUPERSET 7000™ attendant console (PC based).

SUPERCONSOLE 1000 Attendant Console

2.2 The *SUPERCONSOLE 1000* Attendant Console uses one pair of wires and interfaces to a Digital Line Card. The console has an RS-232 printer port and a tilt display.

The Attendant Console, shown in Figure 2-1, weighs 2.27 kg (5.0 lbs) and measures:

Width:	39.4 cm (15.5 inches)
Height:	10.2 cm (4.0 inches)
Depth:	22.9 cm (9.0 inches).

The console assembly consists of the following parts:

Keyboard Printed Circuit Assembly: A printed circuit board assembly with 32 keys, a dial pad (12 keys) and a cable harness. The cable harness plugs into the Console Printed Circuit Assembly.

Housing Top: The plastic moulded top of the Attendant Console housing which encloses and protects the console circuitry.

Alphanumeric Liquid Crystal Display: A 4-line x 80-character LCD which facilitates Attendant Console operation. Each character consists of a 5 X 7 dot matrix display.

LCD Adjustment: The contrast is controlled by repeatedly pressing either CONTRAST key, until the desired level is achieved.

Keyboard Designation Cover: A transparent cover on top of a printed card with silk–screened key designations.

Keyboard Layout: The console keyboard consists of a standard 12 key keypad, 10 Display Associative Keys (softkeys), four cursor control keys, two volume control keys, two contrast control keys, and 14 fixed keys, seven of which have associated LED indicators.

Within the console assembly is a printed circuit board (PCB), which holds the main console circuits, power supply circuitry and processor. Slots are provided into which the keyboard assembly, speaker and handset jack leads are plugged.

The console assembly has the following external connections:

Handset/Headset: The removable handset/headset may be connected to either of two jacks on the left side of the Attendant Console.

Line Cord and Key Board: The line cord and the key board are connected to the two jacks located at the rear of the Attendant Console.

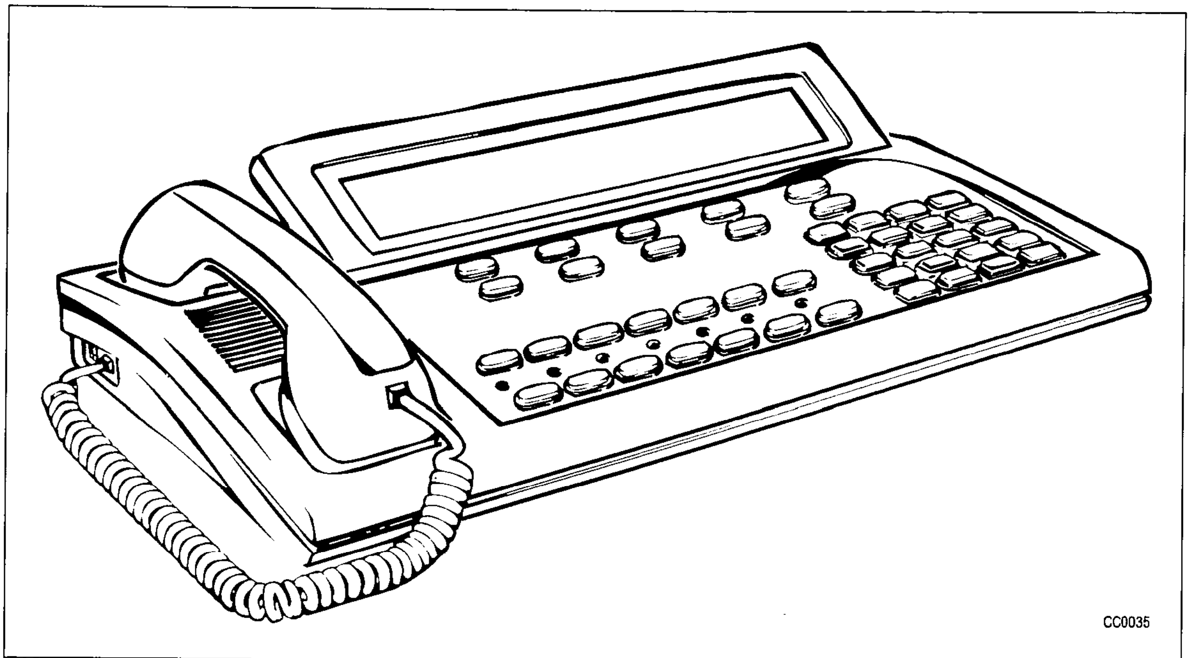


Figure 2-1 Attendant Console

Functional Description

2.3 The Attendant Console Keyboard layout (see Figure 2-2), provides standard Call Handling keys.

Attendant Function Keys (Fixed Keys)

The following fixed keys are available in Call Processing mode: FUNCTION, TRUNK GRP STATUS, CANCEL, RELEASE, PAGE, BLOCK, HOLD 1, HOLD 2, HOLD 3, HOLD 4, and ANSWER.

Display Associative Keys (Softkeys)

The 10 Display Associative Keys (F1 to F0) are located directly below the alphanumeric LCD. Each key is assigned 12 character positions on the third row (F1 to F5) and fourth row (F6 to F0) of the LCD. Softkey prompts indicate the function of these keys on the alphanumeric LCD. The function of these keys varies according to system status at a given time.

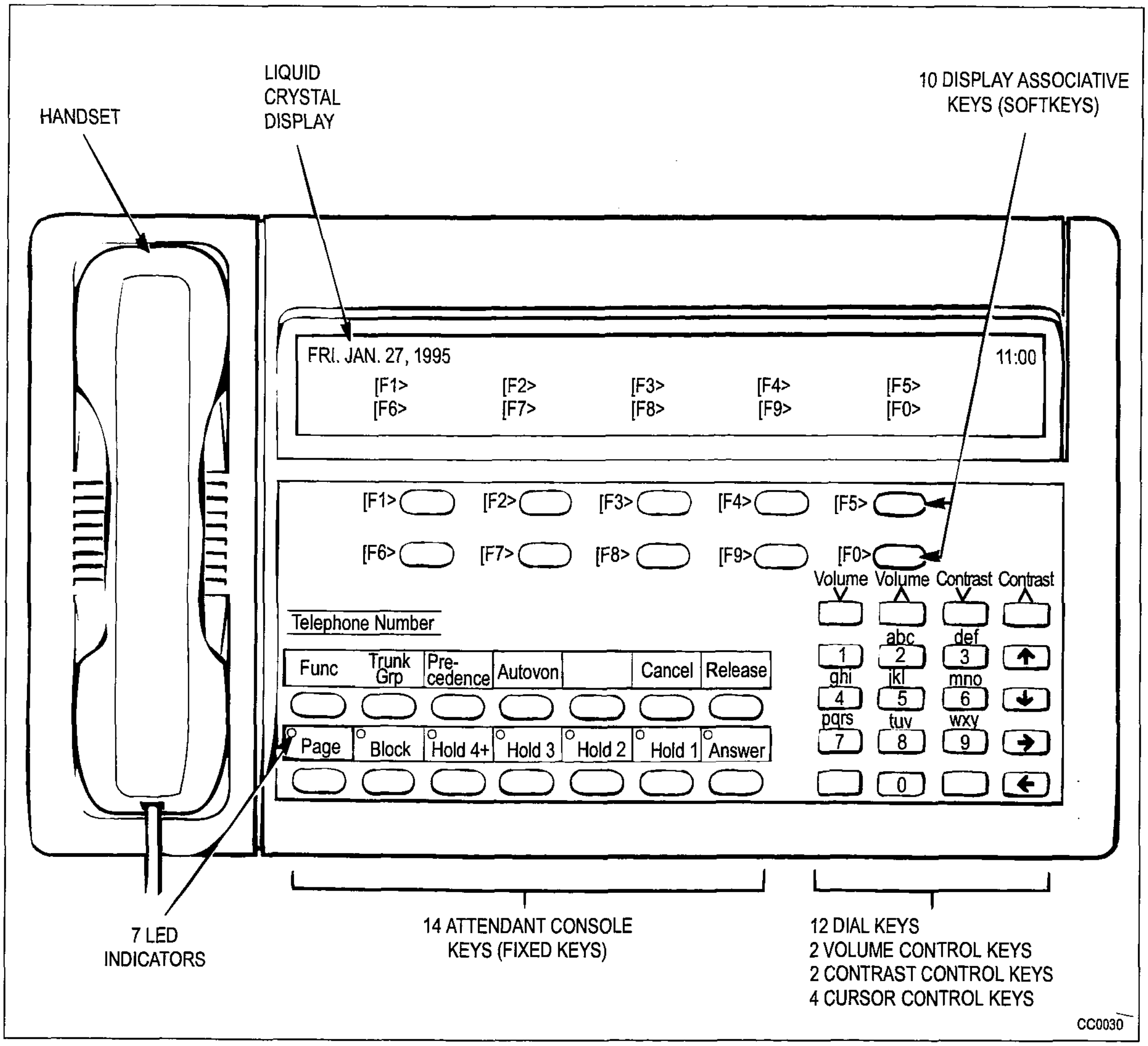


Figure 2-2 Attendant Console Keyboard Layout

Peripheral Devices

Installing a *SUPERCONSOLE 1000* Attendant Console

WARNING: **ANY CONNECTION OF THIS CONSOLE TO AN OFF PREMISE APPLICATION, AN OUT OF PLANT APPLICATION, OR TO ANY OTHER EXPOSED PLANT APPLICATION MAY RESULT IN A SAFETY HAZARD, AND/OR DEFECTIVE OPERATION, AND/OR EQUIPMENT DAMAGE.**

Note:

1. The *SUPERCONSOLE 1000* Attendant Console uses a two-wire connection to connect to one circuit on a digital line card (Tip to Tip and Ring to Ring.) The digital line card must be in a high-powered slot. Four consoles are allowed per digital line card.
2. The system maximum is 11 consoles.
3. The *SUPERCONSOLE 1000* Attendant Console can be used in the maintenance terminal port, by changing system default programming. Refer to the *Installation Information* Practice.
4. The password must be numeric only to use the console as a maintenance terminal.
5. The console operates in either English or French.
6. The console has a programmable printer port.

Chart 2-1 Install *SUPERCONSOLE 1000* Attendant Console

Step	Action	Comments
	Inventory, Unpack, Inspect	
1.	Check off received items against packing list and equipment list.	
2.	Unpack cartons.	Packaging is shown in Figure 2-3.
	Assemble Console	
3.	Connect the headset (or handset) to its cord.	
4.	Plug the cord into the jack at the side of the console.	The two jacks are connected in parallel; either can be used.
	Position Console	
5.	Put the console in its assigned position. Refer to the *Installation Information Practice*.	Maximum loop length is 1000 m (3300 ft).
6.	Plug one end of the console modular cord into the jack at the back of the console, and the other end into its assigned modular telephone jack.	The console jack is marked LINE PORT.
	Connect Console	
7.	Connect the modular telephone jack to the cross-connect field.	The green lead connects through the cross-connect field to a Tip circuit on the digital line card. The red lead connects through the cross-connect field to a Ring circuit on the digital line card.
8.	Connect the cross-connect field to a digital line port on a digital line card.	
	Connect the Printer (Optional)	
9.	Connect the printer to the RS-232 printer connector on the back of the console.	Pin Signal 1 frame ground 2 transmit data 3 receive data 4 ready to send 5 clear to send 6 data set ready 7 signal ground 8 carrier detect 20 data terminal ready Maximum 50 ft between the printer and the printer port on the *SUPERCONSOLE 1000* Attendant Console.
10.	Set the printer baud rate.	Maximum baud rate is 2400 baud.
	Cross Connect	
11.	At the distribution frame, cross connect the cable from the modular jack to the cable from the digital line port on the digital line card.	Refer to the *Installation Information* Practice, for cross-connect cabling information.

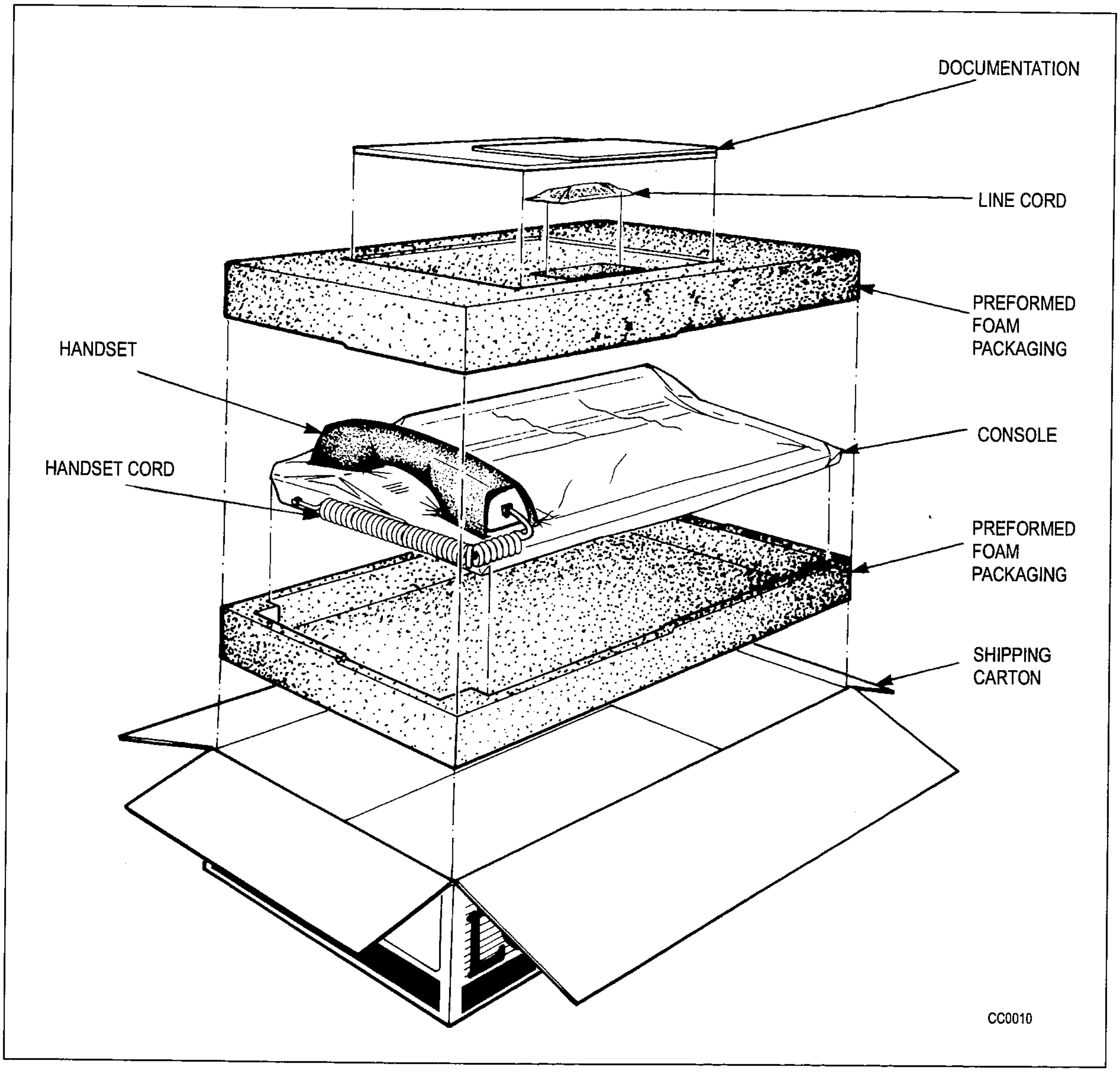


Figure 2-3 Attendant Console Packaging

RS-232C Connector

A printer can be attached to the *SUPERCONSOLE 1000* Attendant Console via its RS-232C port connector which allows direct connection to most printers. The RS-232C interface is programmable from 110 to 2400 baud, and acts like a DATASET 1101 cartridge, with the following exceptions:

- no autobaud
- device ID different from that of data cartridge
- no data loopback test.

The RS-232C connector (J3) is configured such that the console is the data communications equipment (DCE) to allow direct connection to most serial printers. Table 2-1 lists the RS-232C pins recognized by the console's printer port.

Table 2-1 RS-232C Port Pin Configuration - DCE			
Pin	**Designation**	**Status**	**Console Printer Port**
1	Frame Ground	- -	Connected
2	Tx Data	Input	Supported
3	Rx Data	Output	Supported
4	RTS (Request to Send)	Input	Supported
5	CTS (Clear to Send)	Output	Supported
6	DSR (Data Set Ready)	Output	*
7	Digital Ground	- -	Connected
8	DCD (Data Carrier Detect)	Output	*
20	DTR (Data Terminal Ready)	Input	Supported
Note: * DSR and DCD signals are wired together, driven from a single RS-232C output.			

The Attendant does not have an indication of printer status, and does not incorporate the following user interface features:

ATTN key DISC key READY LED DEVICE LED

SUPERSET 7000 Attendant Console

2.4 The *SUPERSET 7000* attendant console is designed to be an alternate answering point that operates on a PC. For example, the console can receive calls from the PABX when the primary attendant console(s) is encountering high traffic levels. The *SUPERSET 7000* application can maximize the PC display whenever a call is routed to it. At all other times, the PC can be used to perform its usual activities.

The *SUPERSET 7000* attendant console runs on an IBM®or compatible PC that uses Windows 3.1 or Windows95. It is equipped with a MITEL PC TALK TO® BX card, a *SUPERSET* 400 series handset, an extended keyboard, and a handset cradle. The PC *TALK TO* BX card uses a standard DNIC interface to communicate to the *SX-200* ML PABX.

CAUTION: The *SUPERSET 7000* attendant console is a secondary console only. It is NOT intended to be used as a Primary Console.

Features such as call handling, phone book utilities with user status, and trunk labeling, run on the *SUPERSET 7000* attendant console. The *SUPERSET 7000* attendant console performs call handling functions and is similar to the *SUPERCONSOLE 1000.* For information on the telephony features accessible through the console see the *Features Description* Practice.

Physical Description

The *SUPERSET 7000* attendant console consists of an IBM or compatible PC console, a video display terminal, a *TALK TO* card, a *SUPERSET* 400 series handset, an extended keyboard (with programmable hard keys, hard keys, softkeys, dial keys, and ASCII keys), and a handset cradle. The *TALK TO* card interfaces with the PABX via a single twisted pair connected to a digital line card circuit. Figure 2-4 shows the *SUPERSET 7000* attendant console.

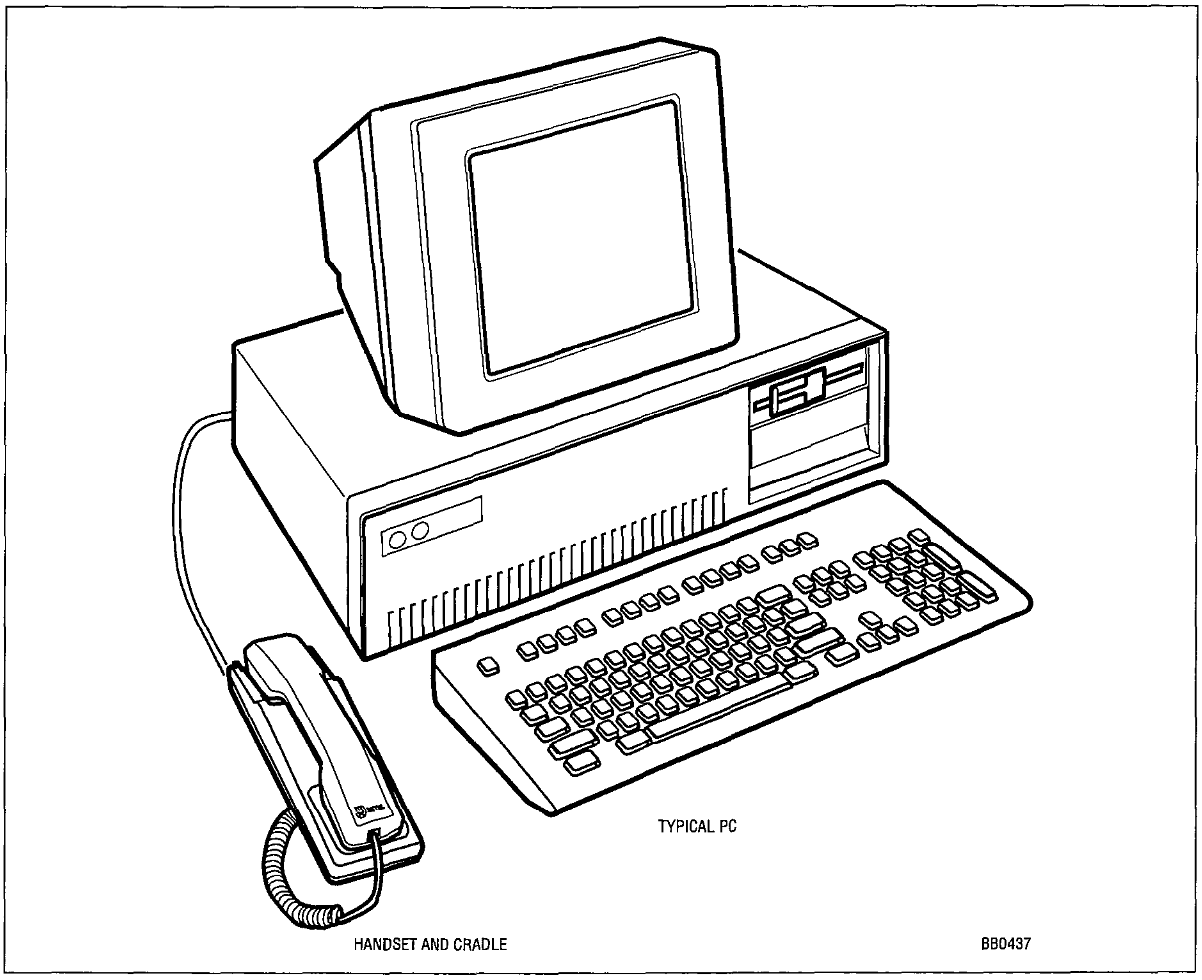


Figure 2-4 *SUPERSET 7000* Attendant Console

Keyboard - The *SUPERSET 7000* attendant console is equipped with an IBM AT, PC enhanced 101 key keyboard and provides the same functions as the *SUPERCONSOLE 1000.* It uses *SUPERCONSOLE 1000* key codes and protocol to interact with the PABX (see Figure 2-5).

PC *TALK TO* Card - The PC *TALK TO* BX card fits into one of the PC card expansion slots. The card has two modular phone jacks. The first jack is the MITEL standard telephone jack used for all DNIC ports or sets; the second jack connects to the handset (4 position, 4 contact jack).

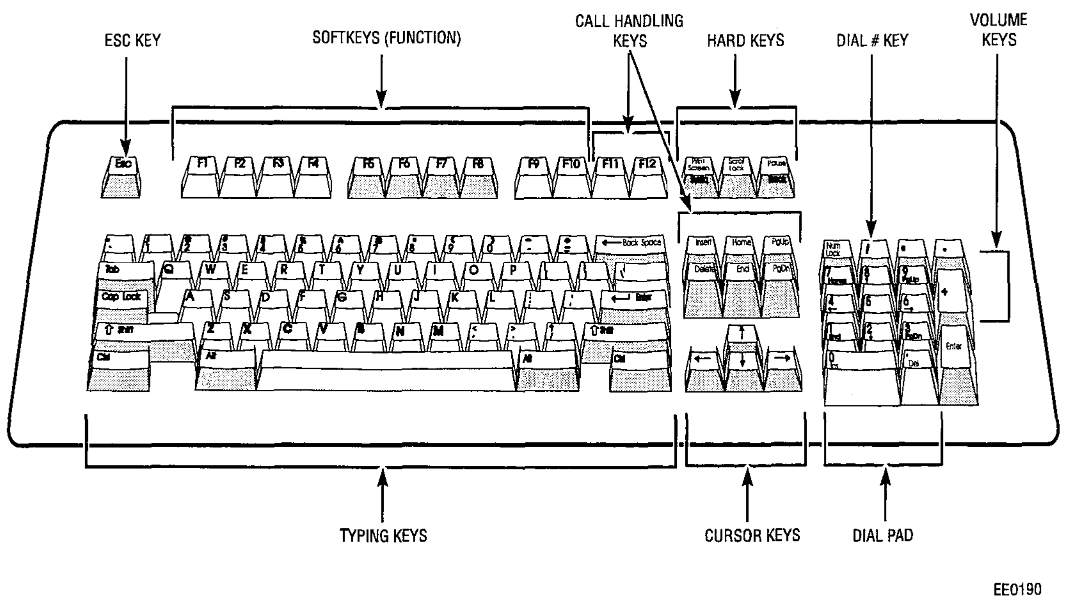


Figure 2-5 ***SUPERSET 7000*** **Keyboard Layout**

Functional Description

The console has 10 softkeys, three programmable firmkeys, eight attendant function keys, a dial pad, volume and arrow keys. and one hold key (see Figure 2-5 and Figure 2-6). It also has an integral handset.

The attendant function keys provide the following functions:

- FUNCTION: allows access to call handling and other applications that run on the *SX-200* ML PABX.
- CANCEL: cancels any call, for example, a misdialed call or a call directed to a busy number.
- RELEASE: releases a call.
- PAGE: accesses paging zones.
- CALL BLOCK: restricts extensions from calling one another (used in conjunction with Hotel/Motel applications).
- HOLD: (one key), the HOLD key is an overflow key, which provides access to calls on hold.
- ANSWER: answers incoming calls.

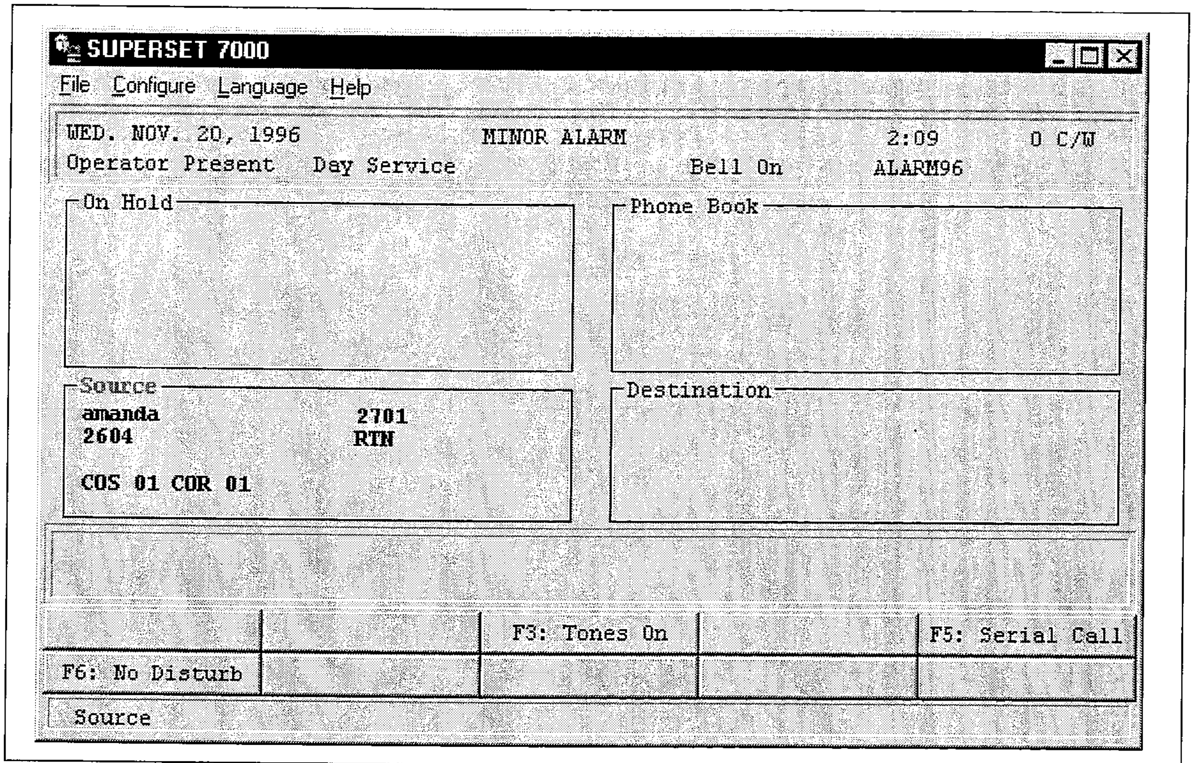


Figure 2-6 ***SUPERSET 7000*** **Display**

Functional Requirements

The *SUPERSET 7000* attendant console interfaces with the *SX-200* ML PABX via a digital line card, which provides telephone trunk and telephone extension information to it. For information on the digital line card interface, refer to the *Engineering Information* Practice.

Requirements for the console and the PC *TALK TO* BX card are:

PC Requirements

Acoustic noise (fan noise):	<45 dBSPL.
Mother board processor:	80486/66 MHz minimum
RAM:	Windows 3.1 - 8 Mb or Windows 95 - 16 Mb

Note: Windows 95 and a Pentium processor is recommended.

Display and Controller:	VGA 640 x 480 or better
Drives:	3.5"/1.44 MB floppy disk minimum
Hard Drive	Minimum 5 Mb free
Keyboard:	AT 101 fully functional enhanced
ISA slots:	one industry standard slot for the *TALK TO* BXcard

Sound Card	to configure ringer cadence and ringer volume (optional)

PC *TALK TO* Card Requirements

Power:	+5 Vdc + / -5% -5 Vdc + / -10% +12 Vdc + / -5% -12 Vdc + / -10%
Operating Temperature:	0 to +40°C
Storage Temperature:	-40 to +50°C
Relative Humidity:	5% to 95% non-condensing
Modular Jacks:	2

Operation

This part provides a brief operational outline. For additional information refer to the *SUPERSET 7000* attendant console user guide.

Power Up - Upon power up, the screen displays a message on line 3 of the PC which reads either, "Waiting for Communication" or "Waiting for Synchronization".

Program Termination - To terminate the *SUPERSET 7000* attendant console program, the attendant must press the 'Ctrl', 'Alt', 'Esc' key. When the software recognizes the terminating command, it will request confirmation before terminating the program.

Keyboard Input - If a sequence of keystrokes is interpreted as a command, a request is passed to the correct application resident on the *SUPERSET 7000* attendant console. If a keystroke sequence is interpreted as a telephony sequence, the information is passed to the console module and the video display indicates that the action being performed.

Handset Audio Interface - The audio interface comprises the handset (or headset). Handset levels are determined by the PABX.

Ringer Audio Interface - Ringer levels are set by the PC speaker circuit. With the optional sound card, ringer volume and pitch can be changed by accessing the sound card application.

Basic Telephone Operation - The *SUPERSET 7000* attendant console performs basic telephony functions such as dialing and generating supervisory signals. Supervisory tones, such as Dial Tone, Busy Tone, and Reorder Tone, are generated by the system and routed to the console through the voice channel.

Installing a *SUPERSET 7000* Console

These installation instructions are for installing a *TALK TO* BX card into the PC only. For information on installing the PC (not MITEL supplied), consult the PC manufacturer's installation guide.

CAUTION: **When setting the *SUPERSET 7000* in place, ensure that there is adequate airflow for cooling purposes. It should not be placed in any enclosure which may restrict this airflow.**

Do not restrict the airflow by placing any objects over the vents of the terminal.

Connect the console only to telephone lines equipped to interface it to the system.

Install the *TALK TO* card into the PC as described in Chart 2-2.

Chart 2-2 Install the *TALK TO* card into the PC

Step	Action	Comments
1.	Set the power switch on the PC unit to the OFF position.	
2.	Set all external power switches (printer, monitor, etc.) to the OFF position.	
3.	Unplug the PC unit power cord from the wall outlet.	
4.	Identify and remove all cables from the PC unit where the *TALK TO* BX card is to be installed.	
5.	Carefully remove the cover from the base of the unit.	Refer to the PC manufacturer's installation guide.
6.	Select a spare system expansion slot (furthest away from the hard disk drive) in which to install the *TALK TO* BX card (refer to Figure 2-7).	
7.	Remove the screw holding the expansion slot cover in place and remove the cover.	Retain the screw for installation of the card.
8.	Wearing an antistatic wrist strap, unpack and inspect the *TALK TO* BX card for damage.	If damaged or incorrect, repack in original package and return.
9.	Set the interrupt jumper setting on the *TALK TO* BX card (default is 15).	
10.	Set the address selection DIP switch (S1) setting on the *TALK TO* BX card (default base address is 0300).	

Page 1 of 2

Chart 2-2 Install the *TALK TO* card into the PC (continued)		
Step	**Action**	**Comments**
11.	Hold the *TALK TO* BX card by the top and insert into slot and press firmly to seat connector. Secure with screw from Step 7.	
12.	Replace cover of the PC unit.	
13.	Reconnect all cables removed in Step 4.	
14.	Plug the console modular cord into the LINE PORT jack located at the rear of *TALK TO* BX card, and into its assigned modular telephone jack.	
15.	Plug handset cord into the voice path jack at the rear of the console.	
16.	Plug the PC unit power cord into the wall outlet.	
17.	Set all external power switches (printer, monitor, etc.) to the ON position.	
18.	Set the power switch on the PC unit to the ON position.	On power-up, the *SUPERSET 7000* console performs an initialization procedure. When the application starts, the video display is enabled.
19.	Install the software and reconfigure the console.	Refer to the *SUPERSET 7000 Attendant Console User Guide* for instructions.
		Page 2 of 2

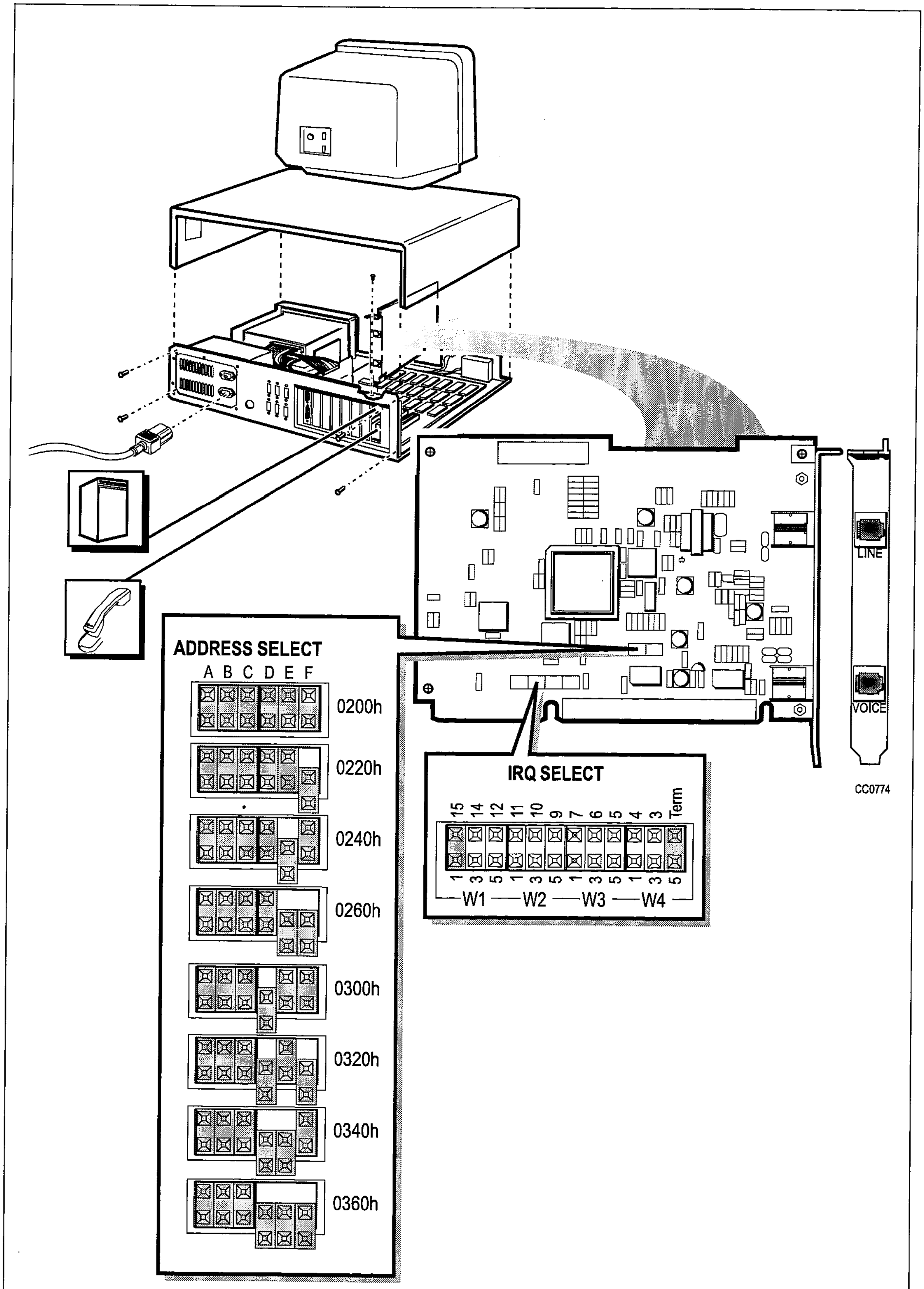


Figure 2-7 Typical *SUPERSET 7000* and *TALK TO* BX Card Installation

3 The *SUPERSET* 400 Series Telephone Sets

General

3.1 The *SUPERSET* 400 family of sets consists of four DNIC based digital sets designed for use on all MITEL DNIC based Digital Switches. The telephone sets are:

- SUPERSET 401+™
- SUPERSET 410™
- SUPERSET 420™
- SUPERSET 430™

The *SUPERSET 401+* set is a single line digital set.

The *SUPERSET 410, SUPERSET 420,* and *SUPERSET 430* sets are multi-line digital sets with LCD displays. They are equipped with a MILINK™ network connection that allows peripherals to be connected to the set on the desktop while only using one telephone port.

Line Status Display Symbols

3.2 The *SUPERSET 410, SUPERSET 420,* and *SUPERSET 430* telephones are equipped with an LCD Line Status Display. The status of each Line Appearance key is displayed using a triangular symbol. The meaning of each state is shown in Figure 3-1.

SYMBOL	DESCRIPTION	MEANING
	CLEAR	LINE IDLE
	DISPLAY SOLID TRIANGLE	THIS LINE IS BUSY AT THIS SET (OR AT ANOTHER SET)
	DISPLAY ALTERNATES BETWEEN SOLID AND CLEAR	INCOMING CALL
	DISPLAY ALTERNATES BETWEEN SOLID AND CLEAR	CALL ON HOLD AT THIS SET (OR AT ANOTHER SET)

EE0093

Figure 3-1 Line Status Display Symbols

Headset Compatability

3.3 Headset assemblies used with the *SUPERSET* 400 series telephone sets must be externally powered. Headsets from *SUPERSET 4* and *SUPERSET 4DN* telephone sets do not operate on *SUPERSET* 400 series telephone sets.

Connection to the *MILINK* Data Module

3.4 To accommodate data call operations, the *SUPERSET 410, SUPERSET 420,* and *SUPERSET 430* telephones are equipped with a *MILINK* network connection. This connection allows a *MILINK* device on the desktop, such as a *MILINK* Data Module, to be connected to the PABX. For *MILINK* Data Module installation instructions see the Dataset 1100 Series chapter in this Practice.

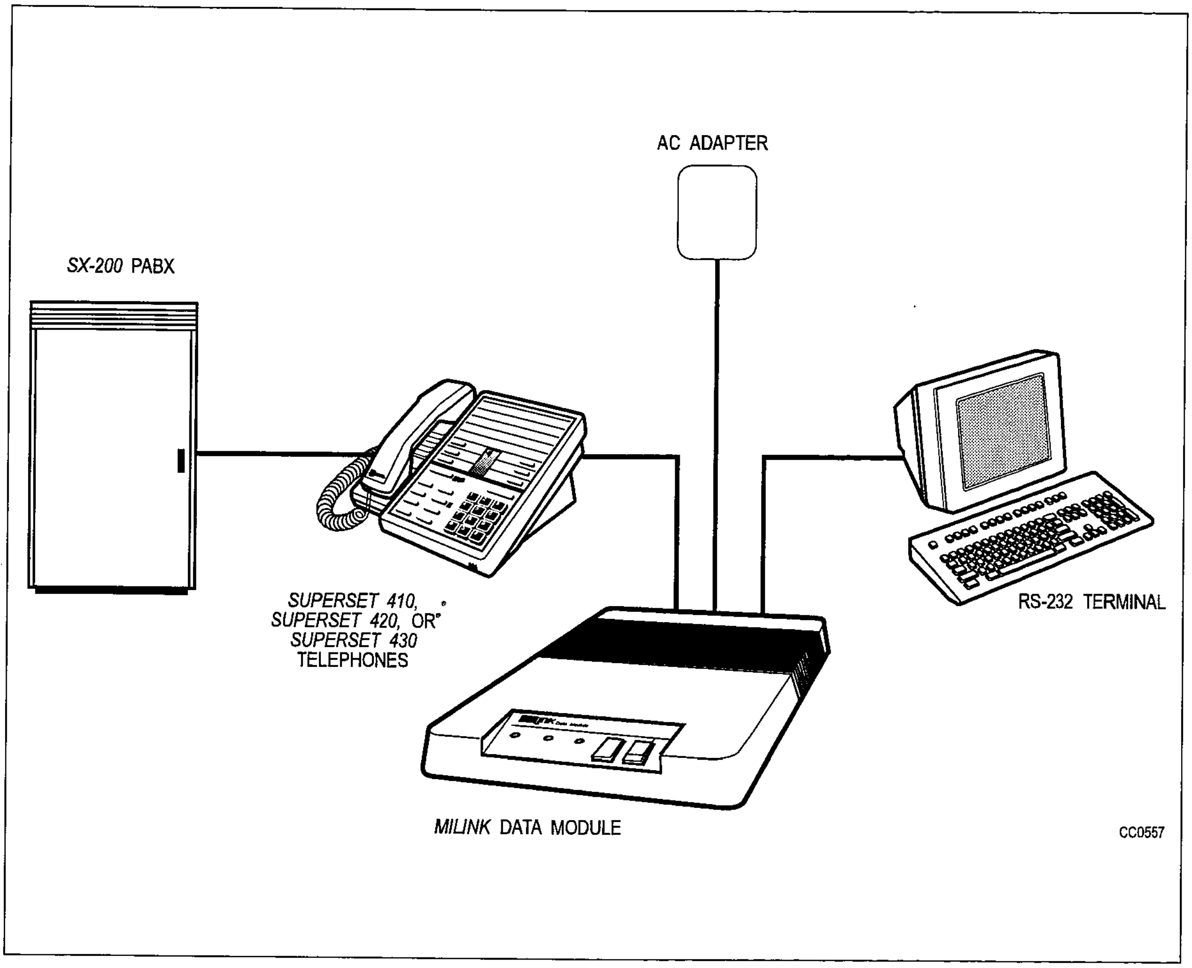


Figure 3-2 ***MILINK*** **Network Connections**

4 *SUPERSET 401+* Telephone

General Description

4.1 The *SUPERSET 401+* telephone is a single line DNIC-based digital telephone set that provides more features than an industry-standard telephone set. It also provides access to many of the sophisticated features available on the *SX-200* PABX. Figure 4-1 shows a *SUPERSET 401+* telephone and the layout of the faceplate. A description of available features is included in the *Features Description Practice.*

The *SUPERSET 401+* telephone has the following features:

- Feature hardkey (FLASH) to provide switch-hook flash
- Callback Messaging via the Message Waiting key
- Message Waiting Lamp
- Six programmable speed dial keys
- HOLD/RETRIEVE key
- Volume control for the handset receive level
- Adjustable ringer volume and tone
- Complete POTS (Plain Old Telephone Service) feature capability
- Built-in hearing aid compatibility in the handset.

The *SUPERSET 401+* telephone is designed for fast, simple installation; it connects to local area wiring by a modular jack and to the PABX by a single twisted pair. Power, signaling, and voice are carried over this pair; additional wiring is not required.

Physical Description

General

4.2 The top control panel of the *SUPERSET 401+* telephone is divided into three functional areas:

- handset and speaker
- fixed function keys and speed dial keys
- telephone keypad.

Dimensions

The set is 145 mm (5.7 inch) wide by 240 mm (9.45 inch) long. It is 86.5 mm (3.4 inches) high with the stand folded and 140 mm (5.5 inches) high with the stand extended.

Handset and Speaker

The handset mounts in a cradle on the left side of the telephone; the speaker is located beneath a grille between the cradle depressions.

Fixed Function Keys and Speed Dial Keys

The fixed function keys are arranged above the telephone keypad as shown in Figure 4-1. A Message Waiting Lamp is located to the right of the Message key. A Hold lamp is located next to the HOLD/RETRIEVE key. For CDE, the Speed dial keys are numbered 1 to 6 from bottom to top.

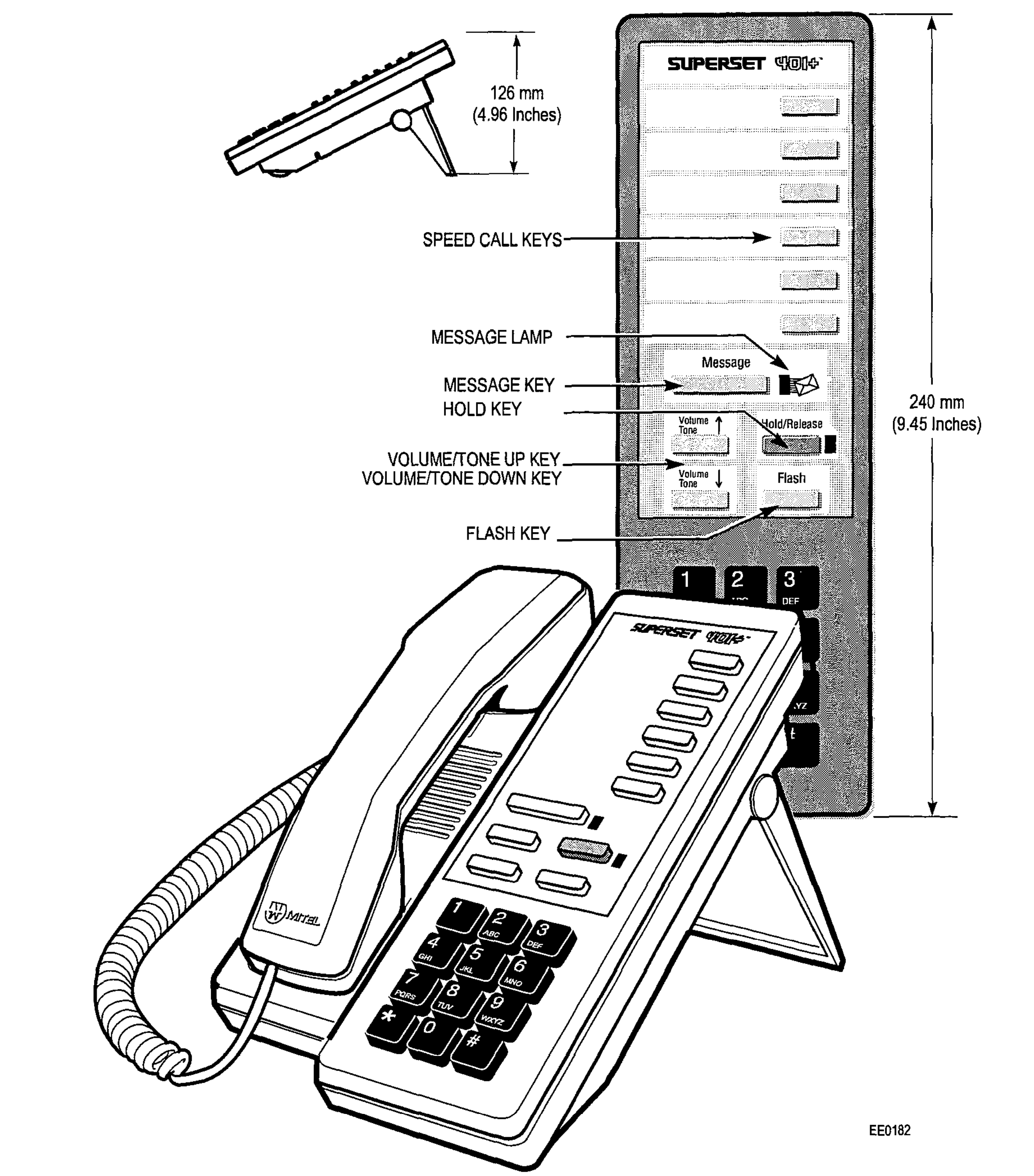


Figure 4-1 ***SUPERSET 401+*** **Telephone Set**

Telephone Keypad

The telephone keypad has 12 keys arranged in the standard pattern. The number keys have associated letters with a slight deviation from the standard keypad: the letter "q" appears on the 7 key and "z" appears on the 9 key. The following list illustrates the key number designation and the corresponding letter designations:

1	2 abc	3 def
4 ghi	5 jkl	6 mno
7 pqrs	8 tuv	9 wxyz
*	0	#

Functional Description

Feature Keys

4.3 There are six feature keys and two LEDs, each associated with a specific feature. They are:

MESSAGE Key: The Message key is used to send a callback message to another telephone set. It is also used to initiate a return call to a telephone set that has left a message.

MESSAGE Lamp: The red Message lamp flashes when another telephone set has left a callback message. The lamp also lights (solid red) when the *SUPERSET 401+* calls a busy or unanswered party capable of receiving a message. Pressing the message key sends a message to the called set. After the message has been sent, the lamp goes out and the call is terminated.

HOLD/RETRIEVE Key: The HOLD/RETRIEVE key is used to place a call on hold; pressing it a second time recalls the held call.

HOLD Lamp: The HOLD lamp flashes while a call is on hold.

VOLUME/TONE UP and VOLUME/TONE DOWN Keys: These two keys are used to adjust handset volume and ringing volume and tone. Each key depression changes the level up or down by one step.

The levels for ringing volume and pitch are stored in the PABX; the handset receive volume level can be adjusted up or down during a call; on every new call the default level is applied.

FLASH Key - This key emulates the switch-hook flash function and is used for Transfer, Conferencing, and PABX Feature Access.

PROGRAM Key: The PROGRAM key is located beneath the identification card, and is not usually visible. It is depressed during programming of a speed dial key.

SUPERSET 401+ Telephone Installation Instructions

4.4 Installation includes unpacking the set, assembling the set and connecting the set to a wall jack. The connections to the system via the cross-connect field are described in the *Installation Information Practice*.

WARNING: **Any connection of this set to an off premise application, an out of plant application, or to any other exposed plant application may result in a safety hazard, and/or defective operation, and/or equipment damage.**

CAUTION: **Do NOT connect *SUPERSET 401+* telephones in parallel, to standard lines, or as power fail transfer extensions. Do not use a hand test telephone (butt-in) to check a *SUPERSET 401+* telephone line (its DNI Line Card does not have a loop detector). The set's on/off hook status is signaled by data transmission.**

Before the set can be used, the system must be programmed and equipped with a DNI Line Card to interface with a *SUPERSET 401+* telephone. Refer to the *Customer Data Entry Practice.*

Chart 4-1 *SUPERSET 401+* Telephone Installation Instructions

Step	Action
1.	Connect the handset cord to the handset and the *SUPERSET 401+* telephone.
2.	Identify the set's number on the identification card.
3.	Install the identification card and protective cover onto the main assembly.
4.	Connect the line cord to the telephone jack and the *SUPERSET 401+* telephone.
5.	The Message Lamp blinks rapidly for about 5 seconds and then goes out. When the lamp goes out, it indicates that communication to the PABX has been established.

Table 4-1 *SUPERSET 401+* Telephone Environmental Specifications

Operating Environment:	
Temperature:	0° to 50°C (32° to 122°F).
Humidity:	0% to 90% RH, non-condensing.
Storage/Shipping Environment:	
Temperature:	-25° to 70°C (-13° to 158°F).
Humidity:	0% to 90% RH, non-condensing.

Wall-Mounting the *SUPERSET 401+* Telephone

4.5 To wall-mount the *SUPERSET 401+* telephone, remove the detachable stand (tilt to 45 degree angle). Plug the short line cord into the set. Plug this cord into its modular jack, and place the set onto the protruding pins of the wall mounted plate. Extend the clip that holds the handset in the vertical position. Refer to Figure 4-2.

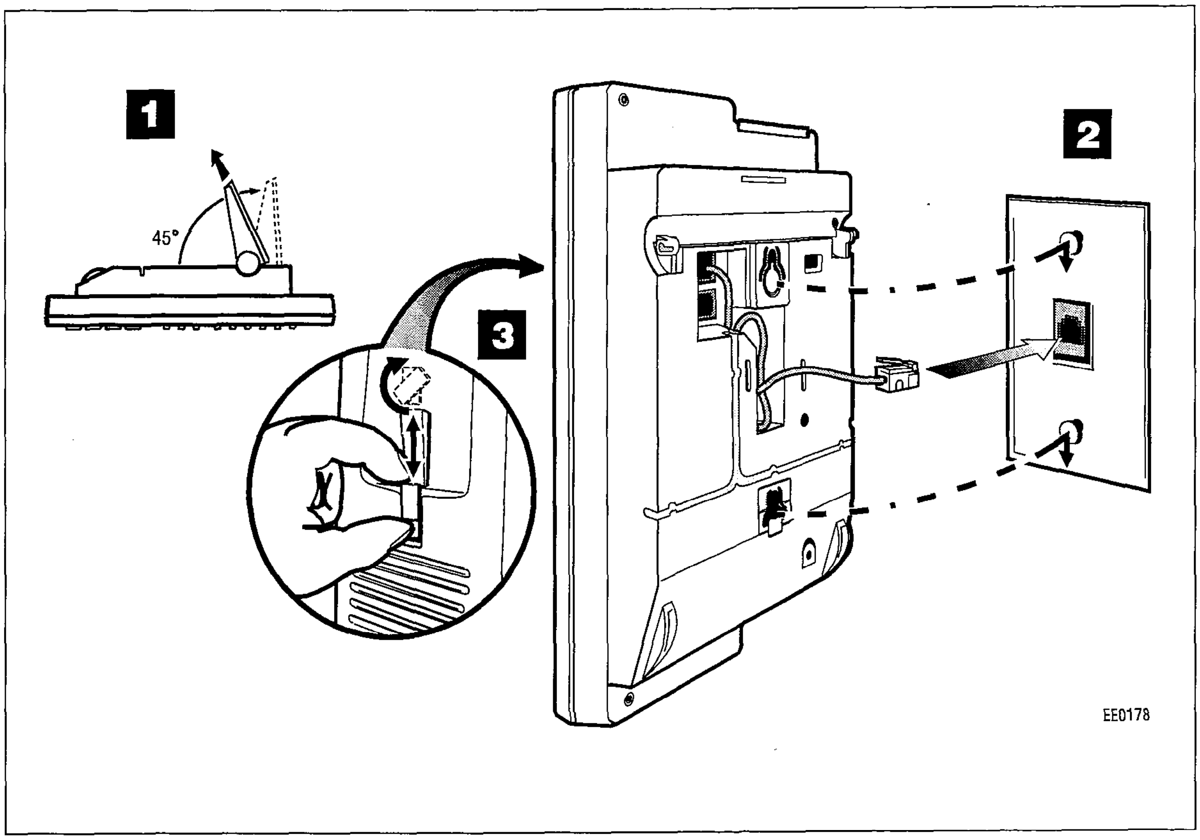


Figure 4-2 Wall Mounting a *SUPERSET 401+* Telephone Set

Visual Indication Of System Communications Problems

4.6 If there is either a synchronization or communication error, the Message Lamp and Hold Indicator flash at double the "Message Waiting" rate.

If the Message and Hold lamps are flashing alternately, the set is unable to detect stable synchronization.

If the Message and Hold lamps are flashing together, the set is unable to establish and/or maintain communications.

5 *SUPERSET 410* Telephone

General Description

5.1 The *SUPERSET 410* telephone is a DNIC-based digital telephone that provides many features to enhance communications in a modern business environment. Figure 5-1 shows a *SUPERSET 410* telephone set while Figure 5-2 shows the layout of the faceplate. A description of the available system features is included in the *Features Description Practice.*

The *SUPERSET 410* telephone has the following features:

- six Line Select/Speed Call/Feature keys with individual line status indicators
- 10 fixed function keys, two with associated LEDs
- automatic selection of prime line
- key selection of non-prime line
- automatic ringing line selection
- single key feature activation
- handsfree dialing
- Handsfree Answer Back
- programmable Call Forwarding and Speed Call keys
- speaker, handset and ringer volume controls
- ringer tone control
- Hold key
- digital interface with the system via the DNI Line Card
- single key Call Transfer and Conferencing activation
- Message Waiting Lamp LED
- optional data access via the *MILINK* Data Module, for integrated voice and data communications over a single twisted pair
- built-in hearing aid compatibility in the handset.

The *SUPERSET 410* telephone is designed for fast, simple installation. It connects to local area wiring by a modular jack and to the PABX by a single twisted pair. Power, signaling, voice, and data are carried over this pair; additional wiring is not required.

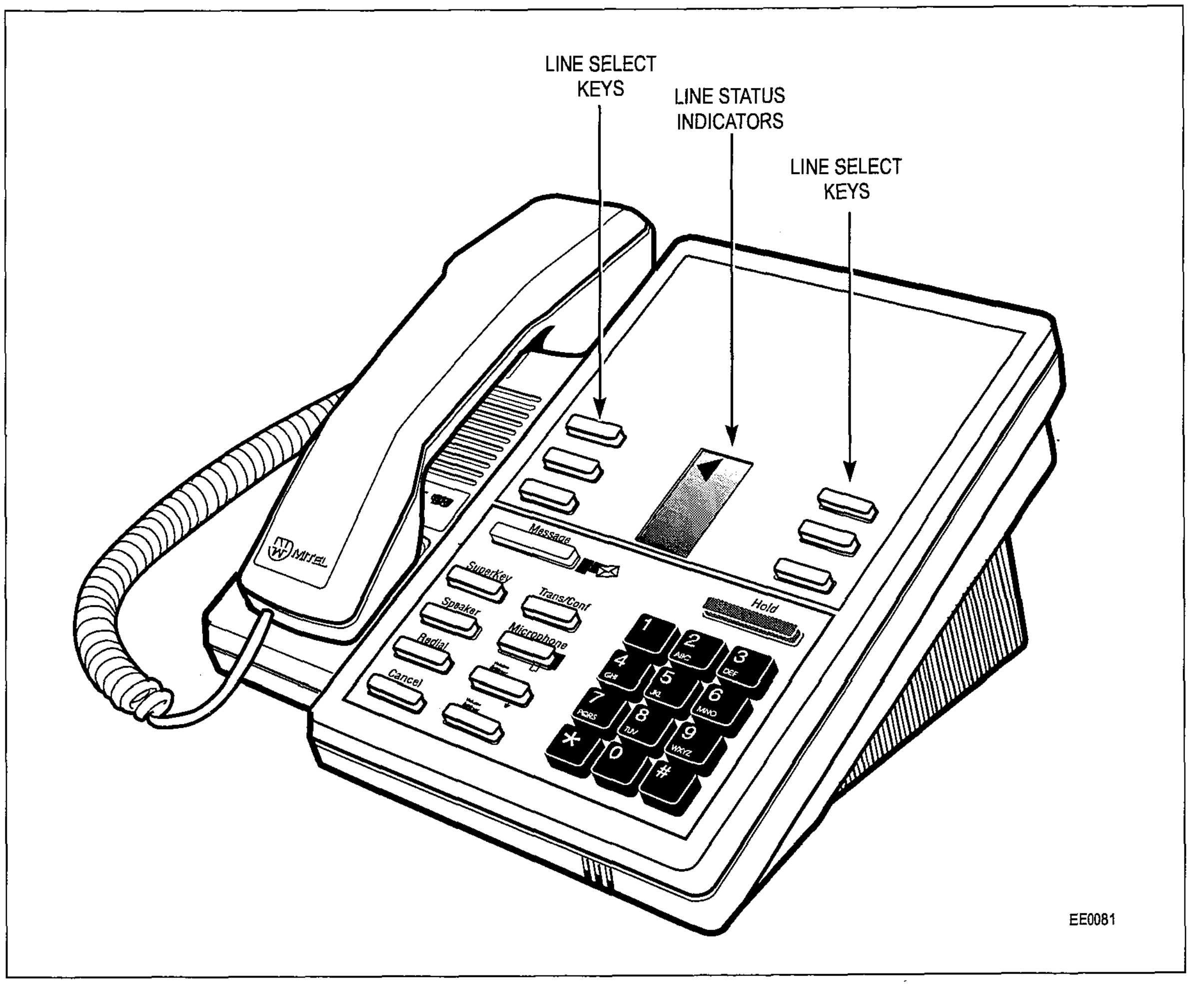


Figure 5-1 ***SUPERSET 410*** **Telephone**

Physical Description

General

5.2 The dimensions of the *SUPERSET 410* telephone are shown in Figure 5-2. The faceplate of the telephone is divided into four functional areas:

- handset, speaker, and microphone
- line appearance keys with status indicators
- telephone keypad
- fixed function keys

Handset, Speaker and Microphone

The handset mounts in a cradle on the left side of the telephone; the speaker is located beneath a grille between the cradle depressions. The microphone is located in the bottom right edge of the telephone, below the " 0 " on the keypad.

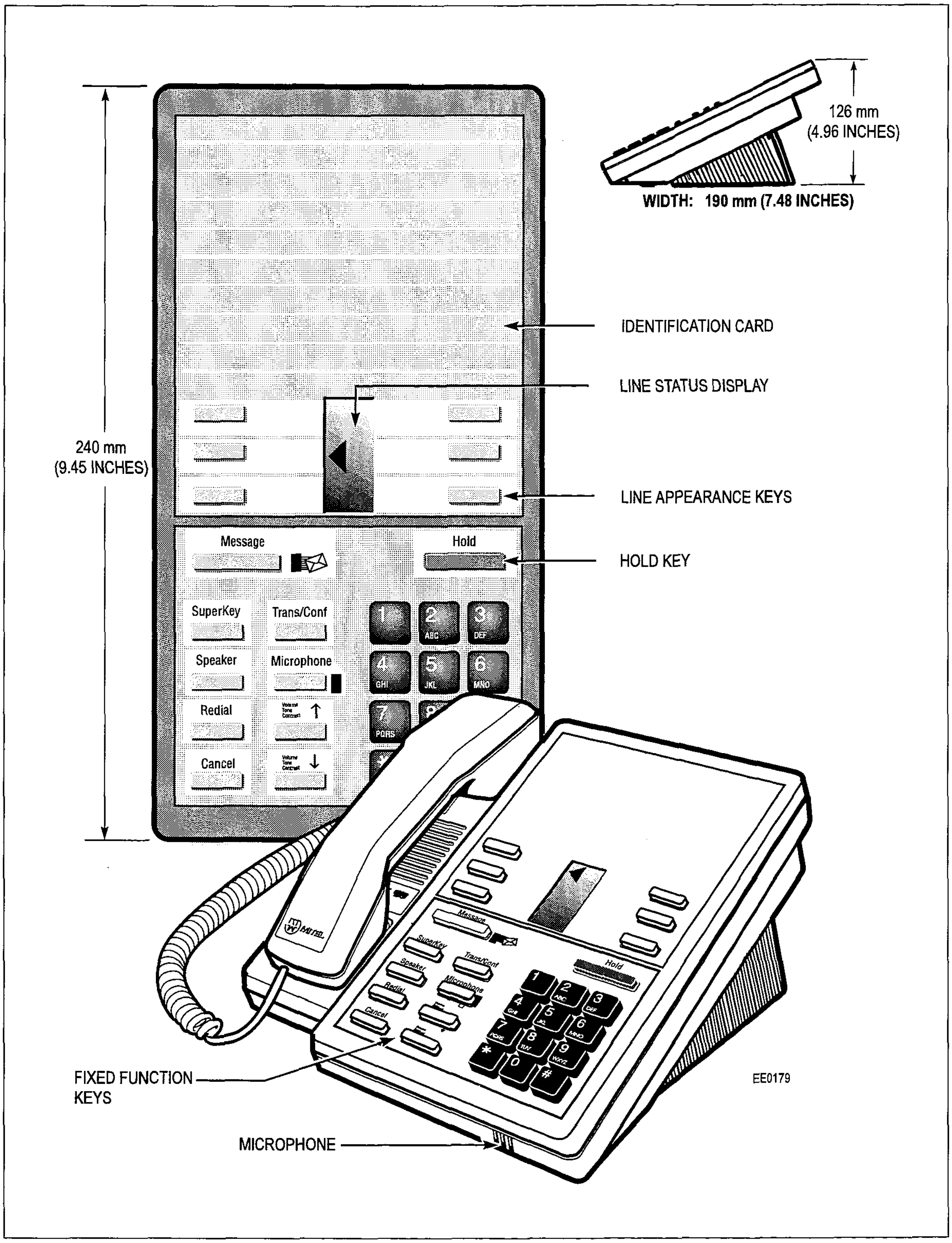


Figure 5-2 *SUPERSET 410* Telephone Dimensions

Line Appearance Keys

There are six Line Appearance keys arranged in two columns in the center of the set. They are located in the top section of the telephone's faceplate. These keys may be programmed to access specific lines, specific features, or to activate Speed Call numbers. Between the two columns of Line Appearance keys is a liquid crystal display (LCD) indicator that provides the status of each line appearance.

Telephone Keypad

The telephone keypad has 12 keys arranged in the standard pattern. It is in the bottom right hand section of the faceplate. The number keys have associated letters with a slight deviation from the standard: the letter "q" appears on the 7 key; "z" appears on the 9 key. The following list illustrates the key number designation and the corresponding letter designations:

1	2 abc	3 def
4 ghi	5 jkl	6 mno
7 pqrs	8 tuv	9 wxyz
*	0	#

Fixed Function Keys

Nine of the 10 fixed function keys are arranged to the left of the telephone keypad. The red Hold key is located above the keypad. The Message Waiting Lamp and the Microphone Lamp have LED indicators.

Functional Description

5.3 Ten fixed function keys surround the keypad. Two of these keys have LED visual indicators. The fixed function keys are:

Fixed Function Keys

HOLD Key: Pressing the red Hold key places the current call on hard hold. The call may be retrieved by pressing the line select key of the call on hold (next to the flashing line select status indicator).

MESSAGE Key: The Message key is used to send a callback message to another telephone set capable of receiving the message when that set is busy, unanswered, or in Do Not Disturb mode. It is also used to initiate a return call to a telephone set that has left a message.

MESSAGE Lamp: The red Message lamp flashes when another telephone set has left a callback message. The lamp lights (solid red) when the *SUPERSET 410* calls a busy or unanswered party capable of receiving a message. Pressing the message key sends a message to the called set. After the message is sent,the lamp goes out and the call is terminated.

SUPERKEY Key: The Superkey is used to initiate programming the Line Appearance keys as Feature keys or Speed Call keys.

SPEAKER Key: This is an On/Off toggle key. With the key in the On state during handsfree dialing, the caller can monitor call progress through the speaker. When the caller hears the called party answer, the caller must lift the handset to respond (handsfree conversation is not available on the *SUPERSET 410*).

REDIAL Key: The last manually dialed number is dialed when the redial key is pressed.

CANCEL Key: This key is used to cancel a dialing sequence, end a call, terminate without saving a programming function, or return to a party on soft hold.

VOLUME/TONE UP and VOLUME/TONE DOWN Keys: These two keys are used to adjust handset volume and ringing volume and tone. Each key depression changes the level up or down by one step.

The levels for ringing volume and pitch is stored in the PABX; changing one function setting does not affect the others. On system or set resets, the levels are restored. Default levels are applied when the set is first programmed. The handset receive volume level can be adjusted up or down during a call. However, the level is not reported to the PABX; therefore, on every new call the default level is applied.

MICROPHONE Key and MICROPHONE Lamp: Handsfree conversation is not supported on the *SUPERSET 410* telephone set. On normal calls, pressing the microphone key will have no effect.

When the *SUPERSET 410* is programmed as a Key System set, the Handsfree Answer Back (HFAB) option is available. The Microphone is then activated by the system software when an internal page is made to the set, if the Microphone key is ON and the HFAB COS Option is enabled. The user is allowed to answer a page without having to pick up the handset. For the duration of the Handsfree Answer Back call the Microphone Lamp is lit to indicate the ON setting of the Microphone.

TRANS/CONF Key - This key is used for Call Transferring and Conferencing. Pressing the key allows the user to put a call on hold, dial a new number and conference the call, or transfer the held call (with privacy) to a new number.

Line Appearance Keys

There are 6 Line Appearance Keys arranged in two columns. For CDE purposes these keys are numbered from bottom to top, right to left, from1 to 6 (as shown below). The first Line Appearance key is always programmed as the Prime Line. The liquid crystal display (LCD) indicator provides the status of each line appearance.

6	5
4	3
2	1

These keys may be programmed to access specific lines and specific PABX features. These keys can be designated during CDE or they can be programmed from the set; refer to the *Features Description Practice.*

Speed Call Keys

Any unused Line Appearance key may also be programmed as a personal Speed Call key, by the user from the *SUPERSET 410.*

SUPERSET 410 Telephone Installation Instructions

5.4 Installation includes unpacking the set, assembling the set, and connecting the set to a wall jack. The connections to the system via the cross connect field are described in the *Installation Information* Practice.

WARNING: **ANY CONNECTION OF THIS SET TO AN OFF PREMISE APPLICATION, AN OUT OF PLANT APPLICATION, OR TO ANY OTHER EXPOSED PLANT APPLICATION MAY RESULT IN A SAFETY HAZARD, AND/OR DEFECTIVE OPERATION, AND/OR EQUIPMENT DAMAGE.**

CAUTION: **Do NOT connect *SUPERSET 410* telephones in parallel, to standard lines, or as power fail transfer extensions. Do not use a hand test telephone (butt-in) to check a *SUPERSET 410* telephone line (it's DNI Line Card does not have a loop detector). The set's on/off hook status is signaled by data transmission.**

Before the set can be used, the system must be programmed and equipped with DNI Line Cards to interface with a *SUPERSET 410* telephone. Refer to the *Customer Data Entry* Practice.

Chart 5-1 *SUPERSET 410* Telephone Installation Instructions

Step	Action
1.	Connect the handset cord to the handset and main assembly (See Figure 5-3).
2.	Write the set's telephone number on the Brand/Extension Card.
3.	Identify the set's extension number, other line appearances and features on the Designation Card.
4.	Install the Brand/extension Card, the Designation Card and the Feature Designation Card and their protective lenses onto the main assembly.
5.	Connect the line cord to the telephone and the telephone jack.
6.	Insert the line cord and the handset cord into the cord channels.
7.	The Message Lamp and the Microphone Lamp go solid red and the triangular line appearance indicators turn on for about 5 seconds. When they go out, it indicates that communication to the PABX has been established.

Table 5-1 *SUPERSET 410* Telephone Environmental Specifications

Operating Environment:	
Temperature:	0° to 35° C (32° to 95° F).
Humidity:	0% to 90% RH, non-condensing.
Storage/Shipping Environment:	
Temperature:	-25° to 70° C (-13° to 158° F).
Humidity:	0% to 90% RH, non-condensing.

Wall-Mounting the *SUPERSET 410* Telephone

5.5 To wall-mount the *SUPERSET 410* telephone reverse the detachable base and plug the line cord into a modular jack. The "cord winder" in the base stores extra cord (See Figure 5-3). The complete wall-mounting instructions can be ordered through normal distribution channels.

Visual Indication of System Communications Problems

5.6 If there is either a synchronization or communication error, the Message Lamp and the Microphone Lamp flashes at double the "Message Waiting" rate.

Connection to the *MILINK* Data Module

5.7 To accommodate data call operations the *SUPERSET 410* telephone is equipped with a *MILINK* network connection. This allows the set to be connected to a *MILINK* Data Module on the desktop and share its telephone port. For *MILINK* Data Module installation instructions, see the Dataset 1100 Series Description section in this Practice.

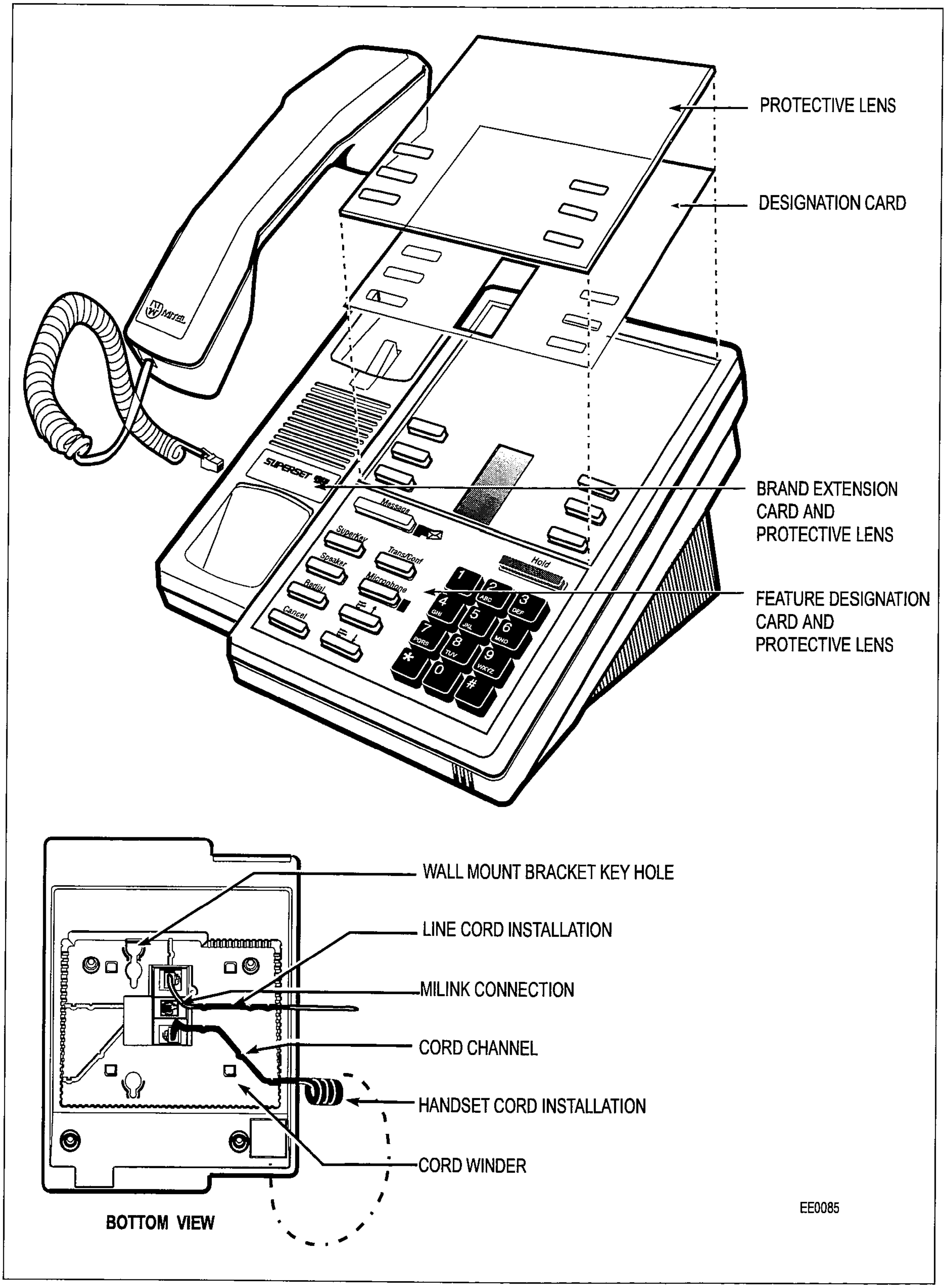


Figure 5-3 ***SUPERSET 410*** **Installation Instructions**

6 *SUPERSET 420* Telephone

General Description

6.1 The *SUPERSET 420* telephone is a DNIC-based digital telephone that provides many features to enhance communications in a modern business environment. It is a highly functional display set that includes an integral alphanumeric LCD display with controllable contrast. Figure 6-1 shows a *SUPERSET 420* telephone set while Figure 6-2 shows the layout of the faceplate. A description of the available features is included in the *Features Description Practice.*

The *SUPERSET 420* telephone has the following features:

- 12 Line Select/Speed Call/Feature keys with individual line status indicators
- 10 fixed function keys, two with associated LEDs
- softkey operation of most features (three softkeys; softkey prompts display the actions which are valid at any given time)
- 2 x 16 alphanumeric Liquid Crystal Display with controllable contrast
- automatic selection of prime line
- key selection of non-prime line
- handsfree operation
- single key feature activation
- automatic ringing line selection
- programmable Call Forwarding and Speed Call keys
- speaker, handset, and ringer volume controls
- ringer tone control
- Hold key
- digital interface with the system via the DNI Line Card
- single key Call Transfer and Conferencing activation
- Message Waiting Lamp LED
- optional data access via the *MILINK* Data Module, for integrated voice and data communications over a single twisted pair
- built-in hearing aid compatibility in the handset.

The *SUPERSET 420* telephone is designed for fast, simple installation. It connects to local area wiring by a modular jack, and to the PABX by a single twisted pair. Power, signaling, voice, and data are carried over this pair; additional wiring is not required.

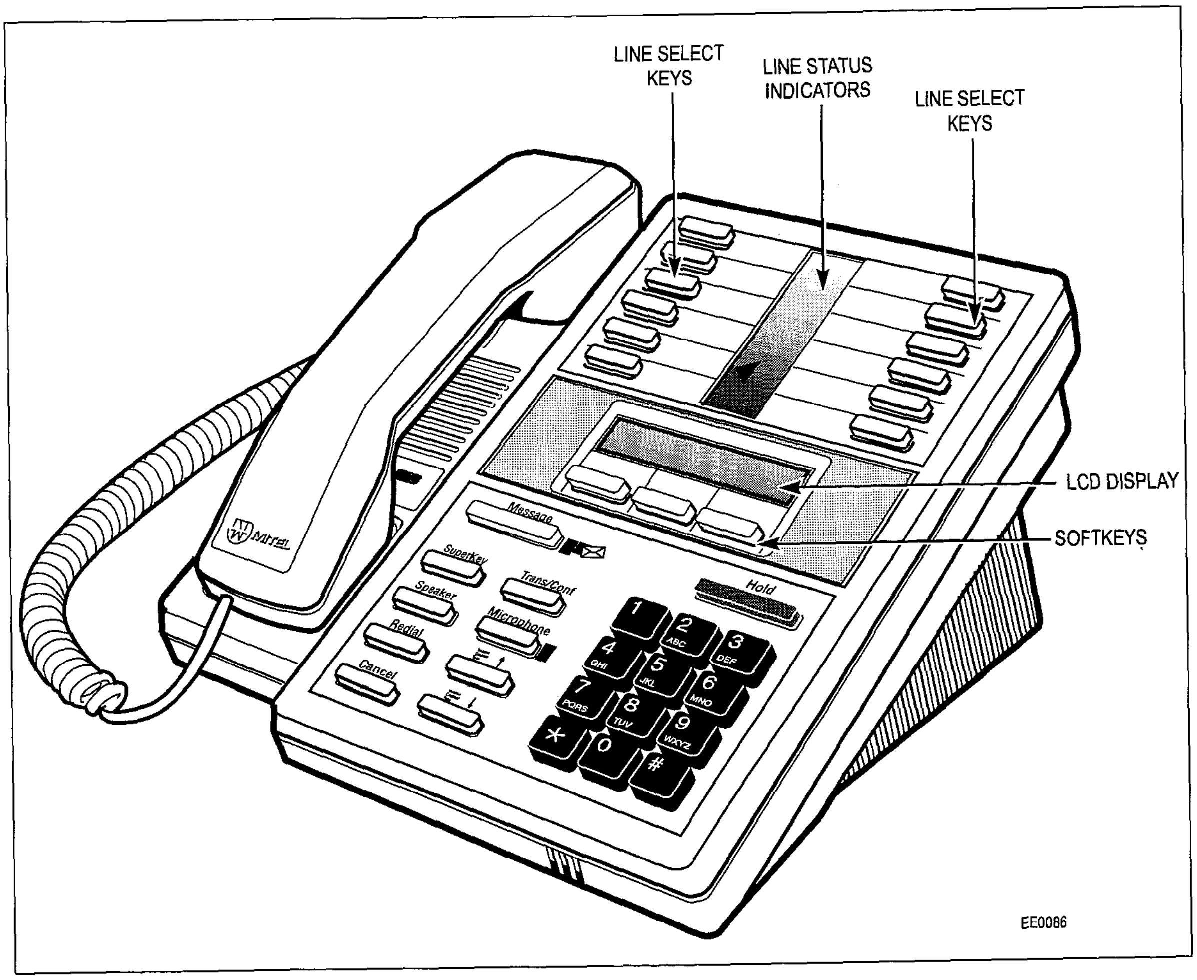


Figure 6-1 ***SUPERSET 420*** **Telephone**

Physical Description

General

6.2 The dimensions of the *SUPERSET 420* telephone are shown in Figure 6-2. The faceplate of the telephone is divided into five functional areas:

- handset, speaker, and microphone
- line appearance keys with status indicators
- softkeys and LCD display
- telephone keypad
- fixed function keys.

Handset, Speaker and Microphone

The handset mounts in a cradle on the left side of the telephone; the speaker is located beneath a grille between the cradle depressions. The microphone is located in the bottom right edge of the telephone, below the " 0 " on the keypad.

Line Appearance Keys

There are 12 Line Appearance keys arranged in two columns in the top section of the telephone's faceplate. These keys may be programmed to access specific lines, specific features, or to activate Speed Call numbers. Between the two columns of Line Appearance keys is a liquid crystal display (LCD) indicator that provides the status of each line appearance.

Softkeys and LCD Display

In the center of the faceplate is a 2 x 16 Liquid Crystal Display (LCD). The top line contains system messages; the bottom line contains the current softkey definitions. The Volume Up and Volume Down fixed function keys at the bottom of the set provide contrast control for varying the intensity of the display. The three blue softkeys are located directly below the LCD display.

Telephone Keypad

The telephone keypad has 12 keys arranged in the standard pattern, located in the bottom right-hand section of the faceplate. The number keys have associated letters with a slight deviation from the standard: the letter "q" appears on the 7 key; "z" appears on the 9 key. The following list illustrates the key number designation and the corresponding letter designations:

1	2 abc	3 def
4 ghi	5 jkl	6 mno
7 pqrs	8 tuv	9 wxyz
*	0	#

Fixed Function Keys

Nine of the 10 fixed function keys are arranged to the left of the telephone keypad. The red Hold key is located above the keypad. The Message Waiting Lamp and the Microphone Lamp have LED indicators.

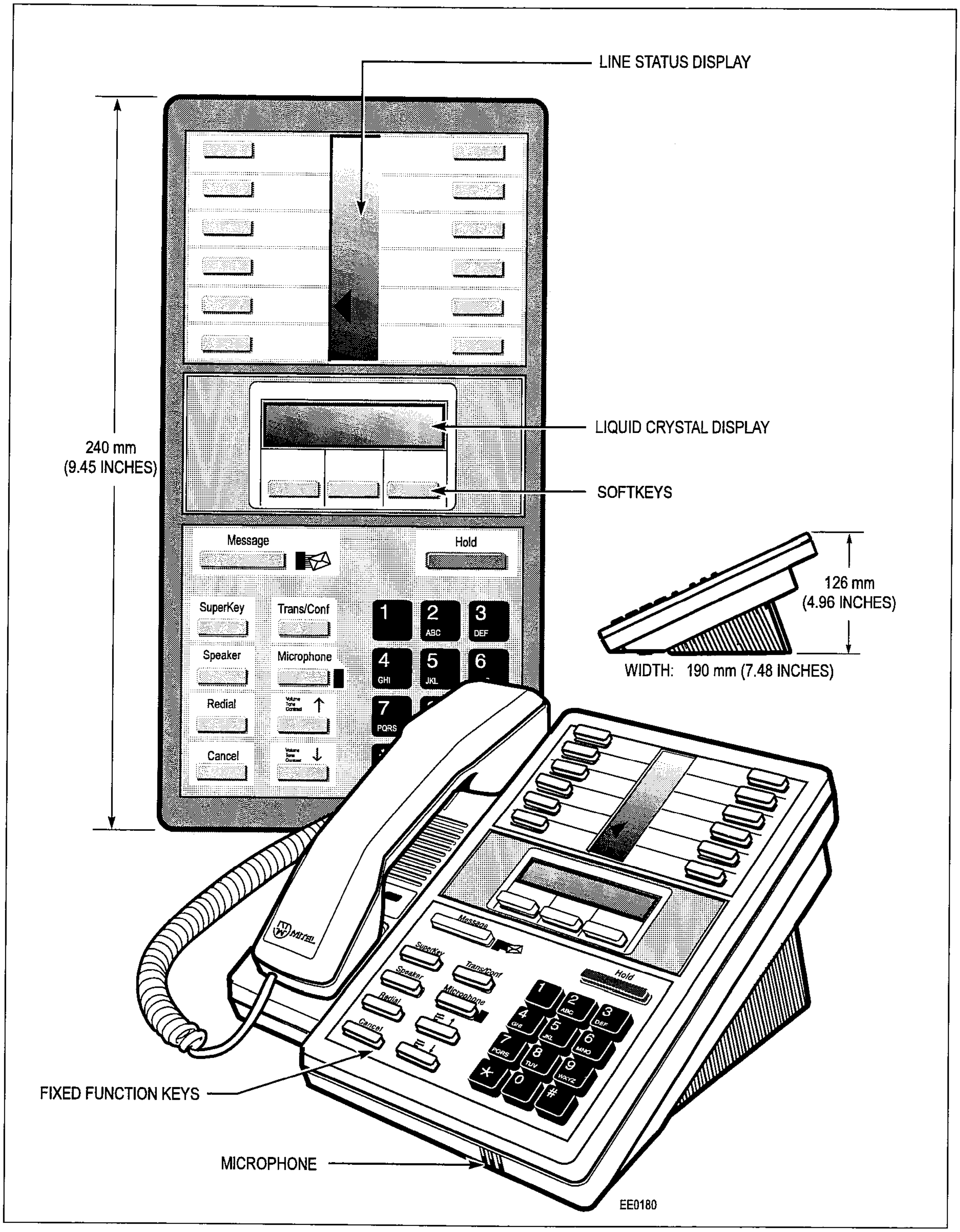


Figure 6-2 ***SUPERSET 420*** **Telephone Dimensions**

Functional Description

Fixed Function Keys

6.3 Ten fixed function keys surround the keypad. Two of these keys have LED visual indicators. The fixed function keys are:

HOLD Key: Pressing the red Hold key places the current call on hard hold. The call may be retrieved by pressing the line select key of the call on hold (next to the flashing line select status indicator).

MESSAGE Key: The Message key is used to send a callback message to another telephone set capable of receiving the message when that set is busy, unanswered, or in Do Not Disturb mode. It is also used to initiate a return call to a telephone set that has left a message.

MESSAGE Lamp: The red Message lamp flashes when another telephone set has left a callback message. The lamp lights (solid red) when the *SUPERSET 420* calls a busy or unanswered party capable of receiving a message. Pressing the message key sends a message to the called set. After the message is sent, the lamp goes out and the call is terminated.

SUPERKEY Key: The SUPERKEY and the softkeys are used to program the Line Appearance keys as Feature keys or Speed Call keys.

SPEAKER Key: The SPEAKER key controls handsfree operation of the set. Pressing this key once switches the speaker ON and selects the prime line. Calls can be originated and/or received handsfree. Successive operation of this key toggles the set between ON/OFF hook states.

REDIAL Key: The last manually dialed number is dialed when the redial key is pressed.

CANCEL Key: This key is used to cancel a dialing sequence, end a call, terminate without saving a programming function, or return to a party on soft hold.

VOLUME/TONE UP and VOLUME/TONE DOWN Keys: These two keys are used to adjust handset volume and ringing volume and tone. Each key depression changes the level up or down by one step.

The levels for ringing volume and pitch are stored in the PABX; changing one function setting does not affect the others. On system or set resets, the levels are restored. Default levels are applied when the set is first programmed. The handset receive level is not reported to the PABX; therefore, the default level is applied on every new call.

MICROPHONE Key and MICROPHONE Lamp: The microphone is turned on by the system software when a handsfree call is activated. The Microphone key is used to switch the microphone off during handsfree mode so that the conversation near the set is not overheard by the caller. Successive operation of the Microphone key toggles the microphone between ON and OFF.

The Microphone lamp is lit (solid red) when the microphone is ON.

TRANS/CONF Key - This key is used for Call Transferring and Conferencing. Pressing the key allows the user to put a call on hold, dial a new number and conference the call, or transfer the held call (with privacy) to a new number.

LCD Display

The LCD Display gives you the following information:

- time and date when the set is idle
- softkey labels during programming and feature access
- call status during telephone calls
- message information
- telephone system error messages.

Softkeys and SUPERKEY

Immediately below the LCD are three unlabeled blue softkeys. Their functions are defined by corresponding prompts displayed on the LCD. The softkeys are used during telephone operations to perform functions or to access system features. The keys' functions vary with the activities of the set. The SUPERKEY is a fixed function key, which is located at the bottom left side of the set. When the SUPERKEY is pressed, the softkeys take on new meanings associated with user programming functions.

Line Appearance Keys

There are 12 Line Appearance keys arranged in two columns. The first Line Appearance Key is always programmed as the Prime Line. The liquid crystal display (LCD) indicator provides the status of each Line Appearance. For CDE, the keys are numbered as follows:

12	11
10	09
08	07
06	05
04	03
02	01

These keys may be programmed to access specific lines and specific PABX features. These keys can be designated during CDE or they can be programmed from the set; refer to the *Features Description* Practice.

Speed Call Keys

Any unused Line Appearance key may also be programmed as a personal Speed Call key by the user from the *SUPERSET 420* telephone.

SUPERSET 420 Telephone Installation Instructions

6.4 Installation includes unpacking the set, assembling the set and connecting the set to a wall jack. The connections to the system via the cross connect field are described in the *Installation Information* Practice.

WARNING: **ANY CONNECTION OF THIS SET TO AN OFF PREMISE APPLICATION, AN OUT OF PLANT APPLICATION, OR TO ANY OTHER EXPOSED PLANT APPLICATION MAY RESULT IN A SAFETY HAZARD, AND/OR DEFECTIVE OPERATION, AND/OR EQUIPMENT DAMAGE.**

CAUTION: **Do NOT connect *SUPERSET 420* telephones in parallel, to standard lines, or as power fail transfer extensions. Do not use a hand test telephone (butt-in) to check a *SUPERSET 420* telephone line (its DNI line card does not have a loop detector). The set's on/off hook status is signaled by data transmission.**

Before the set can be used, the system must be programmed and equipped with DNI Line cards to interface with a *SUPERSET 420* telephone. Refer to the *Customer Data Entry Practice.*

Chart 6-1 *SUPERSET 420* Telephone Installation Instructions

Step	Action
1.	Connect the handset cord to the handset and main assembly (see Figure 6-3).
2.	Write the set's telephone number on the Brand/Extension Card.
3.	Identify the set's extension number, other line appearances and features on the Designation card.
4.	Install the Brand/Extension Card, the Designation Card and the Feature Designation Card and their protective lenses onto the main assembly.
5.	Connect the line cord to the telephone and the telephone jack.
6.	Insert the line cord and the handset cord into the cord channels.
7.	The Message Lamp and the Microphone Lamp will go solid red and the triangular line appearance indicators will be on for about five seconds. When they go out and the LCD displays the time and date, communication to the PABX has been established.

Table 6-1 *SUPERSET 420* Telephone Environmental Specifications

Operating Environment:		
	Temperature:	0° to 35° C (32° to 95° F).
	Humidity:	0% to 90% RH, non-condensing.
Storage/Shipping Environment:		
	Temperature:	-25° to 70° C (-13° to 158° F).
	Humidity:	0% to 90% RH, non-condensing.

Wall-Mounting the *SUPERSET 420* Telephone

6.5 To wall-mount the *SUPERSET 420* telephone, reverse the detachable base and plug the line cord into a modular jack. The "cord winder" in the base stores extra cord (see Figure 6-3). The wall-mount instructions can be ordered through normal distribution channels.

Visual Indication Of System Communications Problems

6.6 If there is a synchronization error, the LCD displays "NO SYNCHRONIZATION" in the uppermost row of the display. If there is a communication error, the display reads "NO COMMUNICATION". If the *SUPERSET 420* is installed on a circuit that is not programmed in CDE as a *SUPERSET 420,* the message "INVALID SET TYPE" and "CHECK CDE" will appear on the display.

Connection to the *MILINK* Module

6.7 To accommodate data call operations, the *SUPERSET 420* is equipped with a *MILINK* network connection. This allows the set to be connected to a *MILINK* Data Module on the desktop using only one telephone port. For *MILINK* Data Module installation instructions, refer to Dataset 1100 Series section in this Practice.

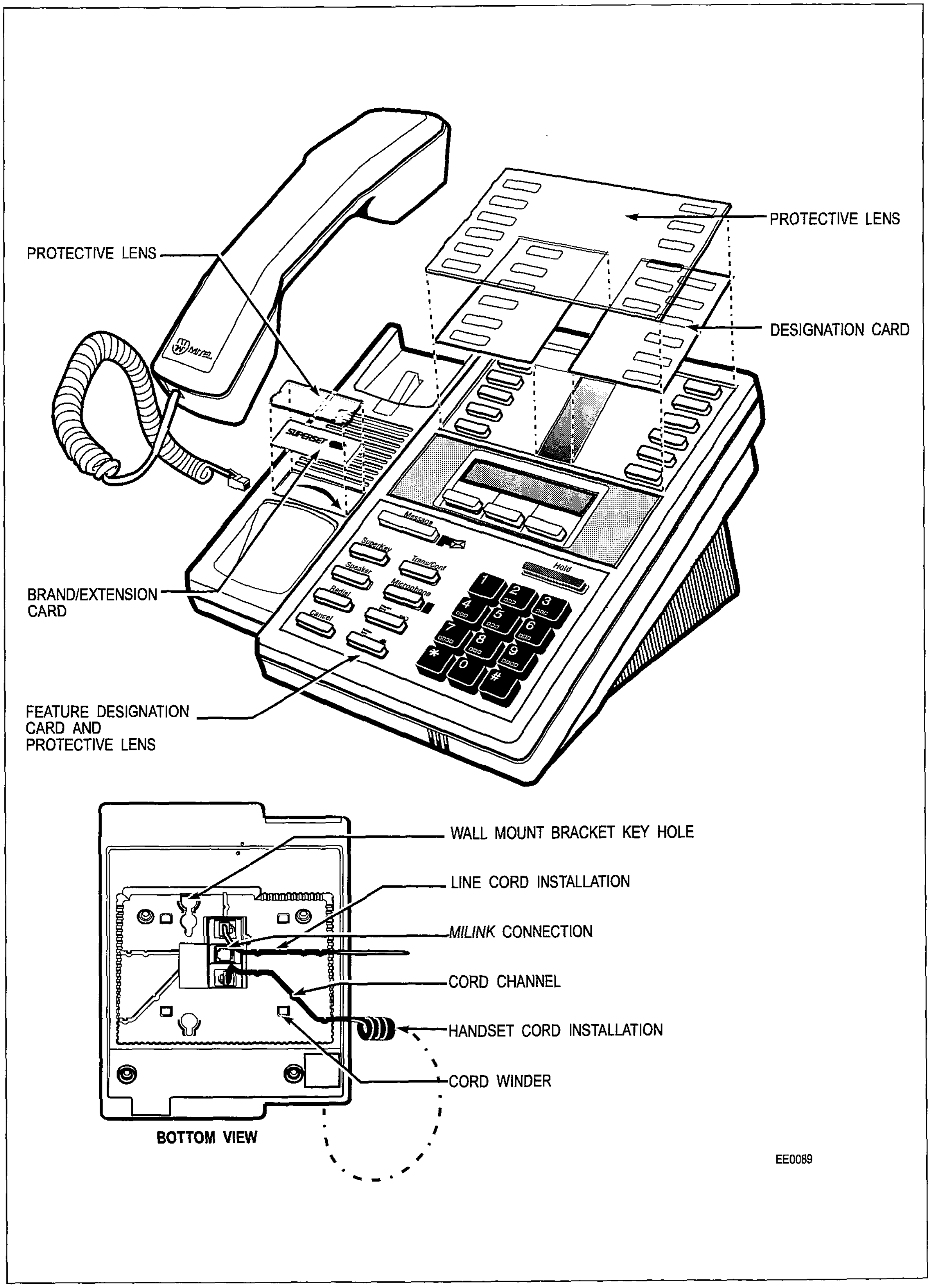


Figure 6-3 *SUPERSET 420* Installation Instructions

Peripheral Devices

7 *SUPERSET 430* Telephone

General Description

7.1 The *SUPERSET 430* telephone is a DNIC-based digital telephone that provides many features to enhance communications in a modern business environment. It is a highly functional executive display set that includes an integral alphanumeric LCD display with controllable contrast. Figure 7-1 shows a *SUPERSET 430* telephone set while Figure 7-2 shows the layout of the faceplate. A description of the available features is included in the *Features Description Practice*.

The *SUPERSET 430* telephone has the following features:

- 12 Line Select/Speed Call/Feature keys with individual line status indicators
- seven Fixed Function keys, two with associated LEDs
- softkey operation of most features (six softkeys; softkey prompts display the actions which are valid at any given time)
- alpha numeric Liquid Crystal Display with controllable contrast
- automatic selection of prime line
- key selection of non-prime line
- automatic ringing line selection
- single key feature activation
- handsfree and auto-answer
- Hold key
- Message Waiting Lamp LED
- programmable Call Forwarding and Speed Call keys
- speaker and ringer volume controls
- ringer pitch control
- digital interface with the system via the DNI Line Card
- built-in 4-function calculator
- optional data access via the *MILINK* Data Module, for integrated voice and data communications over a single twisted pair
- built-in hearing aid compatibility in the handset.

The *SUPERSET 430* telephone is designed for fast, simple installation. It connects to local area wiring by a modular jack, and to the PABX by a single twisted pair. Power, signaling, voice, and data are carried over this pair; additional wiring is not required.

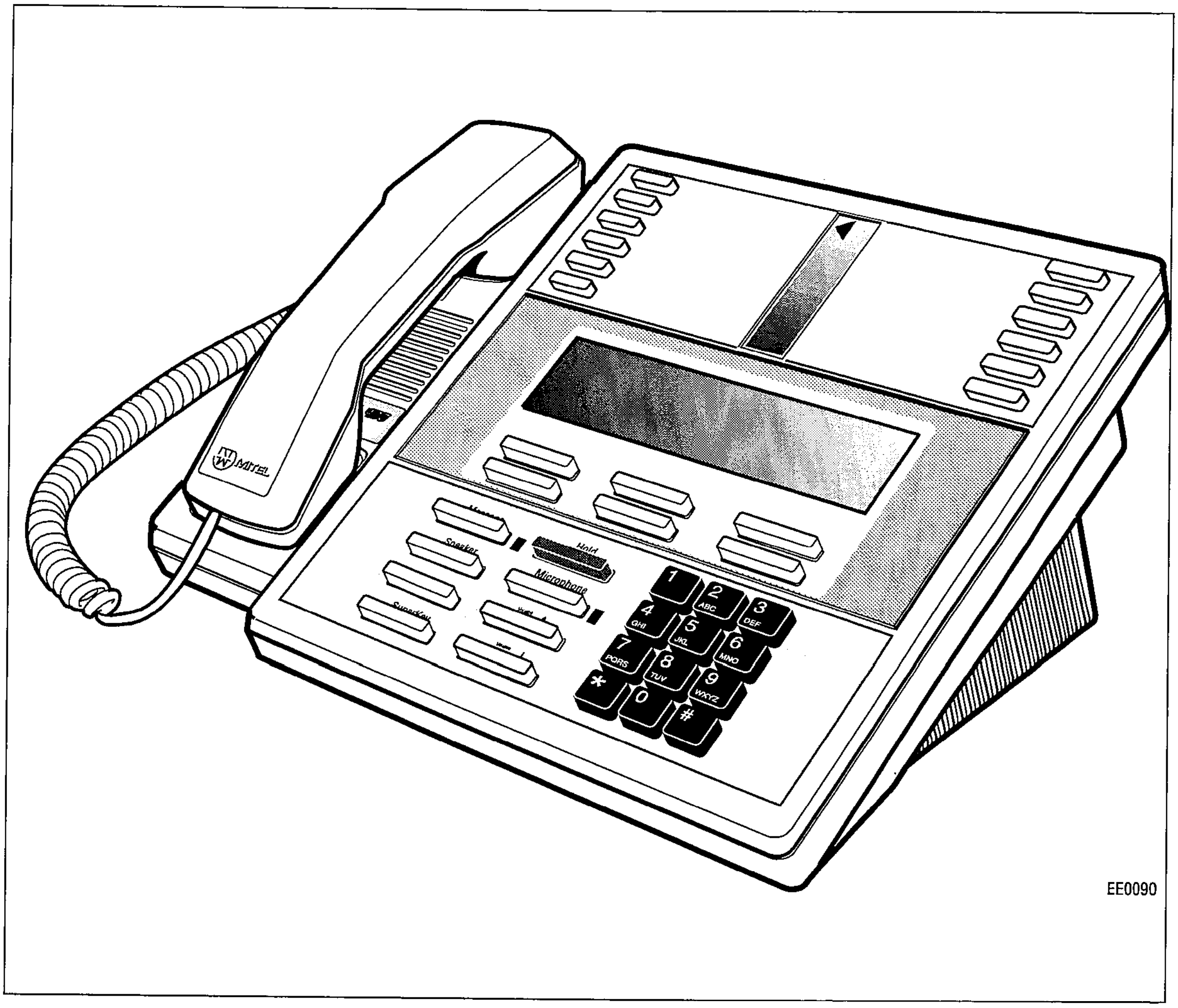


Figure 7-1 ***SUPERSET 430*** **Telephone**

Physical Description

General

7.2 The dimensions of the *SUPERSET 430* telephone are show in Figure 7-2. The faceplate of the telephone is divided into five functional areas:

- handset, speaker, and microphone
- line appearance keys with status indicators
- softkeys and LCD display
- telephone keypad
- fixed function keys.

Handset, Speaker and Microphone

The handset mounts in a cradle on the left side of the telephone; the speaker is located beneath a grille between the cradle depressions. The microphone is located in the bottom right edge of the telephone, below the " 0 " on the keypad.

Line Appearance Keys

There are 12 Line Appearance keys arranged in two columns. They are located in the top section of the telephone's faceplate. These keys may be programmed to access specific lines, specific features, or to activate Speed Call numbers. Between the two columns of Line Appearance keys is a liquid crystal display (LCD) indicator that provides the status of each line appearance.

Softkeys and LCD Display

In the center of the faceplate is a Liquid Crystal Display (LCD) that contains four lines of 40 characters each. The top two lines contain system messages; the bottom two lines contain the current softkey definitions. The Volume Up and Volume down fixed function keys at the bottom of the set provide contrast control for varying the intensity of the display. The six blue softkeys are located directly below the LCD display.

Telephone Keypad

The telephone keypad has 12 keys arranged in the standard pattern. It is in the bottom right-hand section of the faceplate. The number keys have associated letters with a slight deviation from the standard: the letter "q" appears on the 7 key; the letter z" appears on the 9 key. The following list illustrates the key number designation and the corresponding letter designations:

1	2 abc	3 def
4 ghi	5 jkl	6 mno
7 pqrs	8 tuv	9 wxyz
*	0	#

Fixed Function Keys

The eight fixed function keys are arranged to the left of the telephone keypad. The Message Waiting Lamp and the Microphone Lamp have LED indicators. One of the fixed function keys is not labeled. It is reserved for future use.

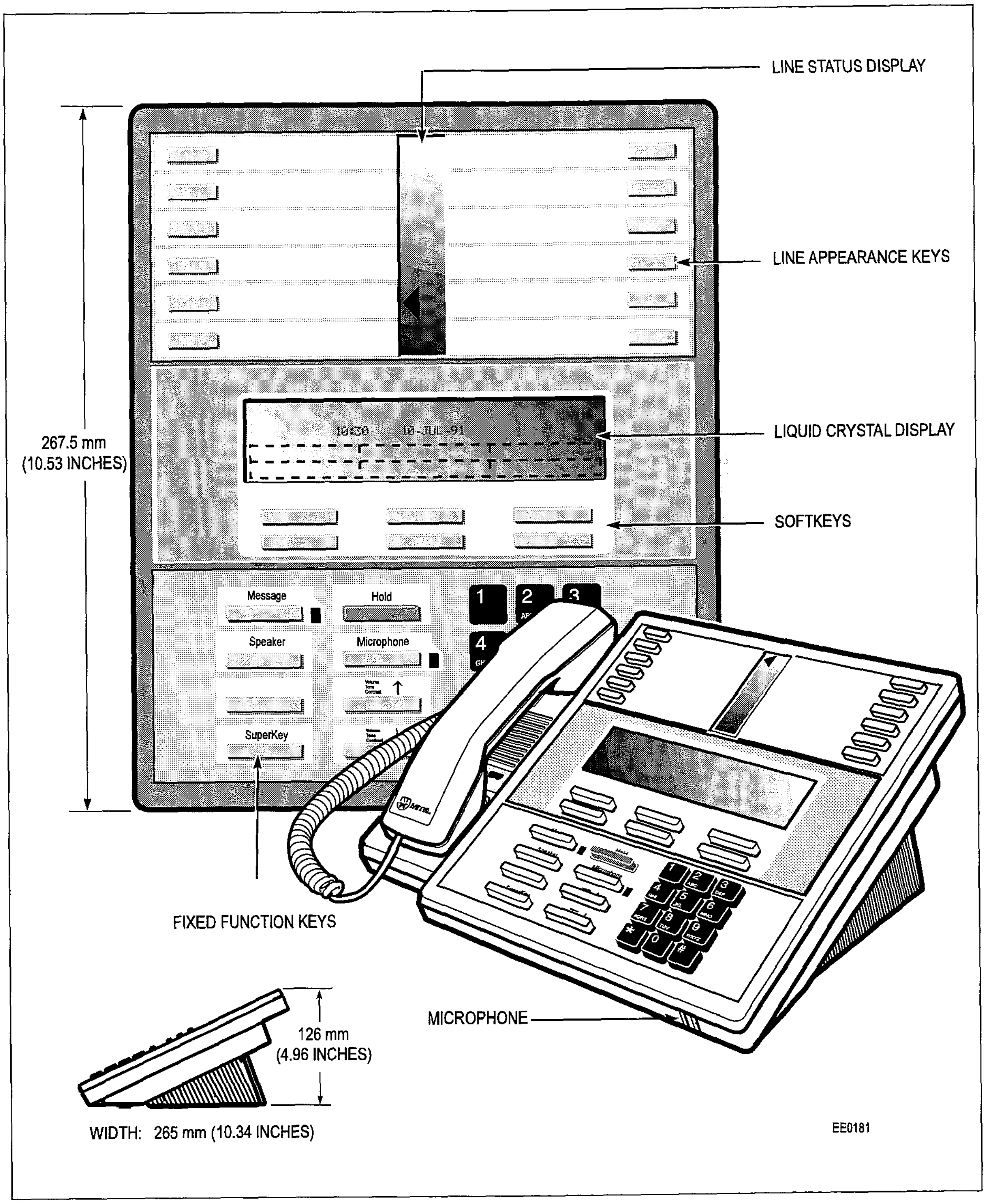


Figure 7-2 *SUPERSET 430* Telephone Dimensions

Functional Description

Fixed Function Keys

7.3 The Fixed Function keys located to the left of the telephone keypad are:

HOLD Key: Pressing the red Hold key places the current call on hard hold. The call may be retrieved by pressing the line select key of the call on hold (next to the flashing line select status indicator).

MESSAGE Key: The Message key is used to send a callback message to another telephone set capable of receiving the message when that set is busy, unanswered, or in Do Not Disturb mode. It is also used to initiate a return call to a telephone set that has left a message.

Note: A Message softkey is also provided. Pressing either the Message hard key or the Messaging soft key provides the same functionality.

MESSAGE Lamp: The Message lamp flashes whenever a message is waiting. The LCD shows the prompts necessary to return the call. The lamp is lit (solid red) when the *SUPERSET 430* calls a busy or unanswered party capable of receiving a message. Pressing the message key sends a message to the called set. After the message is sent, the lamp goes out and the call is terminated.

SPEAKER ON/OFF Key: Pressing this key once switches the speaker on and selects the prime line. Calls can be originated and/or received handsfree. Successive operation of this key toggles the set between on/off hook states.

MICROPHONE Key and MICROPHONE Lamp: The microphone is turned on by the system software when a handsfree call is activated. The Microphone key is used to switch the microphone off during handsfree mode so that the conversation near the set is not overheard by the caller. Successive operation of the Microphone key toggles the microphone between ON and OFF.

The Microphone lamp is lit (solid red) when the microphone is ON.

SUPERKEY Key: The SUPERKEY and the softkeys are used to program any unused Line Appearance Keys as Feature keys or Speed Call keys.

VOLUME UP and DOWN Keys: These keys are used to set levels for: ringing volume, ringing pitch, handsfree volume, background music volume, and LCD contrast. Each key depression changes the volume level or pitch by one step. The level for each function (except for handset volume) is stored in the system; changing one function setting does not affect the others. On system or set resets, the levels are restored. Default levels are applied when the set is first programmed. On each new call the handset volume is restored.

BLANK Key: This hard key has no function at the present time. It is reserved for future functionality.

LCD Display

The LCD Display gives you the following information:

- time and date when the set is idle
- softkey labels during programming and feature access
- call status during telephone calls
- message information
- telephone system error messages.

Softkeys and SUPERKEY

Immediately below the LCD are six unlabeled blue softkeys. Their functions are defined by corresponding prompts displayed on the LCD. The softkeys are used during telephone operations to perform functions or to access system features. The keys' functions vary with the activities of the set. The SUPERKEY is a fixed function key, which is located at the bottom left side of the set. When the SUPERKEY is pressed, the softkeys take on new meanings associated with user programming functions.

Line Appearance Keys

There are 12 Line Appearance keys arranged in two columns. The first Line Appearance Key is always programmed as the Prime Line. The liquid crystal display (LCD) indicator provides the status of each line appearance. For CDE, the keys are numbered as follows:

12	11
10	09
08	07
06	05
04	03
02	01

These keys may be programmed as Line Appearance keys or as Feature Access keys. These functions are designated during CDE and cannot be programmed from the set.

Some CDE-programmable feature keys are not valid on the *SUPERSET 430* telephone; these features are provided through the SUPERKEY. Attempting to program an invalid feature will show "INVALID FEATURE KEY FOR THIS TELEPHONE" when the DISPLAY KEYS function is activated.

Speed Call Keys

Any unused Line Appearance key may also be programmed as a personal Speed Call key by the user from the *SUPERSET 430* telephone.

Ringer Pitch Adjustment

Ringer pitch and cadence combination adjustment is one of the *SUPERSET 430* telephone's built-in functions. To adjust:

- Press SUPERKEY.
- Press the MORE... softkey.

- Press the RING ADJUST softkey.
- Press the RINGER PITCH softkey.
- The set will start ringing.
- Adjust the ringer pitch. The VOL UP key increases the pitch; the VOL DOWN key decreases the pitch.
- Press the SUPERKEY. The pitch setting is saved, the ringer stops, and the set returns to its normal state.

SUPERSET 430 Telephone Installation Instructions

7.4 Installation includes unpacking the set, assembling the set and connecting the set to a wall jack. The connections to the system via the cross connect field are described in the *Installation Information Practice*.

WARNING: **ANY CONNECTION OF THIS SET TO AN OFF PREMISE APPLICATION, AN OUT OF PLANT APPLICATION, OR TO ANY OTHER EXPOSED PLANT APPLICATION MAY RESULT IN A SAFETY HAZARD, AND/OR DEFECTIVE OPERATION, AND/OR EQUIPMENT DAMAGE.**

Do NOT connect *SUPERSET 430* telephones in parallel, to standard lines, or as Power Fail Transfer extensions. Do not use a hand test telephone (butt in) to check *SUPERSET 430* telephone lines; the Digital Line Card does not have a loop detector. The on-/off-hook status of the set is signaled by data transmission.

Before the set can be used, the system must be programmed and equipped with DNI Line Cards to interface with a *SUPERSET 430* telephone. Refer to the *Customer DataEntry Practice.*

Chart 7-1 *SUPERSET 430* Telephone Installation Instructions

Step	Action
1.	Connect the handset cord to the handset and the main assembly.
2.	Write the set's telephone number on the Brand/Extension Card.
3.	Identify the set's extension number, other line appearances and features on the Designation Card.
4.	Install the Brand/Extension Card, the Designation Card and the Feature Designation Card. Install their protective lenses onto the main assembly.
5.	Connect the line cord to the telephone and the telephone jack.
6.	Insert the line cord and the handset cord into the cord channels.
7.	The Message Lamp and the Microphone Lamp will go solid red and the triangular line appearance indicators will be on for about five seconds. When they go out and the LCD displays the time and date, communication to the PABX will have been established.

Table 7-1 *SUPERSET 430* Telephone Environmental Specifications	
Operating Environment:	
Temperature:	0° to 35° C (32° to 95° F).
Humidity:	0% to 90% RH, non-condensing.
Storage/Shipping Environment:	
Temperature:	-25° to 70° C (-13° to 158° F).
Humidity:	0% to 90% RH, non-condensing.

Visual Indication of System Communications Problems

7.5 If there is a synchronization error, the LCD displays "NO SYNCHRONIZATION" in the uppermost row of the display. If there is a communication error, the display reads "NO COMMUNICATION". If the *SUPERSET 430* is installed on a circuit that is not programmed in CDE as a *SUPERSET 430* the message "INVALID SET TYPE" and "CHECK CDE" will appear on the first two lines of the display.

Connection to the *MILINK* Data Module

7.6 To accommodate data call operations, the *SUPERSET 430* telephone is equipped with a *MILINK* network connection. This connection allows the set to be connected to a *MILINK* Data Module on the desktop and share the one telephone port. For *MILINK* Data Module installation instructions refer to the Dataset 1100 Series Section in this Practice.

8 Programmable Key Module

General Description

8.1 The Programmable Key Module (PKM) provides *SUPERSET 410*, *SUPERSET 420*, and *SUPERSET 430* telephones with 30 additional personal keys. You can program these personal keys through CDE with the following functions:

- speed call keys
- feature keys
- key line appearances
- personal outgoing line keys
- key system appearances
- multi-call line appearances
- co line keys
- busy lamp field/direct trunk select keys.

The keys are arranged in two vertical rows on the module. Beside each key is a Line Status Display that indicates the status of the key. The keys are numbered as follows:

30	29
28	27
26	25
24	23
22	21
20	19
18	17
16	15
14	13
12	11
10	09
08	07
06	05
04	03
02	01

The flash rates for the Line Status Displays on the PKM are identical to those on the *SUPERSET 410*, *SUPERSET 420* and *SUPERSET 430* telephones. Figure 8-1 shows the PKM. Table 8-1 lists the environmental specifications for the PKM.

Table 8-1 PKM Environmental Specifications

Operating Environment:	
Temperature:	0° to 35° C (32° to 95° F).
Humidity:	0% to 90% RH, non-condensing.
Storage/Shipping Environment:	
Temperature:	-25° to 70° C (-13° to 158° F).
Humidity:	0% to 90% RH, non-condensing.

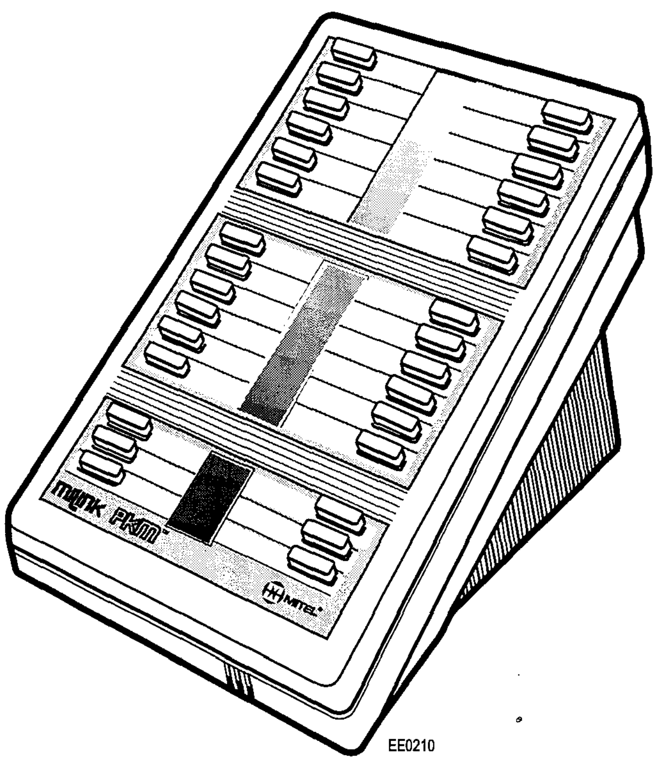


Figure 8-1 Programmable Key Module (PKM)

Installation Instructions

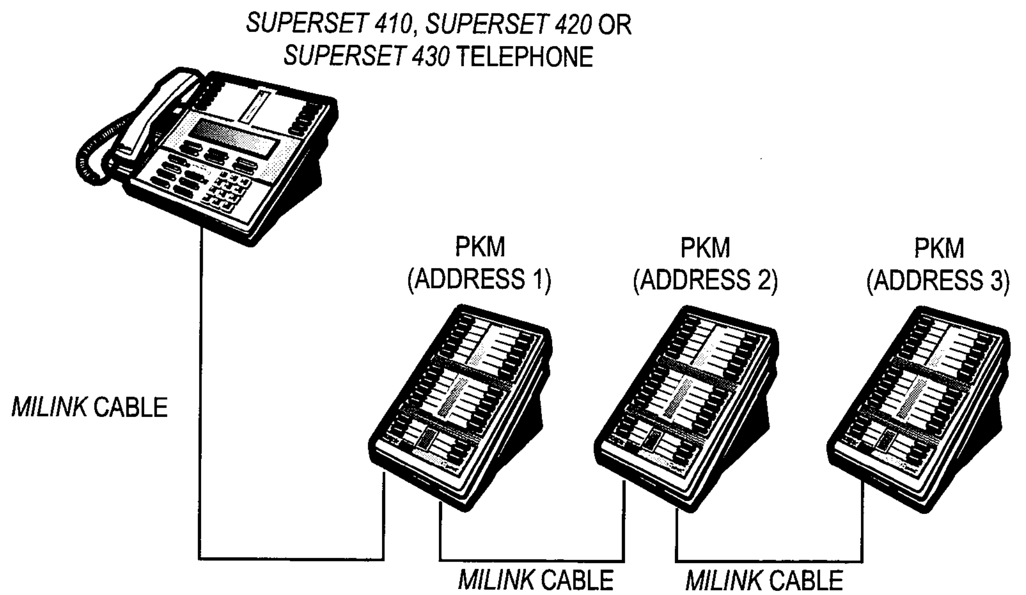
8.2 Installation involves unpacking the PKM, assembling the PKM, and connecting it to a *SUPERSET 410*, *SUPERSET 420*, or *SUPERSET 430* telephone.

WARNING: **ANY CONNECTION OF THIS SET TO AN OFF PREMISE APPLICATION, AN OUT OF PLANT APPLICATION, OR TO ANY OTHER EXPOSED PLANT APPLICATION MAY RESULT IN A SAFETY HAZARD, AND/OR DEFECTIVE OPERATION, AND/OR EQUIPMENT DAMAGE.**

Up to three PKMs can be used with a set. Each *SUPERSET 410* telephone, *SUPERSET 420* telephone, *SUPERSET 430* telephone, has one *MILINK* port. Each PKM has two *MILINK* ports. The first PKM connects to the *MILINK* port located on the base of the set. The other two PKMs are connected in series (daisy-chained together) to the first. You can connect a *MILINK* Data Module anywhere in the chain of PKMs.

Chart 8-1 PKM Installation Instructions

Step	Action
1.	Before installing the PKM(s), program the *SUPERSET 410*, *SUPERSET 420*, or *SUPERSET 430* telephone with the required number of PKMs. You should also program the PKM personal keys. Refer to the *Feature Descriptions Practice*, and the *Customer Data Entry Practice*, for instructions.
2.	Unpack the shipping container. Each PKM that you ordered should have the following parts: • PKM unit (including a key overlay and plastic lens) • a 0.3 meter *MILINK* cable • an AC power adaptor (MITEL PN 9112-200-001-NA).
3.	Write the extension numbers of the line appearance keys and the functions of the feature keys on the key overlays.
4.	Position the key overlays on the PKMs and install the clear plastic lenses.
5.	Position the PKMs next to the telephone on a flat surface. The PKMs must be located near 120 Volt electrical outlets for the power adaptors.
6.	Set the address switches on the base of each PKM to the required settings as indicated in Figure 8-2. Each PKM must have a different address.
7.	Connect the *MILINK* cables between the telephone and PKMs as shown in Figure 8-2. • Plug one end of the *MILINK* cable into the offset latch, modular socket located on the base of the telephone and plug the other end into into either of the two offset latch, modular sockets on the base of PKM1. Figure 8-3 shows how to connect the cables to PKM1. • Connect PKM2 to PKM1 in the same manner, if required. • Connect PKM3 to PKM2, if required.
8.	Plug the AC power adaptors into 120 Volt AC electrical outlets.
9.	Insert the AC power adaptor input plug into the jack on the base of each PKM.
10.	Observe the status indicators on each PKM when you plug in the AC power adaptor plug. All the status display indicators should turn on for approximately half a second, to enable you to verify that all indicators are working. The indicators then flash in unison until communication with the system is established. If communication is established quickly, you may miss seeing the indicators flash.



NOTE: THE ADDRESS DIP SWITCHES ARE LOCATED ON THE BASE OF THE PKM.

ADDRESS SWITCH SETTINGS		
PKM ADDRESS	**SWITCH A**	**SWITCH B**
ADDRESS 1	OPEN	CLOSED
ADDRESS 2	CLOSED	OPEN
ADDRESS 3	CLOSED	CLOSED
ADDRESS 3	OPEN	OPEN

EE0216

Figure 8-2 PKM Address Switch Settings

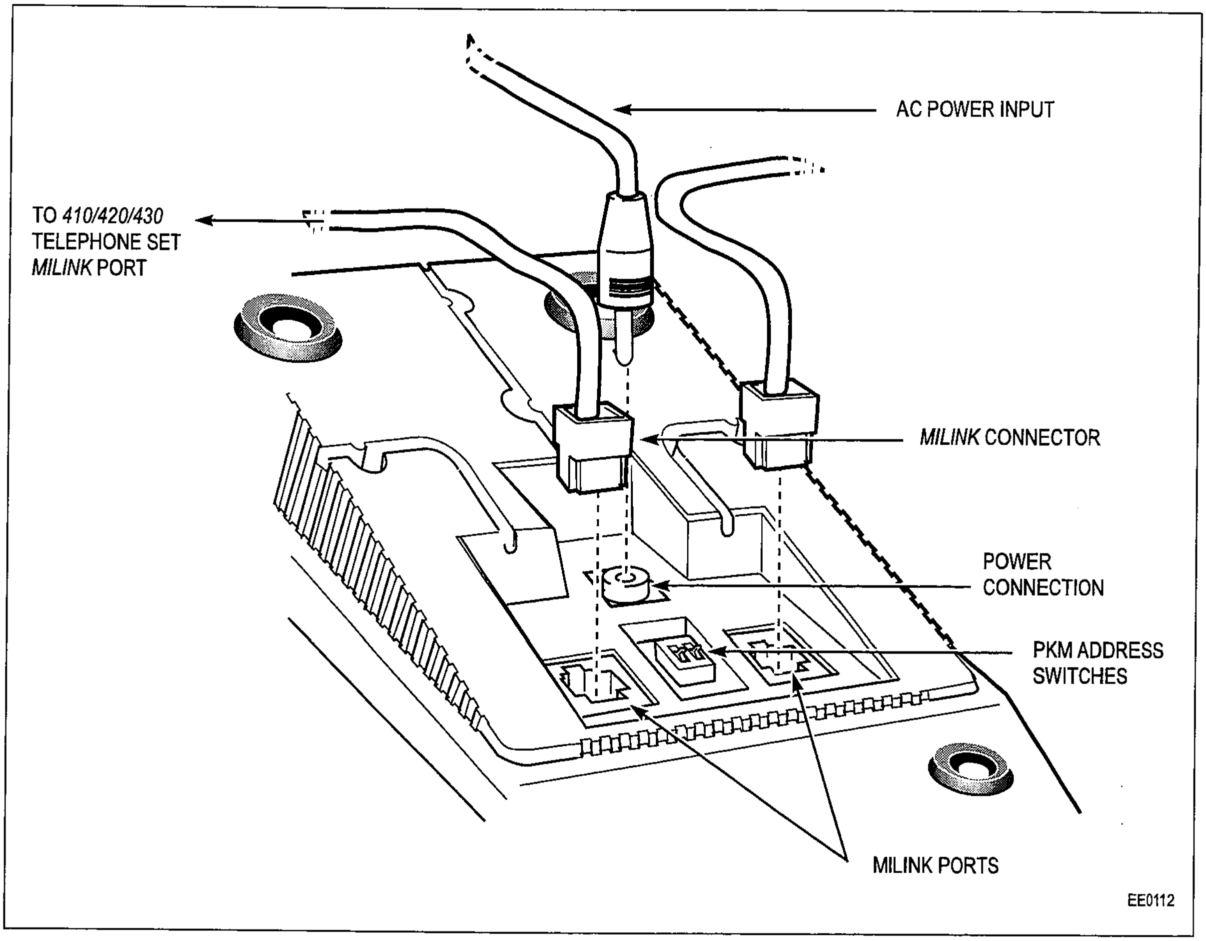


Figure 8-3 Cable Connections for PKM1

Visual Indication of Error Conditions

8.3 Table 8-2 provides some basic troubleshooting information for the PKM.

Table 8-2 PKM Error Conditions

Symptoms	Error Condition	Possible Corrective Action
PKM status display indicators 1, 5, 9, 13, 17, 21, 25 & 29 flash; then indicators 2, 6, 10, 14, 18, 22, 26, & 30. This flash cycle is repeated at a slow rate.	No synchronization signal is present at the PKM.	Check that the *MILINK* cable is connected properly. Try connecting the cable to other *MILINK* ports on the PKM (a connector pin may be bent or broken). Use a different *MILINK* cable. Verify that the PKM is programmed correctly in CDE.
All the PKM status display indicators flash in unison.	The system is not communicating with the PKM.	Verify that the PKM is programmed correctly in CDE.
The PKM status display indicators appear to be flashing quickly and randomly.	The PKM doesn't have access to the *MILINK* data bus.	Disconnect the line cord from the base of the *SUPERSET* telephone. Wait 10 seconds and then connect the line cord to the set again. All the status display indicators should turn on for approximately half a second and enable you to verify that all indicators are working. The vertical columns of indicators flashes alternately until communication with the system is established. If communication is established quickly, you may miss seeing the indicators flash.

9 DATASET 1100 Series Description

The DATASET 1100 series asynchronous datasets are composed of two variants: a Standalone DATASET 1103 unit, and a *MILINK* Data Module (DS1101M). The DATASET 1100 series support asynchronous data communications at rates up to 19.2 kilobits per second. The DATASET 1103 interfaces with an *SX-200* DIGITAL PABX through a DNI Line Card within the PABX. The *MILINK* Data Module interfaces with an *SX-200* ML PABX via the *MILINK* Bus (available with specific software).

These datasets (except the *MILINK* Data Module) operate with the MITEL Digital Network Interface Circuit (DNIC) digital link technology which is composed of three channels: a 64-kbps voice or data (B) channel, a 64-kbps data (B) channel, and a 16-kbps (D) channel for control communications between the set and the PABX.

The Standalone DATASET 1103 interfaces peripheral devices to the PABX.

The *MILINK* Data Module is packaged in a flat case like the DATASET 1103 and is designed to fit under the *SUPERSET 410, SUPERSET 420*, and *SUPERSET 430* telephones. It interfaces peripheral devices to the PABX via the *SUPERSET* 400 series telephone sets.

Data terminal equipment (DTE) or data communications equipment (DCE) mode is selected by the use of a Modem Adapter plugged into RS-232 (EAI232) interface of the DATASET 1103 and *MILINK* Data Module.

Note: Fax Modems are not supported when connected through datasets.

Baud Rates

9.1 The DATASET 1100 series operate at the following baud rates:

110 - 150 - 200 - 300 - 600 - 1200 - 2400 - 4800 - 9600 - 19200

Connector Pin Assignment Tables

9.2 The RS-232 connectors used on the DATASET 1100 series have the following pin assignments when operating in Data Communications Equipment (DCE) mode:

Signal	Designation	Pin	Direction
	frame ground	1	
TXD	transmit data	2	to dataset
RXD	receive data	3	from dataset
RTS	request to send	4	to dataset
CTS	clear to send	5	from dataset
DSR	dataset ready	6	from dataset
	signal ground	7	
DCD	data carrier detect	8	from dataset
DTR	data terminal ready	20	to dataset
RI	ring indicator	22	from dataset
HW	flow control/RI in	25	to dataset

Standalone DATASET 1103 Description

9.3 A Standalone DATASET 1103 is packaged in a flat case which can be placed under a standard desk telephone. It is functionally the same as the DATASET 1101 Cartridge. The Standalone DATASET 1103 can be connected to the PABX using a four-wire connection; two wires connect the dataset to the Digital Line Card, and two different wires connect the telephone set Tip-Ring pair to an ONS or COV line card; it may also be connected to a Digital Line Card within the PABX by a single twisted pair (the telephone set is then connected independently). Figure 9-1 shows typical applications of a Standalone DATASET 1103 connected to a telephone set and to a personal computer or terminal. The DATASET 1103 case is 206 mm wide x 270 mm long x 35 mm high (8.1 in. x 10.6 in. x 1.4 in.).

Controls, Indicators, and Connectors

Its keys and LEDs are:

- ATTN key
- DISC key
- POWER LED (includes SYNC to PABX information)
- READY LED (includes RxD traffic information)
- DEVICE (DTR) LED (includes TxD traffic information)

Standalone DATASET 1103 RJ-11 modular connector pins are:

Line

Pin	Signal	Typical modular jack wire colors
1	no connection	
2	data Tip	black
3	voice Tip	red
4	voice Ring	green
5	data Ring	yellow
6	no connection	

Telephone

Pin	Signal	Typical modular jack wire colors
1	no connection	
2	no connection	
3	voice Ring	green
4	voice Tip	red
5	no connection	
6	no connection	

Installation

WARNING: **ANY CONNECTION OF THIS DEVICE TO AN OFF PREMISE APPLICATION, AN OUT OF PLANT APPLICATION, OR TO ANY OTHER EXPOSED PLANT APPLICATION MAY RESULT IN A SAFETY HAZARD, AND/OR DEFECTIVE OPERATION, AND/OR EQUIPMENT DAMAGE.**

Install a Standalone DATASET 1103 by placing it near its associated telephone set and data peripheral (its case is designed to lie flat on the desk under a standard telephone).

Complete Customer Data Entry for the system to allow the Standalone DATASET 1103 cartridge to operate with the PABX.

Connect the telephone set cord to the jack marked PHONE; connect the modular telephone line cord to the jack marked LINE and to the wall or floor jack which connects to the PABX main distributing frame (MDF). The connection to the MDF is a four-wire circuit; the voice Tip and Ring (red and green) connect to an ONS or COV line card (depending on the type of set being used), and the dataset Tip and Ring (yellow and black) connect to a DNI Line Card.

An alternative Tip and Ring wiring arrangement is: connect the telephone set through a two-wire connection to its ONS or COV line card; connect the modular telephone line cord (data Tip and Ring) to the jack of the Standalone DATASET 1103 marked LINE, through the MDF to its DNI Line Card in the PABX.

Plug the power supply into an ac outlet and its cord into the power socket on the Standalone DATASET 1103. All LEDs (except POWER) should flash 5 times, indicating that the dataset has passed its self-test. If this does not happen, the dataset is faulty and must be replaced.

The POWER LED should be on continuously, indicating connection to the Digital Line Card. If the POWER LED is flashing, check the wiring between the dataset and the PABX, and Customer Data Entry (CDE) for the device.

Plug the RS-232 interconnecting cable into the RS-232 connector at the back of the Standalone DATASET 1103 and connect its other end to the peripheral device. When the RS-232 connector is attached, the DEVICE LED should be on, unless the device is an auto-answer device, and defined as such in CDE (usually only destination devices such as host computers are auto-answer). If the device is not auto-answer, and the DEVICE LED does not come on, do the following:

1. Make sure the device is powered on.
2. Use an RS-232 break-out box to verify that the device provides DTR. If the device does not provide DTR, select CDE option "RS-232 Force High" for that device.

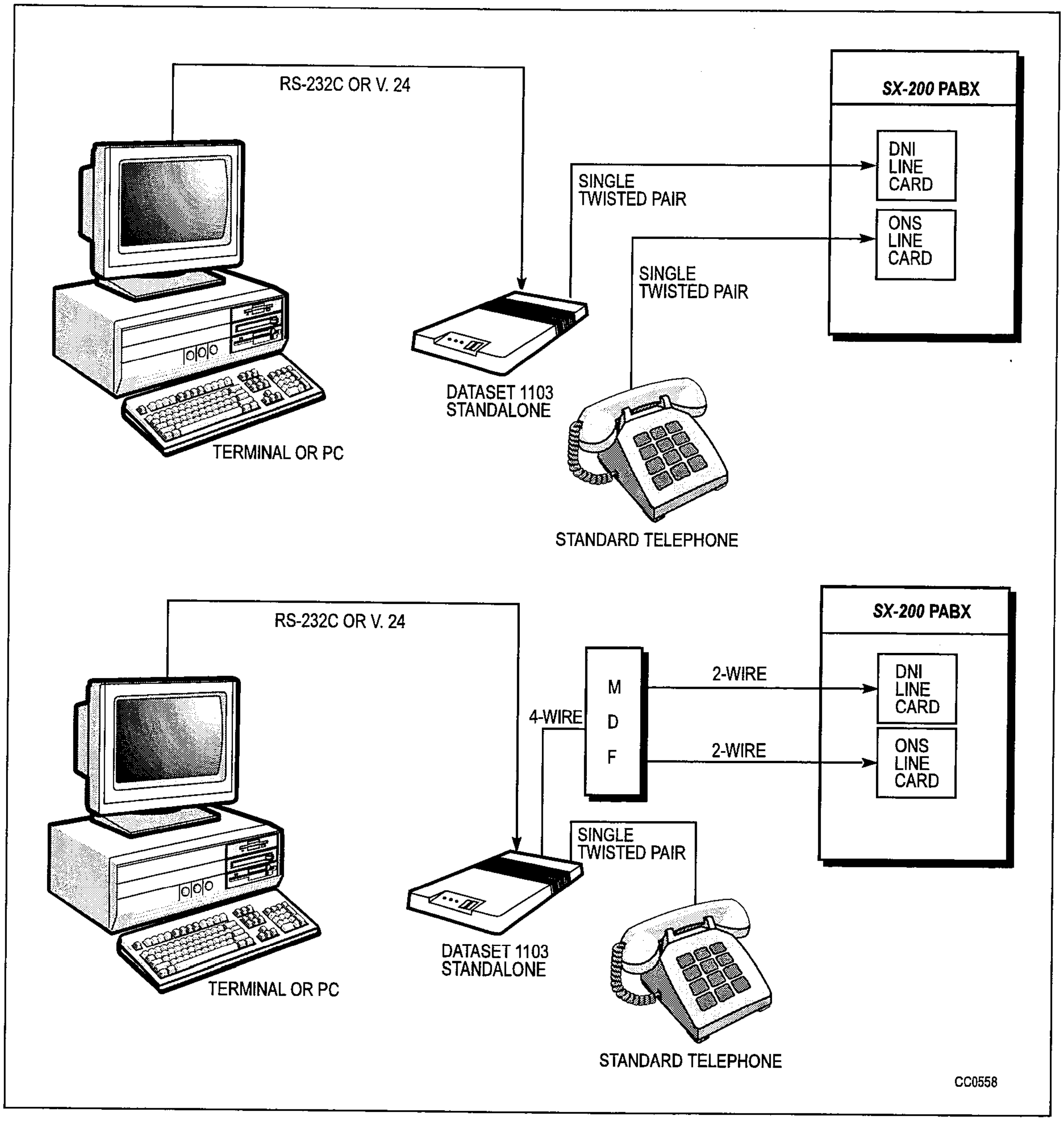


Figure 9-1 Typical Standalone DATASET 1103 Applications

Circuit Description

9.4 The microprocessor controls the operations within the dataset. It adds High-level Data Link Control (HDLC) control bytes to incoming data received through its RS-232 port, before the data is passed to the HDLC Controller to be packetized and sent out via the DNIC to the PABX. Similarly data received through the PABX has its address and HDLC control bytes stripped off before it is sent out on the RS-232 port. Once a call is established by the use of D-channel signals to the PABX, data communication occurs on the B-channel, and passes straight through the PABX to its destination. Figure 9-2 is a block diagram of the DATASET 1100 series.

The LEDs and keys, which interface the DATASET 1100 series to the user, connect to the microprocessor.

The external random access memory (RAM) provides buffering for error correction and speed conversion, as well as improving flow control through the DATASET 1100 series. Error correction is by retransmitting frames in which errors had been detected.

The HDLC controller is a single channel interface between the microprocessor and the DNIC/*MILINK*; control signals are sent on the D-channel, while data is sent on the B2-channel.

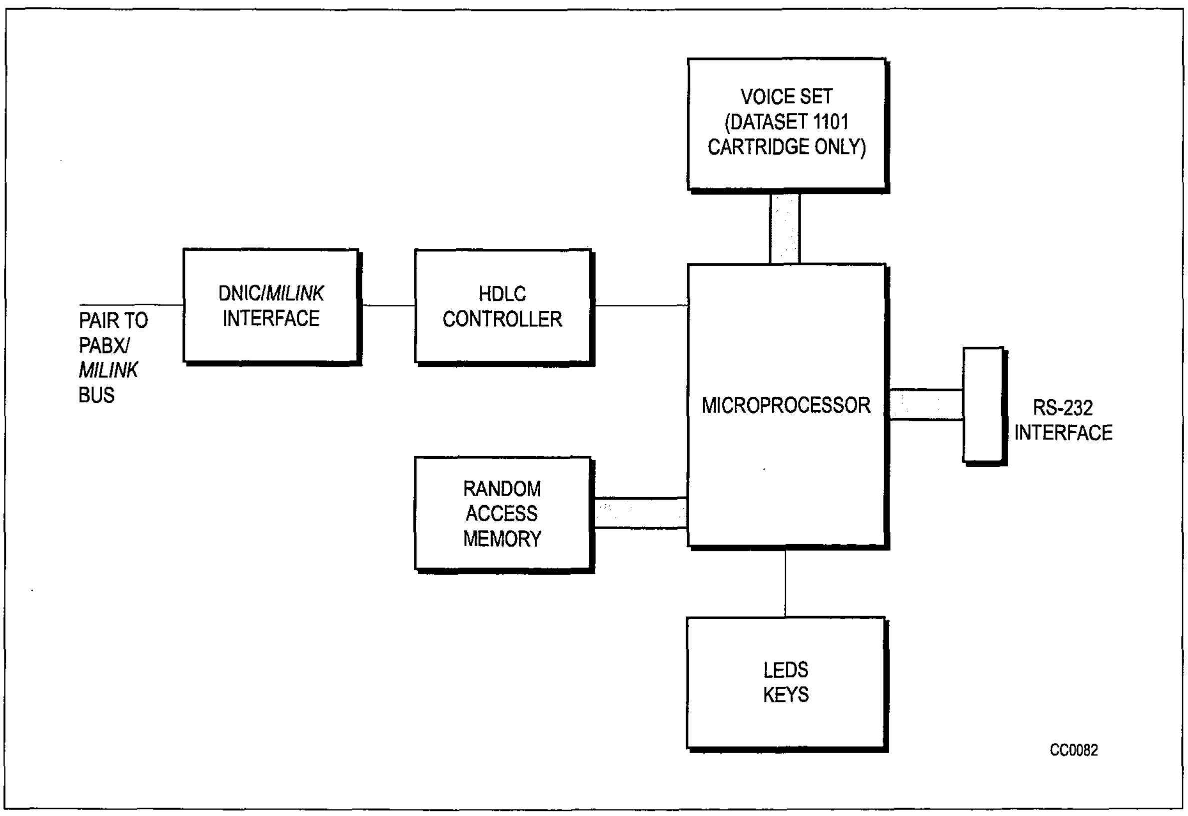


Figure 9-2 Typical DATASET 1100 Series Block Diagram

MILINK Data Module

General Description

9.5 The standalone *MILINK* Data Module (1101M) is packaged in a flat case and is designed to be placed under multiline *SUPERSET 410, SUPERSET 420,* and *SUPERSET 430* telephones. It has its own ATTENTION and DISCONNECT keys like the DATASET 1103.

Peripheral Devices

Table 9-1 Controls, Indicators, and Connectors
KEYS AND LEDs
- ATTN key - DISC key - POWER LED (includes SYNC to system information) - READY LED (includes RxD traffic information) - DEVICE (DTR) LED (includes TxD information)

Table 9-2 Connector Pin Assignment Table

Signal	Designation	Pin	Direction
	frame ground		
TXD	transmit data	2	to data set
RXD	receive data	3	from data set
RTS	ready to send	4	to data set
CTS	clear to send	5	from data set
DSR	dataset ready	6	from data set
	signal ground	7	
DCD	data carrier detect	8	from data set
DTR	data terminal ready	20	to data set
RI	ring indicator	22	from data set
	reserved input/ test mode output	25	

Installation

WARNING: **ANY CONNECTION OF THIS DEVICE TO AN OFF PREMISE APPLICATION, AN OUT OF PLANT APPLICATION, OR TO ANY OTHER EXPOSED PLANT APPLICATION MAY RESULT IN A SAFETY HAZARD, AND/OR DEFECTIVE OPERATION, AND/OR EQUIPMENT DAMAGE.**

The following chart lists the steps required to install a *MILINK* Data Module.

Chart 9-1 *MILINK* Data Module Installation Instructions

Step	Action	Comment
1.	Verify that CDE for the *MILINK* Data Module (1101M) is complete.	
2.	Connect an AC adaptor between the AC outlet and *MILINK* Data Module.	All LEDs should flash five times, if not replace the *MILINK* Data Module.
3.	Connect *a SUPERSET 410, SUPERSET 420*, or *SUPERSET 430* telephone set to the *MILINK* Data Module at *MILINK* A or *MILINK* B (another *Milink* devide can be connected serially to the remaining *MILINK* connection).	POWER LED stops flashing
4.	Connect RS-232 cable between the *MILINK* Data Module and the peripheral device.	Device LED should be on, see note 1.

Note: **1.** When the RS-232 connector is attached, the DEVICE LED should be on, unless the device is an auto-answer device, and defined as such in CDE (usually only destination devices such as host computers are auto-answer). If the device is not auto-answer, and the DEVICE LED does not come on, do the following steps:

- Make sure that the device is powered on.
- Use an RS-232 break-out box to verify that the device provides DTR. If the device does not provide DTR, select CDE option DTR always for that device.

2. The POWER LED should be on continuously (except when first connected to AC power), indicating connection to the Digital Set. If the POWER LED is flashing, check the wiring between the *MILINK* Data Module and the set. If the POWER LED continues to flash, verify Customer Data Entry (CDE) for the device.

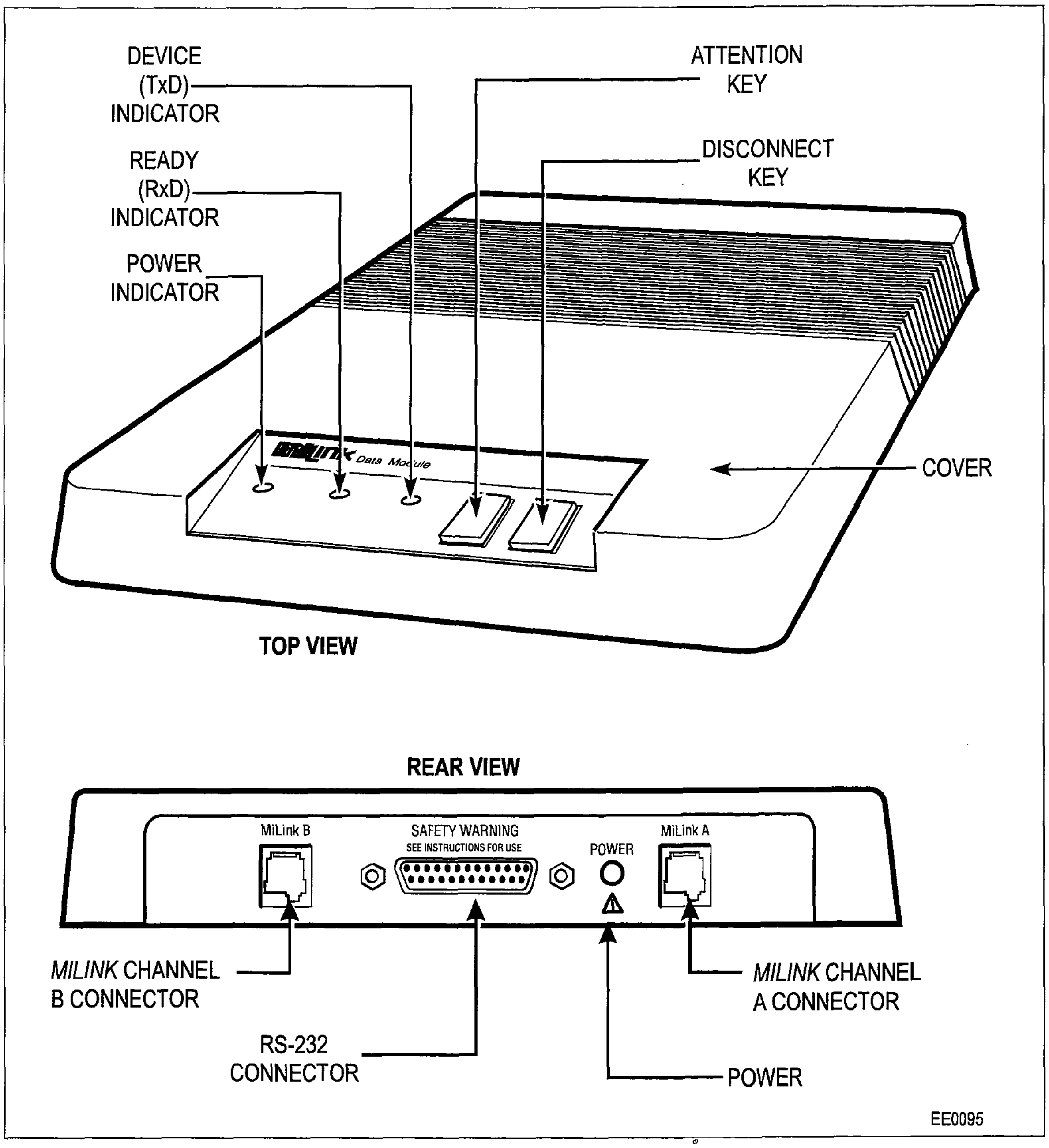


Figure 9-3 ***MILINK*** **Data Module, Connectors and Indicators**

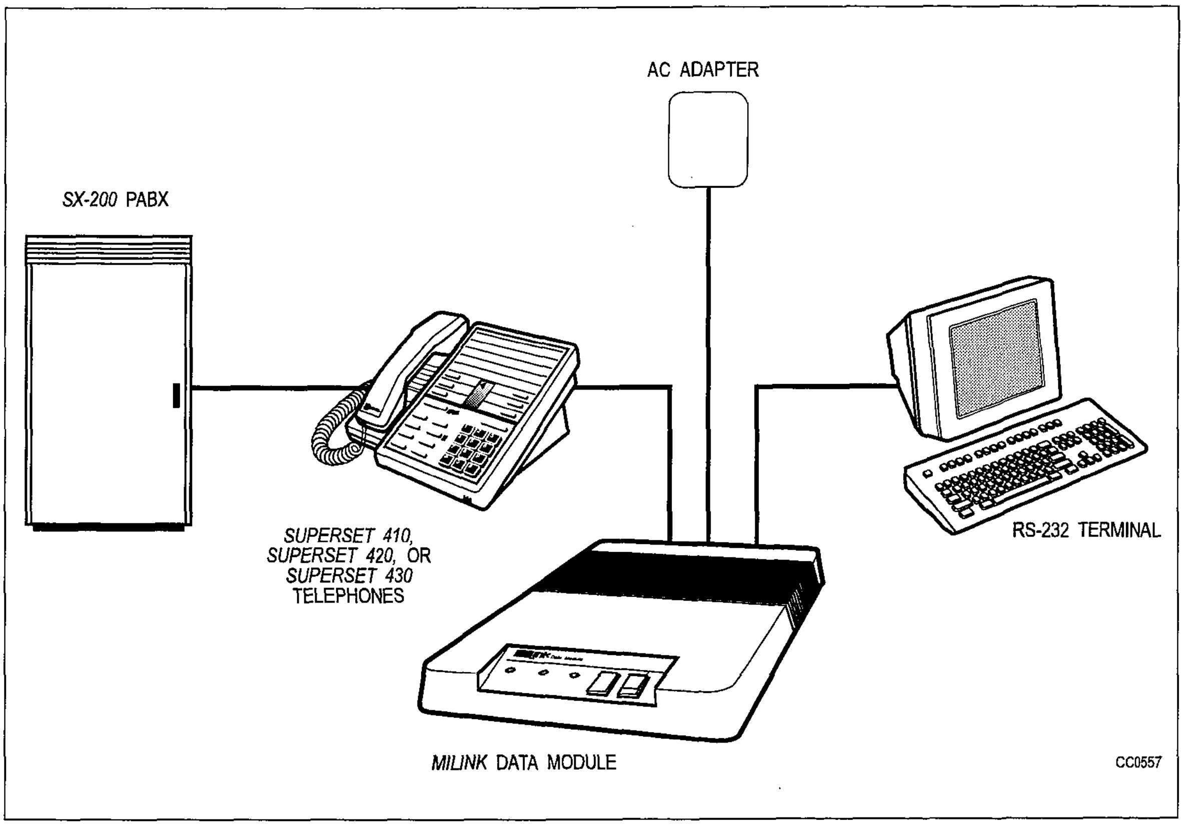


Figure 9-4 *MILINK* Data Module Installation

Power Supplies

9.6 The Standalone DATASET 1103 and the *MILINK* Data Module all receive their power from a plug-in transformer which supplies 9 Vac to a power connector on the back of the set. Circuitry within the dataset converts this voltage to the required dc voltages. The digital telephone set voice operation receives its power from the PABX.

10 DATASET 2100 Series Description

The DATASET 2100 Series is composed of a Standalone DATASET 2103. The DATASET 2100 Series supports synchronous and asynchronous data communications at rates up to 19.2 kilobits per second. This series interfaces with MITEL's proprietary Digital Network Interface Circuit (DNI) Line Card within the PABX.

These datasets operate with DNIC digital link technology which is composed of three channels: a 64 kbps data channel, a 64 kbps voice or data channel (B), and a 16 kbps (D) channel for control communications between the set and the PABX. Only one B channel is used with the DATASET 2100 Series.

The Standalone DATASET 2103 interfaces peripheral devices to the PABX.

Note: FaxModems are not supported when connected through datasets.

Environmental Specifications

Operating Environment:

Ambient Temperature: 4 to 49° C (39.2 to 120° F).
Ambient Humidity: 10 to 90% RH, noncondensing.

Storage/Shipping Environment:

Ambient Temperature: -40 to 60° C (-40 to 150° F).
Ambient Humidity: 10 to 90% RH, noncondensing.

Baud Rates

10.1 The DATASET 2100 Series operates at the following baud rates:

Asynchronous	Synchronous
110	1200
150	2400
200	4800
300	9600
600	19200
1200	
2400	
4800	
9600	
19200	

Connector Pin Assignment Tables

10.2 The RS-232 connector on the DATASET 2100 Series has the following pin assignments when operating in Data Communications Equipment (DCE) mode:

Pin	Signal	Designation	Direction
1		frame ground	
2	TXD	transmit data	to dataset
3	RXD	receive data	from dataset
4	RTS	request to send	to dataset
5	CTS	clear to send	from dataset
6	DSR	data set ready	from dataset
7		signal ground	
8	DCD	data carrier detect	from dataset
20	DTR	data terminal ready	to dataset
22	RI	ring indicator	from dataset
25		HW flow control	to dataset

The RS-232 connector on the DATASET 2100 Series has the following pin assignments when operating in Data Terminal Equipment (DTE) mode with an attached Modem Adapter:

Pin	Signal	Designation	Direction
1		frame ground	
2	TXD	transmit data	from dataset
3	RXD	receive data	to dataset
4	RTS	request to send	from dataset
5	CTS	clear to send	to dataset
6	DSR	data set ready	to dataset
7		signal ground	
8	DCD	data carrier detect	to dataset
15	DB	SCT	to dataset
17	DD	SCR	to dataset
19	n/a	remote digital loopback	from dataset
20	DTR	data terminal ready	from dataset
22	RI	ring indicator	to dataset
23	CH	data signal rate selector	from dataset
24	DA	XCLK	from dataset
25	CN	analog loopback	from dataset

Synchronous Operation

10.3 **Transparent** - The DATASET 2100 Series can operate in transparent synchronous mode. Because the connections are usually established for a long time (compared to asynchronous connections) and are important parts of a large data communications network, the D channel connection is held up while the connection is established. The network management process can obtain statistics on traffic and error rates while a call is in progress from this D channel. When one DATASET 2100 drops its connection, it cannot signal to the other; the PABX must drop this connection with a

Terminate B channel command, which it can do because the D channel is active while the B channel session is in progress.

Data is clocked to and from the shift registers at a constant rate by the synchronous clocks on the RS-232 interface. The data set firmware converts this data into packets which are passed through the PABX in UI frames. No error detection or correction is performed on this data; synchronous data usually includes its own error detection and correction protocol, and it is more important to maintain the synchronization between the two endpoints. The synchronization is done by a software Phase Locked Loop algorithm, which actually becomes a frequency locking loop.

X.31 - It is a protocol for adapting a data rate from a synchronous interface to a 64-kbps channel, such as is used by MITEL PABXs. This mode will allow Mitel equipment to be connected to another manufacturer's equipment without having protocol conversion in between. The synchronous data that is being received must be HDLC based, with a maximum frame size of 512 bytes.

The DATASET 2100 decodes the received HDLC frames and reformats them into new HDLC frames using the X.31 protocol on the B channel. The rate conversion is done by transmitting HDLC flags during B channel idle time. Received frames are depacketized and transmitted to the attached device in HDLC format at the RS-232 interface synchronous clock rate.

Asynchronous Operation

10.4 The DATASET 2100 Series can operate in asynchronous mode in simplex, half duplex, or full duplex modes. Once the communications are established between data sets in asynchronous mode, the D channel connection is dropped, and only the B channel connection remains. The B channel passes through the PABX, without supervision by the PABX, connected only to the two endpoints.

Back-to-Back Operation

10.5 Two DATASET 2100 Series may be connected back to back to communicate directly with each other (typically to bypass a failed PABX). At least one of the data sets must be in hunt mode; the data set then alternates between master and slave timing until synchronization is achieved. If both data sets are in hunt mode, it can take up to one minute to achieve synchronization.

A back to back connection may be established as above in either synchronous mode or in asynchronous mode between two DATASET 2100 series. If the DATASET 2100 Series is in both master mode and asynchronous mode, the back to back connection can also be made with a DATASET 1100 series. The data set that is operating in master mode scans its DIP switches on power up, and sets its operating parameters according to the settings of the switches.

Timing Modes

10.6 There are four modes of timing available with a DATASET 2100 series that are used according to the current mode of operation.

Internal - The Internal clock is used when the DATASET 2100 Series is operating in Data Communication Equipment (DCE) mode. The transmit clock (SCT) to the Data Terminal Equipment (DTE) is generated within the dataset's baud rate generator and is not synchronized with PABX timing. The dataset's receive clock (SCR) to the DTE is extracted from the data received from the far end dataset by the Phase Locked Loop (PLL) circuit within the dataset.

System - The System clock is used when the DATASET 2100 Series is operating in Data Communication Equipment (DCE) mode. The dataset's transmit clock (SCT) and receive clock (SCR) to the DTE are extracted from the data received from the far end devices by the Phase Locked Loop (PLL) circuit within the dataset.

Transmit External - The DATASET 2100 Series uses the Transmit External clock in either DTE or DCE mode; it is received from an external clock signal from the attached device. From a DTE device, the clock is received on RS-232 pin 24; from a DCE device, the clock is derived from the DCE device's SCR. The dataset's receive clock (SCR) to the DTE is extracted from the data received from the far end device by the Phase Locked Loop (PLL) circuit within the dataset.

Transmit and Receive External - When the DATASET 2100 Series is operating in DTE mode, the Transmit and Receive External clocks are derived from the attached DCE device. The external transmit clock is received from the modem's SCR (pin 17) and transferred to pin 24 by the DCE adapter; the external receive clock is received from the modem's SCT (pin 15) and transferred to pin 18 by the DCE adapter.

Modem Adapter

10.7 DATASET 2100 Series are DCE devices, as are modems. To connect a DATASET 2100 Series to a modem requires that a Modem Adapter (pn 9141-100-100-NA) be connected to the RS-232 connector on the data set. The Modem Adapter makes the DATASET 2100 Series appear as a DTE device to the modem.

DATASET 2100 Switch Settings

10.8 There is an 8-position DIP switch on the back of a Standalone DATASET 2103. It is read once immediately after DNIC Sync is achieved and the operating parameters are set according to the settings, as defined in Table 10-1. When viewed from the rear the DIP switches are labelled as follows:

1 2 3 4 5 6 7 8
User accessible switches

Table 10-1 DATASET 2100 Dip Switch Settings				
SWITCH 1				**DATASET OPERATING MODE**
DOWN				SET - PABX OPERATION
UP				HUNT - BACK TO BACK MODE
	SWITCH 2			**ASYNCHRONOUS/SYNCHRONOUS MODE**
	DOWN			ASYNCHRONOUS
	UP			SYNCHRONOUS
ASYNCHRONOUS MODE				
		SWITCH 3	**SWITCH 4**	**ASYNC FLOW CONTROL**
		DOWN	DOWN	FLOW CONTROL DISABLED
		UP	DOWN	XON/XOFF FLOW CONTROL ENABLED
		DOWN	UP	CTS FLOW CONTROL ENABLED
		UP	UP	FLOW CONTROL DISABLED
SWITCH 5	**SWITCH 6**	**SWITCH 7**	**SWITCH 8**	**ASYNC SPEED**
DOWN	DOWN	DOWN	DOWN	AUTOBAUD
UP	DOWN	DOWN	DOWN	110
DOWN	UP	DOWN	DOWN	150
UP	UP	DOWN	DOWN	200
DOWN	DOWN	UP	DOWN	300
UP	DOWN	UP	DOWN	600
DOWN	UP	UP	DOWN	1200
UP	UP	UP	DOWN	2400
DOWN	DOWN	DOWN	UP	4800
UP	DOWN	DOWN	UP	9600
DOWN	UP	DOWN	UP	19200
SYNCHRONOUS MODE				
		SWITCH 3	**SWITCH 4**	**SYNC OPERATING MODE**
		DOWN	DOWN	INTERNAL CLOCK
		DOWN	UP	SYSTEM CLOCK
		UP	DOWN	TX EXTERNAL CLOCK
		UP	UP	TX AND RX EXTERNAL CLOCK
				Page 1 of 2

Table 10-1 DATASET 2100 Dip Switch Settings (continued)				
SWITCH 5				**SYNCHRONOUS MODE ONLY**
DOWN				TRANSPARENT MODE
UP				X.31 MODE
	SWITCH 6	**SWITCH 7**	**SWITCH 8**	**SYNC SPEED**
	DOWN	DOWN	DOWN	1200
	UP	DOWN	DOWN	2400
	DOWN	UP	DOWN	4800
	UP	UP	DOWN	9600
	X	X	UP	19200 (X = don't care)
				Page 2 of 2

Note: **1.** PARAMETER CHANGES MUST BE FOLLOWED BY A POWER RESET.

2. ALL SWITCH PARAMETERS, EXCEPT SWITCH 1 CAN BE OVERRIDDEN BY THE PABX CDE PARAMETERS.

Standalone DATASET 2103 Description

10.9 A Standalone DATASET 2103 is packaged in a flat case which can be placed under a standard desk telephone set. It is functionally the same as the Rack Mounted DATASET 2102 (they share the same base printed circuit board). The Standalone DATASET 2103 can be connected to the PABX using a four-wire connection; two wires connect the dataset to the Digital Line Card, and two different wires connect the telephone set Tip-Ring pair to an ONS or COV line card; it may also be connected to a Digital Line Card within the PABX by a single twisted pair (the telephone set is connected independently). Figure 10-1 shows typical applications of a Standalone DATASET 2103 connected to a telephone set and to a personal computer or terminal, while Figure 10-2 shows a Standalone DATASET 2103. The DATASET 2103 case is 206 mm wide x 270 mm long x 35 mm high (8.1 in. x 10.6 x 1.4 in.).

Controls, Indicators, and Connectors

The keys and LEDs of a Standalone DATASET 2103 are:

- ATTN key
- DISC key
- POWER LED
- READY RxD LED
- DEVICE TxD LED
- ASYNC

The Standalone DATASET 2103 RJ-11 modular connector pins are:

Line		**Telephone**	
Pin	Signal	Pin	Signal
1	no connection	1	no connection
2	data Tip	2	modem MI
3	voice Tip	3	voice Tip
4	voice Ring	4	voice Ring
5	data Ring	5	modem MIC
6	no connection	6	no connection

Peripheral Devices

Installation

WARNING: **ANY CONNECTION OF THIS SET TO AN OFF PREMISE APPLICATION, AN OUT OF PLANT APPLICATION, OR TO ANY OTHER EXPOSED PLANT APPLICATION MAY RESULT IN A SAFETY HAZARD, AND/OR DEFECTIVE OPERATION, AND/OR EQUIPMENT DAMAGE.**

Chart 10-1 Installation of a Standalone DATASET 2103

Step	Action
1.	Install a Standalone DATASET 2103 by placing it near its associated telephone set and data peripheral (its case is designed to lie flat on the desk under a standard telephone).
2.	Complete Customer Data Entry for the system to allow the Standalone DATASET 2103 to operate with the PABX.
3.	Set the eight DIP switches as required for this installation. Refer to Table 10-1.
4.	Connect the telephone set cord to the jack marked PHONE; connect the modular telephone line cord to the jack marked LINE and to the wall or floor jack which connects to the PABX main distributing frame (MDF). The connection to the MDF is a four-wire circuit; the voice tip and ring (red and green) connect to an ONS or COV line card (depending on the type of set being used), and the Dataset Tip and Ring (yellow and black) connect to a Digital Line Card.
5.	An alternative Tip and Ring wiring arrangement is: connect the telephone set through a two-wire connection via the MDF to its ONS or COV line card; connect the modular line cord (data Tip and Ring) to the jack of the Standalone DATASET 2103 marked LINE, through the MDF to its Digital Line Card in the PABX.
6.	Plug the power supply into an ac outlet and its cord into the power socket on the Standalone DATASET 2103. All LEDs should flash 5 times, indicating that the dataset has passed its self-test. If this self-test does not happen, the dataset is faulty and must be replaced. Then the POWER LED comes on, to indicate that the data set is communicating with the PABX. If the POWER LED flashes, check the wiring.
7.	Plug the RS-232 interconnecting cable into the RS-232 connector at the back of the Standalone DATASET 2103 and connect its other end to the peripheral device. When the RS-232 connector is attached, the DEVICE LED should be on, unless the device is an auto-answer device, and defined as such in CDE (usually only destination devices such as host computers are auto-answer). If the device is not auto-answer, and the DEVICE LED does not come on, make sure the device is powered on, and then use an RS-232 break-out box to verify that the device provides DTR. If the device does not provide DTR, select CDE option "RS-232 Force High" for that device.

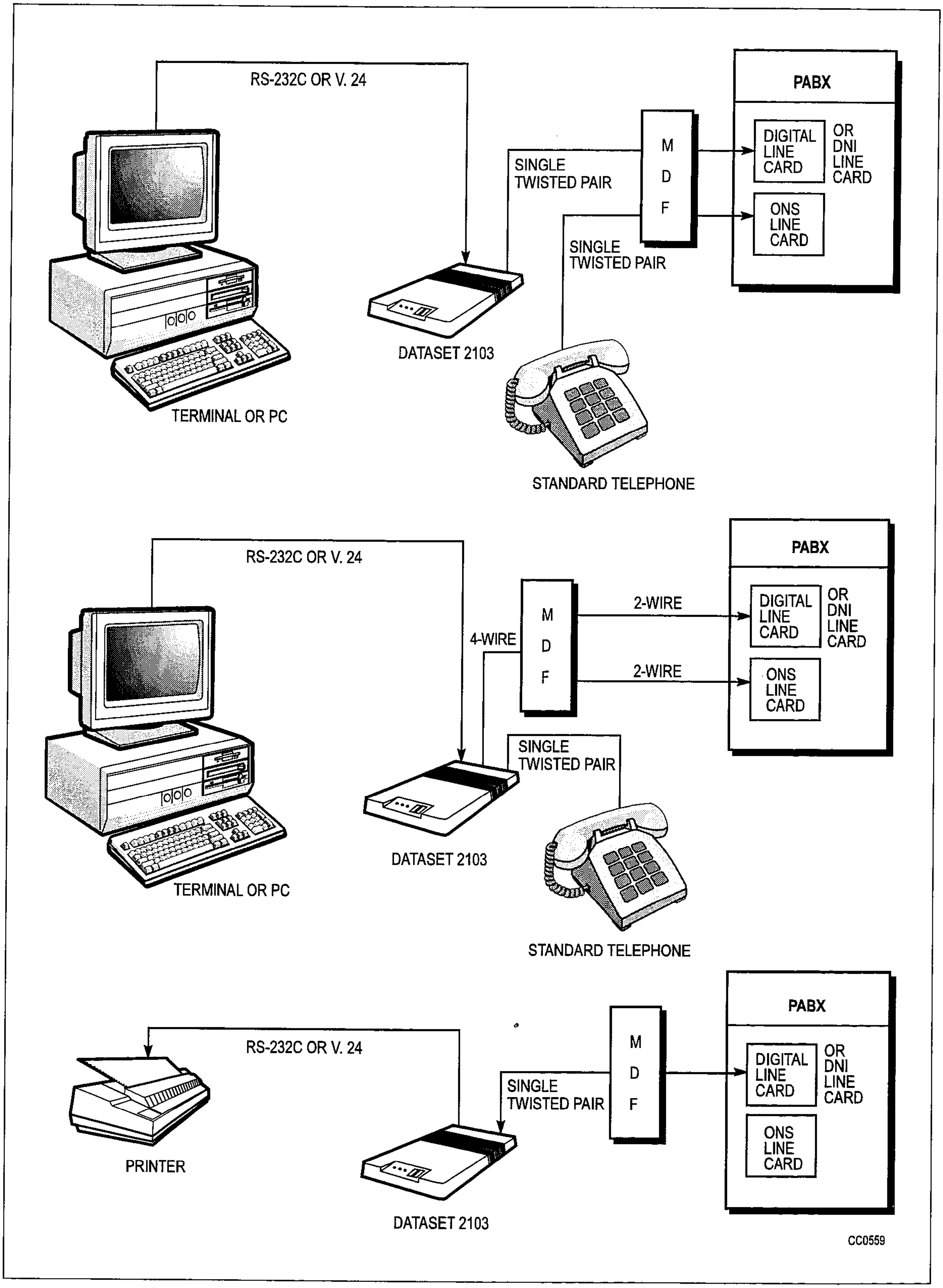


Figure 10-1 Typical Standalone DATASET 2103 Applications

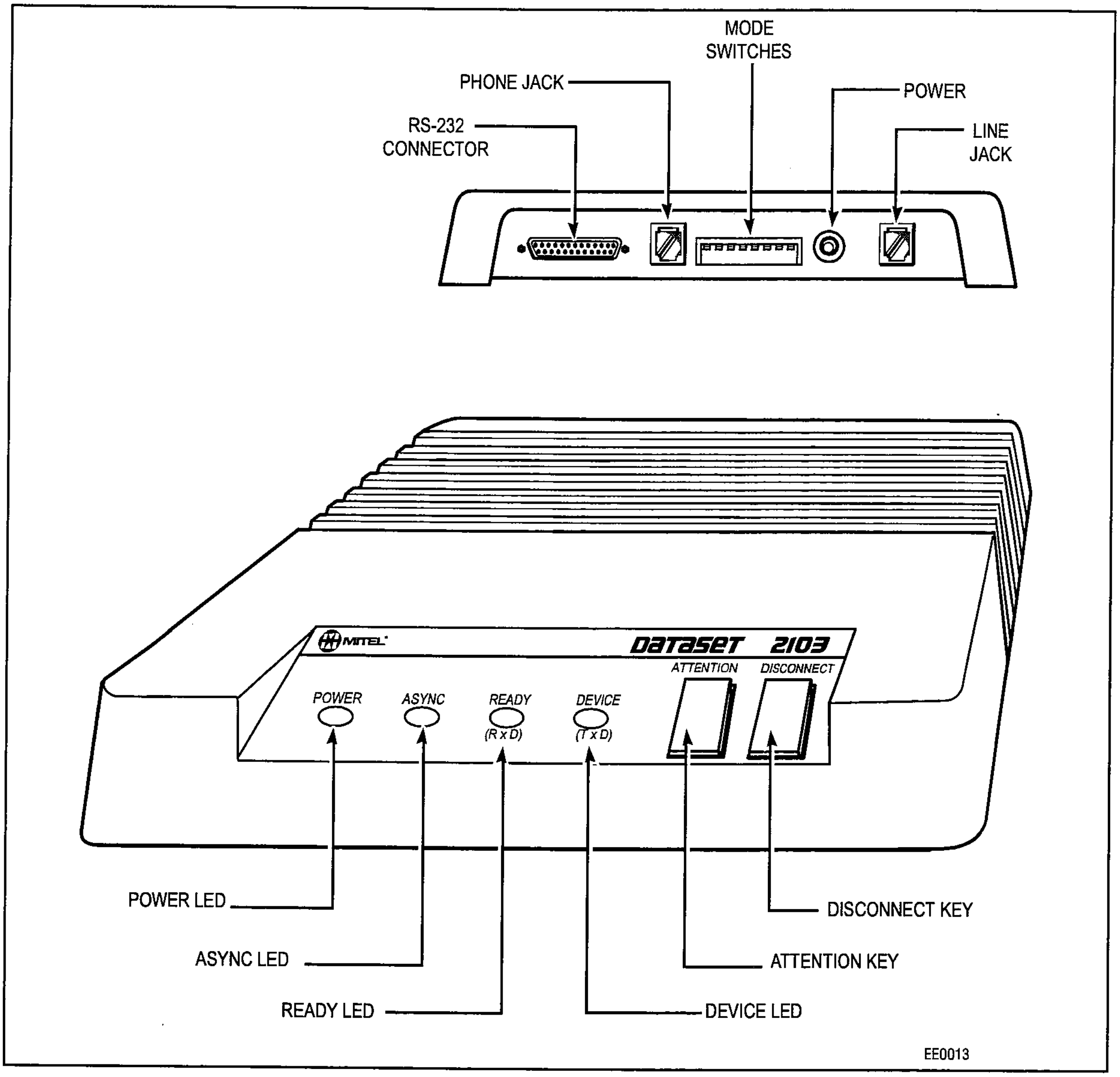


Figure 10-2 Standalone DATASET 2103, Showing Connectors and Indicators

Circuit Description

10.10 The DATASET 2100 Series circuit contains six basic blocks, as shown in Figure 10-3. The microcontroller is the central processor for the dataset; it contains two timers, the BAUD rate generator, the Universal Asynchronous Receiver Transmitter (UART) and the system High-level Data Link Control (HDLC). The switches on the data set interface directly to the microcontroller; this way, the firmware can override the switch settings.

The serial/parallel converter converts serial data from the RS-232 port to parallel for the processor, and parallel data from the processor to serial for the RS-232 port. In asynchronous mode, the microprocessor does the conversion; in synchronous mode, shift registers and latches interface the TxD and RxD lines with the databus to the processor.

The Phase Lock Loop and Clock circuits synchronize the receive data rate of the data set to the rate of the received data from the other device.

The DNIC Interface Circuit provides the interface to the PABX; its clock signals operate the dataset in either set or system mode.

The RS-232 Interface contains the drivers and receivers that buffer the signal lines. It also contains a loopback circuit which can connect together transmit data from the microcontroller to receive data to the microcontroller (RxD to TxD) for self-testing functions.

The Power Supply receives unregulated ac voltage and rectifies it to become +12 vdc unregulated, -12 vdc unregulated, and +5 vdc regulated for logic circuits.

Power Supplies

10.11 The Standalone DATASET 2103 receives its power from a plug-in transformer which supplies 9 vac to a power connector on the back of the set. Circuitry within the data set converts this voltage to the required dc voltages.

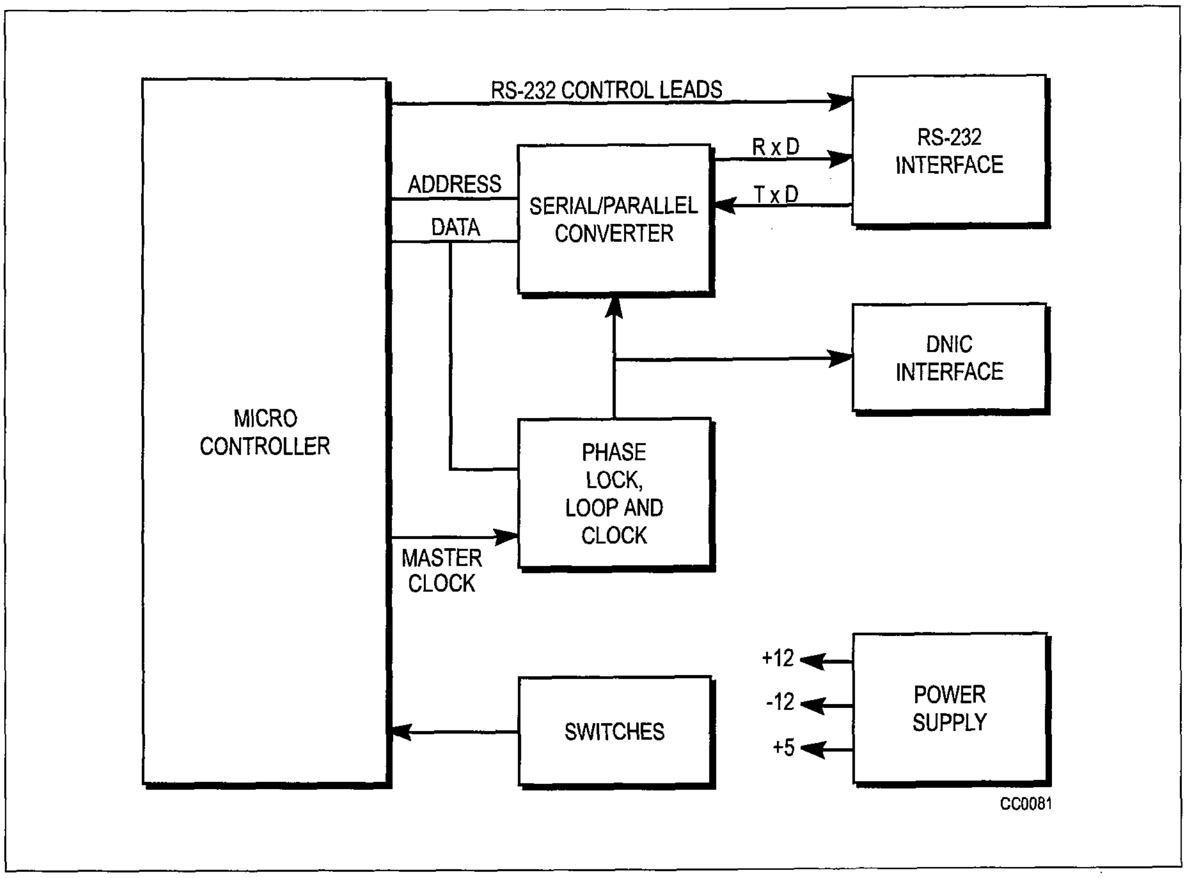


Figure 10-3 Typical DATASET 2100 Series Block Diagram

11 DATASET Hardware Flow Control

Flow Control

11.1 The dataset is designed to conform to the RS-232C specification. However, RS-232C does not define a control lead for DTE flow control of a DCE device. Several printer manufacturers use Pin 20 (DTR) for this purpose; others use Pin 11 and/or other RS-232 leads for flow control purposes. Some manufacturers force a lead high to indicate a flow controlled state; others force a lead low. A standard has not been defined.

To fully support devices which use only hardware flow control, pin 25 may be programmed in CDE to be used by an attached device to flow control the dataset. Pin 25 may also be used as a Ring Indicator (RI) input to the dataset, to signal an incoming call when a modem is attached to the dataset. Pin 25 does not have to be explicitly defined in CDE as an RI input; the data set assumes this use unless the Pin 25/CTS flow control option is selected. This implies that hardware flow control of the dataset and incoming modem calls are mutually exclusive.

The flow control options available through CDE are:

- **XON/XOFF.**
- **CTS** - The dataset flow controls the attached device by dropping CTS; the device cannot flow control the dataset.
- **None.**
- **Pin 25 High/CTS** - The attached device flow controls the dataset by raising Pin 25, the dataset flow controls the device by dropping CTS.
- **Pin 25 Low/CTS** - The attached device flow controls the dataset by dropping Pin 25, the dataset flow controls the device by dropping CTS.

Compatibility

Datasets supporting the new hardware flow control options are compatible with older versions of PABX software; however, pin 25 flow control will not be available.

Older datasets can also be used with new PABX software; however, selecting the pin 25 flow control options in CDE will have no effect.

Printer Connecting Cables

11.2 Cables which support three common methods of hardware flow control are described next. Data Circuit Descriptors are described in the Customer Data Entry (CDE) Practice of the associated PABX.

Connecting Printers That Use RS-232 Cable Pin 4 (RTS) For Flow Control

TO PRINTER		TO DATASET
Chassis Gnd 1	———	1 Chassis Gnd
TxD 2	———	2 TxD
RxD 3	———	3 RxD
RTS 4	———	25 HW Flow Control / RI In
CTS 5	———	5 CTS
DSR 6	———	6 DSR
Signal Gnd 7	———	7 Signal Gnd
DCD 8	———	8 DCD
DTR 20	———	20 DTR

Connecting Printers That Use RS-232 Cable Pin 11 For Flow Control

TO PRINTER		TO DATASET
Chassis Gnd 1	———	1 Chassis Gnd
TxD 2	———	2 TxD
RxD 3	———	3 RxD
RTS 4	———	4 RTS
CTS 5	———	5 CTS
DSR 6	———	6 DSR
Signal Gnd 7	———	7 Signal Gnd
DCD 8	———	8 DCD
11	———	25 HW Flow Control/RI In
DTR 20	———	20 DTR

Connecting Printers That Use RS-232 Cable Pin 20 (DTR) For Flow Control

TO PRINTER		TO DATASET
Chassis Gnd 1	———	1 Chassis Gnd
TxD 2	———	2 TxD
RxD 3	———	3 RxD
RTS 4	———	4 RTS
CTS 5	———	5 CTS
DSR 6	———	6 DSR
Signal Gnd 7	———	7 Signal Gnd
DCD 8	———	8 DCD
DTR 20	———	25 HW Flow Control / RI In

12 Music-on-Hold/Pager Unit

General Description

12.1 The Music-On-Hold/Pager unit interfaces a standard *SX-200* ML DNIC port to the following external equipment:

- External music source for Music-on-Hold
- External paging amplifier (with or without answerback capability)
- Up to two night bells
- An external alarm

The unit is powered by the *SX-200* ML PABX and does not require a separate power source. A single 25 pair amphenol connects to the *SX-200* ML PABX via the main distribution frame. A single LED indicator provides basic status information. The unit can be wall-mounted next to the *SX-200* ML PABX.

Each Music-on-Hold/Pager Unit supports a single paging zone. If more than one paging zone is required, additional Music-on-Hold/Pager Units can be added as required.

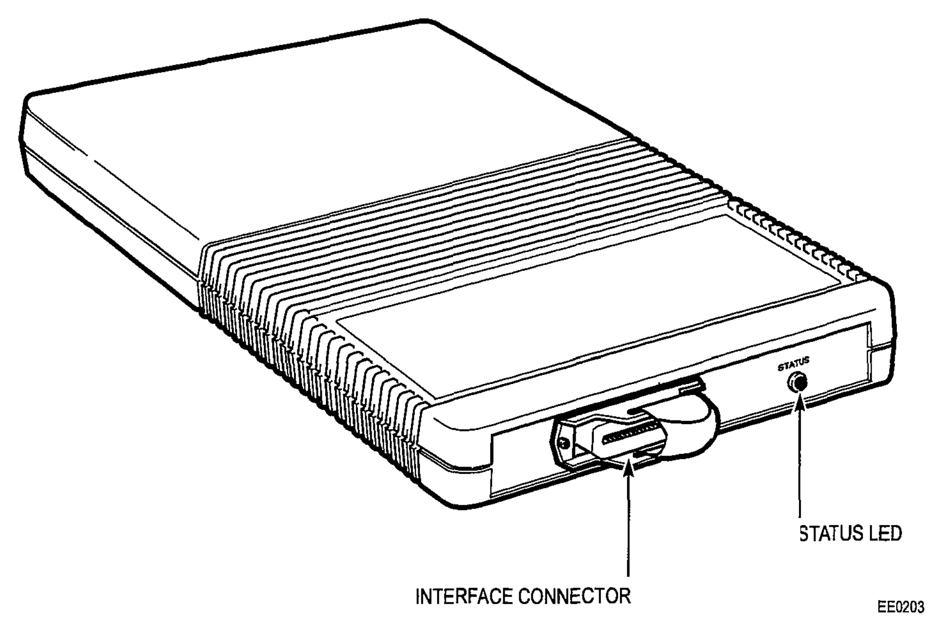


Figure 12-1 Music-On-Hold/Pager Unit

LED Indicator

The LED indicates the following states:

Table 12-1 Music-On-Hold/Pager Unit LED Indicator	
Indication	**Status**
OFF	No power from *SX-200* ML PABX DNIC
Flashing (2 Hz - On 250 ms, Off 250 ms.)	Loss of synchronization with *SX-200* ML PABX, the MOH/Pager Unit may be faulty
On Solid	MOH/Pager Unit is operating
Winking (2 Hz - ON 50 ms, Off 1 sec.)	MOH/Pager Unit is operating and the paging amplifier is being accessed

Interface Technical Specifications

MOH Input

12.2 The MOH input is a transformer coupled input with impedance of 600 ohms. The gain is fixed at -4 dB A/D. To meet signal power limits applied to such interfaces by FCC part 68 and Industry Canada CS03, the audio level at the digital point will be limited to -15 dBm0 and the frequency response is rolled off (3 dB point) at approximately 3 Khz. The amplitude limiting begins to affect the input signal when it exceeds approximately 100 mVrms. Input signals should be in the range of 10 to 100 mVrms.

Paging Input/Output

The paging interface is a transformer coupled loop start trunk with an impedance of 600 ohms. The DC termination is activated by a relay when the paging amplifier is being accessed. The gain is fixed at -4 dB A/D and +3 dB D/A.

When used with an “answer back” paging amplifier, this interface must meet signal power limits applied to such interfaces by FCC part 68 and Industry Canada CS03, the audio level at the digital point will be limited to -15 dBm0 and the frequency response is rolled off (3 dB point) at approximately 3 Khz. The amplitude limiting begins to affect the input signal when it exceeds approximately 100 mVrms, therefore, answerback signals should be in the range of 10 to 100 mVrms.

Relays

Relays are provided for Paging, Night Bells, and Alarms. The contacts are rated at 0.1 A @90 Vac or 0.5 A @48 Vdc.

Paging Control Relay. Both normally open and normally closed contacts are provided to control the external paging amplifier. These relay contacts are switched whenever the paging amplifier is being accessed.

Night Bell Relays. Two independent, normally open relays are provided to control external night bells.

Alarm Relay. One normally open relay is provided to control external devices such as an alarm.

Appendix A

Loop Length Specifications for Connections to a Digital Line Card

A.1 The following rules for loop lengths between the DNI Line Card within the PABX and *SUPERSET 400* series telephone sets, DATASETS, or *SUPERCONSOLE 1000* Attendant Console must be followed for proper operation of the device:

Maximum loop length (twisted pair) 24 or 26 AWG	see Table A-1 or Table A-2
Maximum length of quad cable (22 AWG)	50 m (160 ft)
Modular Line Cord	3 m (10 ft)

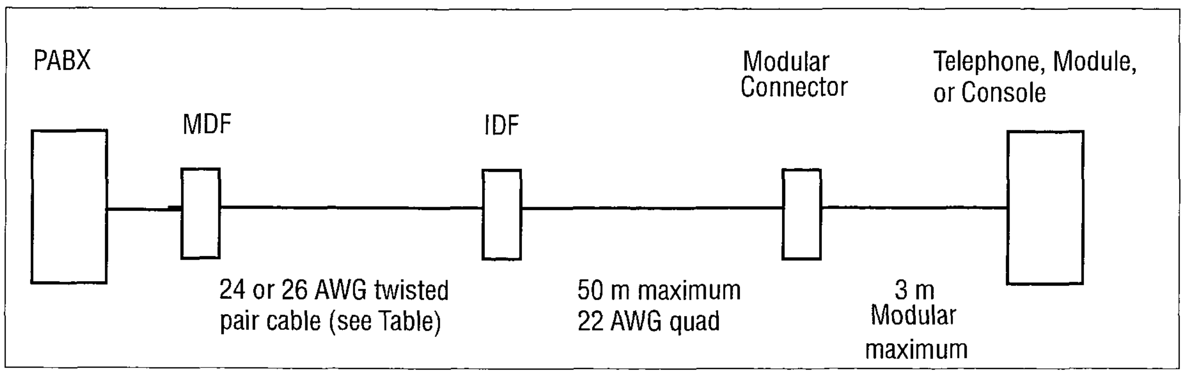


Table A-1 Maximum Loop Lengths for Peripheral Devices

Peripheral Device	Maximum Loop Length	
	Without Bridge Tap	With Bridge Tap
SUPERSET 401+	1000 m	1000 m
SUPERSET 410	1000 m	1000 m
SUPERSET 420	1000 m	1000 m
SUPERSET 430	1000 m	1000 m
DATASET 1103	2000 m	1000 m
DATASET 2103	2000 m	1000 m
SUPERCONSOLE 1000 console	1000 m	1000 m

Table A-2 Loop Lengths for *SUPERSET* Telephones		
Card Type	**Wire Gauge (AWG)**	**Loop Length**
Digital Line Card	24	1000 m (3300 ft)
	26	1000 m (3300 ft)

NOTES

NOTES

9109-098-180-NA Issue 1
March 1997

SX-200® ML PABX

Engineering Information

MITEL®

NOTICE

IMPORTANT SAFETY INSTRUCTIONS

These instructions are intended to be used as a general guide to provide basic installation information which is necessary for the proper and safe functioning of this equipment.

WARNING: Failure to follow all instructions may result in improper equipment operation and/or the risk of electric shock.

General

- Read and understand all instructions. Keep these instructions with the equipment.
- Do not attempt to install or service this equipment unless you are skilled in the installation and maintenance of electronic telecommunication equipment and have successfully completed specific training for this equipment.
- This product must be installed and serviced in accordance with this document and the information contained in this set of technical practices. Practice Index documents are 9109-098-501-NA, 9109-098-502-NA, and 9109-098-503-NA.
- Follow all the steps outlined in this document in the sequence that is given.
- Configure this product only with the assemblies specified and in the locations stated in this document and in this set of technical practices.
- Replace all guards or barriers. Close and lock doors at the completion of installation or before returning the equipment to service.
- Grounding circuit continuity is vital for safe operation of telecommunication equipment. Never operate telecommunication equipment with the grounding conductor disconnected. Ensure that the grounding conductor is installed before connecting telecommunication cabling to any system. (See Note, below).

Note: All cabinets must be unplugged from the ac mains during servicing. Unplugging an *SX-200* ML cabinet means that the cabinet is "floating", and thus presenting a potential static problem. To reduce static susceptibility on an *SX-200* ML cabinet, always attach the wrist strap from the cabinet being serviced, and immediately place any item removed from a node into an antistatic bag.

Installation of Telecommunication Wiring

Telecommunication wiring to this product shall conform to all applicable safety and electrical wiring regulations. Installation of telecommunication wiring shall be performed by following precautions in accordance with standard industry practice. The precautions to be followed include:

- Never install telephone wiring during a lightning storm.
- Never install telephone jacks in wet locations unless the jack is specifically designed for wet locations.
- Never touch uninsulated telephone wires or terminals unless the telephone line has been disconnected at the network interface.
- Use caution when installing or modifying telephone lines.

Use of Notices and Symbols

The following information provides an explanation of the notices and symbols which appear on the product and in the practices for this product.

DANGER	Danger indicates an imminently hazardous situation which, if not avoided, will result in death or serious injury.
WARNING	Warning indicates a potentially hazardous situation which, if not avoided, could result in death or serious injury.
CAUTION	Caution indicates a potentially hazardous situation which, if not avoided, may result in minor or moderate injury and/or damage to the equipment or property

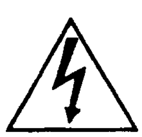
 **DANGEROUS VOLTAGE**	The lightning flash with arrowhead symbol, within an equilateral triangle, is intended to alert the user to the presence of uninsulated "dangerous voltage" within the product's enclosure that may be of sufficient magnitude to constitute a significant risk of electric shock to persons.
 **INSTRUCTIONS**	The exclamation point within an equilateral triangle is intended to alert the user to the presence of important operating and maintenance (servicing) instructions in the literature accompanying the product.
 **PROTECTIVE GROUNDING TERMINAL**	The ground symbol within a circle identifies the terminal which is intended for connection to an external protective conductor. This connector must be connected to earth ground prior to making any other connections to the equipment.

Table of Contents

List of Figures

List of Tables

1 Introduction

General

1.1 This Practice provides basic engineering information for the SX-200® ML PABX. It describes:

- physical aspects of the system
- onfigurations
- technical characteristics.

Reason for Issue

1.2 This section is issued to provide basic engineering information.

Disclaimer

The following products have been manufacture-discontinued by Mitel. These products are supported but not described in *SX-200* ML Practices:

- SUPERSET 3™ and SUPERSET 4® telephone sets
- SUPERSET 3DN™ and SUPERSET 4DN™ telephone sets
- DATASET 1101 data cartridge
- SUPERSET™ DSS module.

The following products and peripheral devices are not supported on the *SX-200* ML PABX and are not described in *SX-200* ML Practices:

- Modem Interconnect Panel
- DATASET 1102 Rack-mounted Dataset
- DATASET 2102 Rack-mounted Dataset
- DATACABINET 9000 data cabinet
- DATASHELF 9100 datashelf
- ISDN Node
- Fiber Interface Module (and associated products)
- Peripheral Node
- LCD Console (and Console module for Universal Card).

2 Features

General

2.1 The *SX-200* PABX offers many features which are provided by a software package. A full description of these features appears in the *Features Description* Practice. Certain limitations which apply to System Features are listed in Table 2-1.

Table 2-1 Feature Limitations

Feature	Limitations
Maximum number of simultaneous calls	48
Maximum number of speech paths or channels used by any call	2
Max. number of simultaneous consultations	5
Maximum number of simultaneous add-on (3-way) calls	3
Maximum number of simultaneous station-controlled conference calls	3
Maximum number of parties in conference at one time	5
Maximum number of calls that can simultaneously be camped on to a station, trunk group, or hunt group	200
Maximum number of simultaneous callbacks that can be enabled	200
Maximum number of simultaneous call forwards that can be enabled	400
Maximum number of simultaneous "Dial 0" calls	48
Max. number of messages queued in the system	250
Maximum number of hunt groups	100
Maximum number of hunt groups in ACD	99
Maximum number of calls that can be simultaneously connected to Music-on-Hold	48
Maximun number of stations in a station hunt group	50
Maximum number of stations in a call pickup group	50
Maximum number of dial call pickup groups	50
Maximum number of trunks assignable to night stations	200
Maximum number of trunks in a trunk group	50
Maximum number of trunk groups	50
Maximum number of calls that can override a given extension	1
Maximum number of attendant consoles	11
Maximum number of calls that can be simultaneously held by one attendant	8
	Page 1 of 2

Table 2-1 Feature Limitations (continued)

Feature	Limitations
Maximum number of incoming calls that can be separately identified at the attendant console	8
Maximum number of LDNs that can be identified at the attendant console	9
Maximum number of LDNs	100
Maximum Number of Night Bells	25
Maximum number of calls waiting that can be displayed at console	99
Maximum number of calls that can be waiting at console	200
Maximum number of abbreviated dial numbers	1000
Maximum number *SUPERSET* Speed Dial numbers	2212
Maximum number of trunk buffers for SMDR	200
Maximum number of DATA SMDR buffers	128
Maximum number of SUPERSET 401+™, SUPERSET 410™, SUPERSET 420™, and SUPERSET 430™ telephones and PKMs.	96
Maximum number of lines	96
Maximum number of T1 Trunks per system	2
Maximum Number of Page Groups	50
Maximum Number of Stations in a Page Group	16
Maximum Number of Sub-attendants	25
Maximum Number of LDN Appearances	16
Maximum Number of Line Appearances	16
	Page 2 of 2

3 System Overview

General

3.1 *SX-200* ML LIGHTWARE™ 16 software is available in a base load plus optional features. The password on the System ID module enables optional features. Software is not operational without the System ID module. *SX-200* ML *LIGHTWARE* 16 features include:

- ANI and DNIS
- Hotel/Motel
- Property Management System
- Automated Attendant
- ONS Voice Mail
- Printing
- Automatic Call Distribution (ACD)
- Key Sets - Off-Hook Voice Announce.

Maintenance

3.2 Modular design and functional packaging of the equipment permits rapid location and replacement of defective components. Circuit malfunctions are detected by diagnostic routines automatically initiated by the Main Control Card II (MCC II). Diagnostic routines, detailed in the *General Maintenance Information* Practice, and the *Troubleshooting* Practice, direct service personnel to the defective circuit card or assembly, and identify the required field-replaceable unit. Diagnostic routines and maintenance procedures do not interfere with users unaffected by the malfunction.

4 Technical Description

General

4.1 The *SX-200* ML PABX uses Pulse Code Modulation (PCM) time-division multiplexing as its digital conversion format. The PABX employs both digital peripheral cards and digitally-interfaced analog peripheral cards. System architecture is illustrated in Figure 4-1. The major component blocks are described in the paragraphs below.

Circuit Switch Links

4.2 The subsystems of the PABX communicate over 2,048 kHz serial links. Each link is a continuously repeated data frame comprising 32 channels; each channel contains an 8-bit word which occupies an equal time slot within the frame. Certain channels on some links are segregated into message channels; the remainder are used for circuit switch applications.

Circuit switch links provide paths through the system for the transfer of peripheral-related data; i.e., PCM audio, or TDM data. Each digital peripheral interface card has dedicated to it one-and-one-half circuit switch links which connect the card to the peripheral switch.

Control

4.3 The main processor, which has overall control of the system, is a 32-bit MC68020 CPU. It is supported by 4.0 megabytes of Dynamic Random Access Memory which is used for the storage of system software, programmed devices, abbreviated dial digit strings, ARS digit strings, and *SUPERSET* line appearances.

The Bay Control Card has 256 kilobytes of RAM and interfaces its bay to the MCC II. It is controlled by the MC68008 microprocessor.

The SUPERCONSOLE 1000™ Attendant Console interfaces to a Digital Line card by Tip and Ring. It contains a HD6303 CPU, supported by 16 kilobytes of EPROM, and 8 kilobytes of static RAM. A printer can be attached to the console by an RS-232C port. The connector is configured so that the console is data communication equipment (DCE).

Digital Switching

4.4 A custom analog/digital combined integrated circuit is used to implement the analog-to-digital and digital-to-analog conversion functions. The basis of this encode/decode process is the MT8960 Integrated PCM Filter/Codec (commonly known as a CODEC), which is used throughout the system to convert analog to PCM and PCM to analog. The CODEC combines a low-pass filter and an analog-to-digital PCM encoder in the transmit direction (towards the PABX) and a digital-to-analog decoder and low pass filter in the receive direction (towards the peripheral).

The PABX uses another custom VLSI circuit as its main PCM switching matrix element - the 8 link x 32 channel MT8980 Digital Time/Space Crosspoint Switch (DX Chip). The DX device is arranged with eight incoming links and eight outgoing links; each link comprising 32 channels. Under control of the main CPU, any channel of the incoming links can be connected to any channel of the outgoing links. Thus, one DX chip is equivalent to a 65,536 (256 x 256) crosspoint array.

The peripheral control requires three links, and several links are required for the HDLC messaging and DSP resources.

Main Control Card II

4.5 The Main Control Card II (MCC II) is the highest level in the intelligence hierarchy of the PABX. It contains the following components:

- 68020 CPU
- Flash Memory module containing system power-up routines
- Dynamic RAM
- Two communication Ports (UARTs)
- Digital Signal Processor for tone generation/detection and conferencing
- DX module containing the Circuit Switch Matrix, and the Peripheral Switch Message interface (HDLC).

Circuit Switch

The circuit switch matrix provides a nonblocking switching matrix which interconnects the digital peripherals (i.e., digital lines and digital trunks). As a nonblocking matrix, the circuit switch matrix is fully switchable; i.e., all incoming links have access to all outgoing links.

In the *SX-200* ML PABX there are three links from the circuit switch matrix to each Bay Control Card.

Table 4-1 PCM Circuit Switch Link Assignments

Link #	PCM Link Assignment
0	Bay 1 voice link and ringing
1	Bay 1 voice link and messaging
2	Bay 1 voice link and miscellaneous tone
3	not assigned
4	not assigned
5	not assigned
6	HDLC message link
7	Digital Signal Processor link

Message Subsystem

4.6 The message subsystem facilitates the transfer of control messages and program loading between the main Peripheral Control Processor and lower node processors such as Bay Control Cards, *SUPERSET* telephones, and consoles. This transfer of information occurs over the PCM links of the Circuit Switch Matrix. The protocol used in the message system is based on the OSI (Open Systems Interconnection) widely used HDLC (High Level Data Link Control) protocol format.

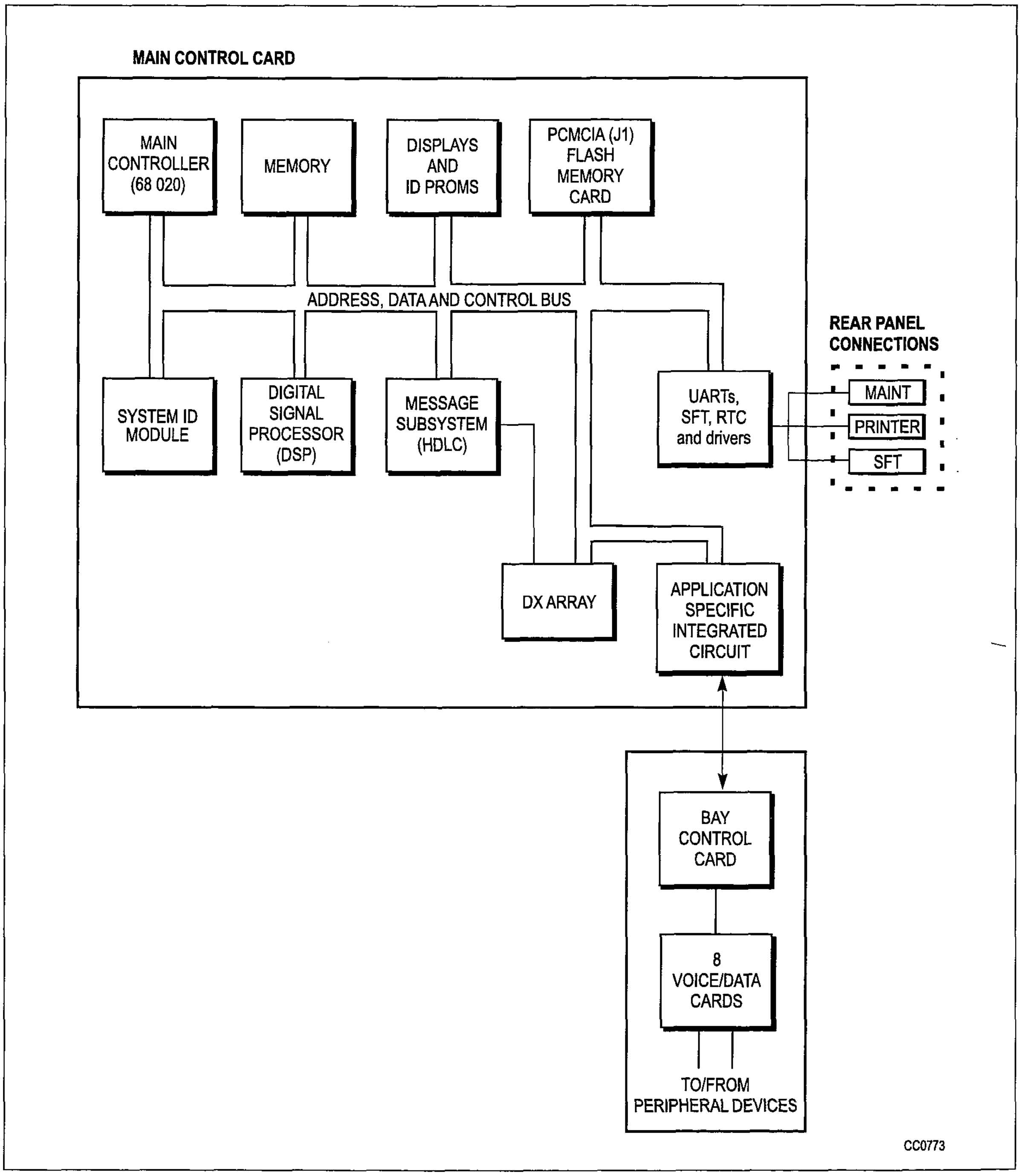


Figure 4-1 ***SX-200*** **ML -** ***LIGHTWARE*** **16 ML System Architecture**

Engineering Information

Peripheral Interface Cards

4.7 Peripheral interface cards provide an interface between the terminations of the PABX (lines and trunks) and the circuit switch. The card type depends upon the type of peripheral interfaced to the system; i.e., an E&M Trunk card caters to E&M trunks only.

The number of interface circuits that each card provides varies with the type, complexity, and space requirements of the circuit. A description of the peripheral interface cards is provided in the following chapter.

Operation of the digital peripheral card is by the MCC II through the Bay Control Card.

The clocks provided by the MCC II for the digital peripheral cards are as follows:

Frame Pulse. This signal synchronizes the start of a PCM frame which consists of a complete sequence of 32 channels. It occurs every 125 microseconds (8 kHz).

244 ns Clock (C244+). The positive going edge of each clock pulse is used to generate channel counting.

5 Circuit Card Descriptions

General

5.1 This chapter describes the cards used in the *SX-200* ML PABX.

Main Control Card II

Description

5.2 This section describes the Main Control Card II (MCC II). There are two variants of the MCC II, one with a Stratum 4 clock (part number 9109-070-000-NA), the other with a Stratum 3 clock (part number 9109-070-001-NA).

The MCC II and the Switch Matrix perform all call processing for the entire system.

CAUTION: **Do not insert or remove the Main Control Card II (MCC II) with the power on. Damage to onboard circuits may result.**

Facilities

Facilities on the Main Control Card II are:

- HDLC link for high speed message communication and bulk data transfers to peripheral processors or Bay Control Cards
- Two UARTs (one printer port, one maintenance port)
- Real Time Clock for time of day and interrupt generation, with 3-hour non-volatile backup
- System PCM Clock Generation
- Memory: Dynamic RAM, 3-day non-volatile CMOS RAM
- PCMCIA card carrier (for flash memory card for software loading)
- Hex and LED displays for diagnostics
- System identification for software security
- Digital signal processor to generate and detect progress and ringing tones and to provide conferencing
- Switch Matrix (digital time/space crosspoint switch) for circuit and message switching
- Fault detection hardware to monitor PCM clock and processor sanity (watchdog).

Engineering Information

Operation

The correct System ID Module must be installed or the software will not run.

At power-up the Main Control Card II is held in reset for about 1 second after the power supply voltages reach their normal levels. After reset, the processor first runs the test software contained in its flash memory. This test software consists of tests for the RAM memory, the CPU and the DMA controller. A routine to load the system software into RAM from flash memory is run only after all preliminary tests have been passed.

RAM

Brief Description

The RAM carries the dynamic memory required for program storage and call processing work areas.

Switch Matrix

Brief Description

The Switch Matrix is integrated into the Main Control Card II.

Facilities

The Switch Matrix includes a DX (digital crosspoint) switch.

Electrical Description

The DX array is a non-blocking array that provides bi-directional links with balanced drivers and receivers.

T1 Clock

Brief Description

The T1 Clock is a digitally controlled oscillator which synchronizes the system clock to an incoming T1 frame rate clock signal and is only required when a T1 card is present in the system. There are two MCC II variants: one with a Stratum 3 clock and one with a Stratum 4 clock.

Electrical Description

Every 100 ms the T1 Trunk Card sends a counter value representing the phase difference between the incoming T1 clock and the *SX-200* PABX system clock. On the MCC II a digital to analog converter generates a corresponding voltage to control the clock oscillator. This forms a phase-locked loop which keeps the clocks in phase.

System ID Module

Brief Description

The System ID module plugs into the connector side of the MCC II. It must be present for the PABX to load its software after power-up.

Operational Description

At power-up, the PABX verifies the presence of the System ID module before allowing software to load. The System ID module and the Mitel options password enable the purchased options for the PABX. If the System ID module is missing or the Mitel options password is incorrect, the PABX does not operate.

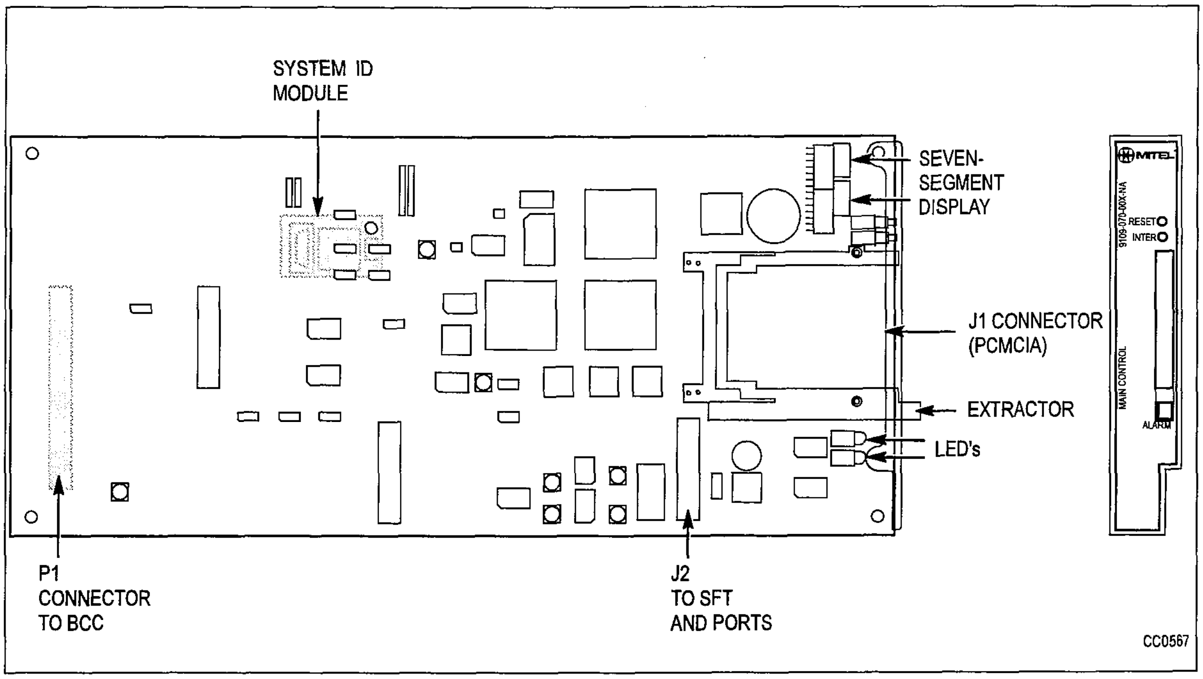


Figure 5-1 Main Control Card II

Bay Control Card

Brief Description

5.3 The Bay Control Card (BCC), part number 9109-017-000, is required in the lower rightmost slot.

CAUTION: This card must not be inserted or removed with the power on.

Functions

The Bay Control Card provides:

- Control of operations within the cabinet
- Monitoring of lines, trunks and other circuits within the bay; reports are sent to the Main Control Card II via HDLC message links
- Ringing signal conversion. (The wave form comes from the Main Control Card II as a PCM signal.)
- Connector for the Main Control Card II.

Indicators

The Bay Control Card has Alarm LED, Tx (transmit), and Rx (receive) indicators for the HDLC message link.

Electrical Description

Connection to the Main Control Card II is via one 96 DIN connector on the Bay Control Card.

There are two pairs of switches on the card; see Figure 5-2. All four switches must be set to closed for normal operation.

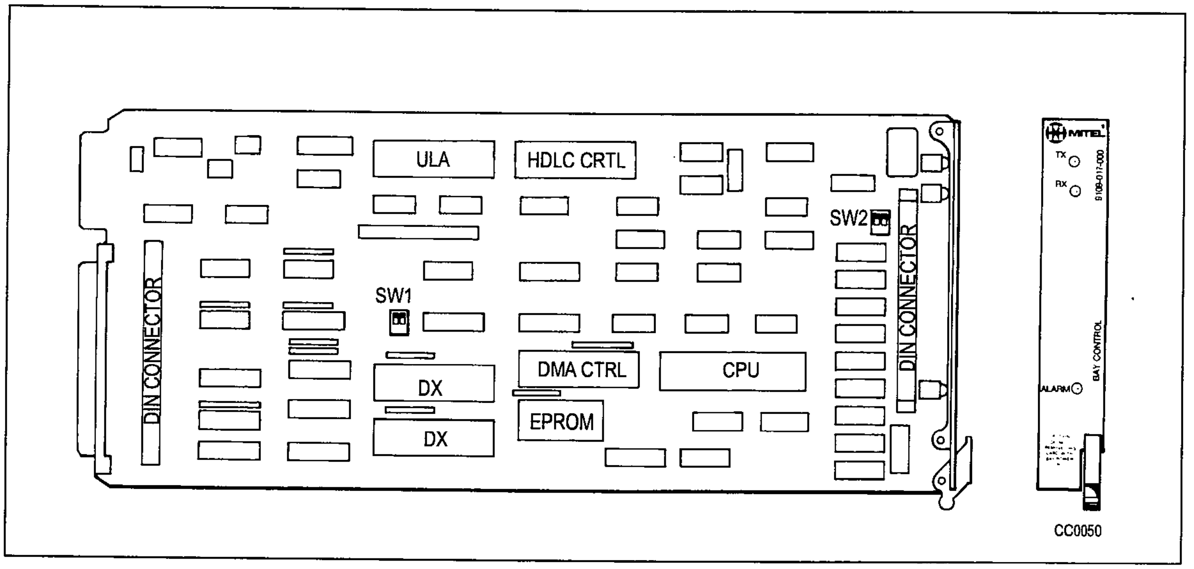


Figure 5-2 Bay Control Card

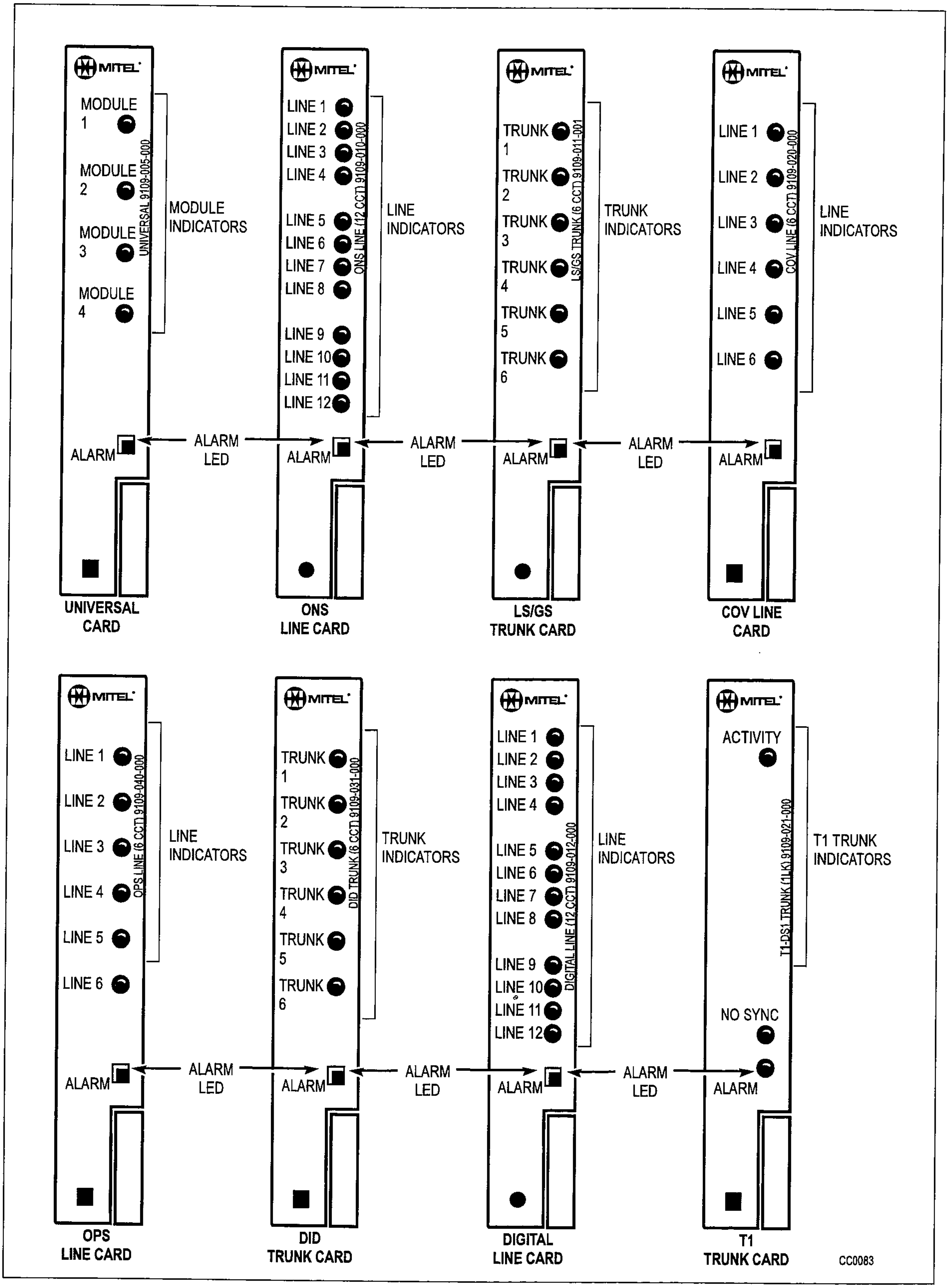


Figure 5-3 Typical Digital Circuit Card Face Plates

Universal Card

Brief Description

5.4 The Universal Card, part number 9109-005-000, interfaces up to four modules to the backplane. The card modules are illustrated in Figure 5-4.

Facilities

Facilities provided by the Universal Card include:

- module mounting positions (four)
- module activity LEDs (four)
- software-controlled failure alarm LED.

Physical Description

A module measures 145 mm x 83 mm (5.7 in. x 3.25 in.). Each module has two vertical 32-pin female DIN connectors which mate to male connectors on the Universal Card. Mechanical connection to the Universal Card is assisted by a standoff.

Up to four modules may be mounted on a Universal Card, subject to electrical power limitations. See the Electrical Description paragraph.

Electrical Description

The combination of modules on a Universal Card is limited by the power available from the card. Each module has a power rating number. The total of these numbers must not exceed 10. The Universal Card can be mounted only in a high-power (upper) card slot of a digital bay. The Module power ratings are as follows:

Module Type	Power Rating
Empty module position	0
Music-on-Hold/Paging	1
DTMF Receiver/Relay	2
E&M Trunk	3

Each of the four module positions is assigned Universal Card tip and ring connections as shown below.

Module Position	Module Tip/Ring		
1	T1/R1	T2/R2	T3/R3
2	T4/R4	T5/R5	T6/R6
3	T7/R7	T8/R8	T9/R9
4	T10/R10	T11/R11	T12/R12

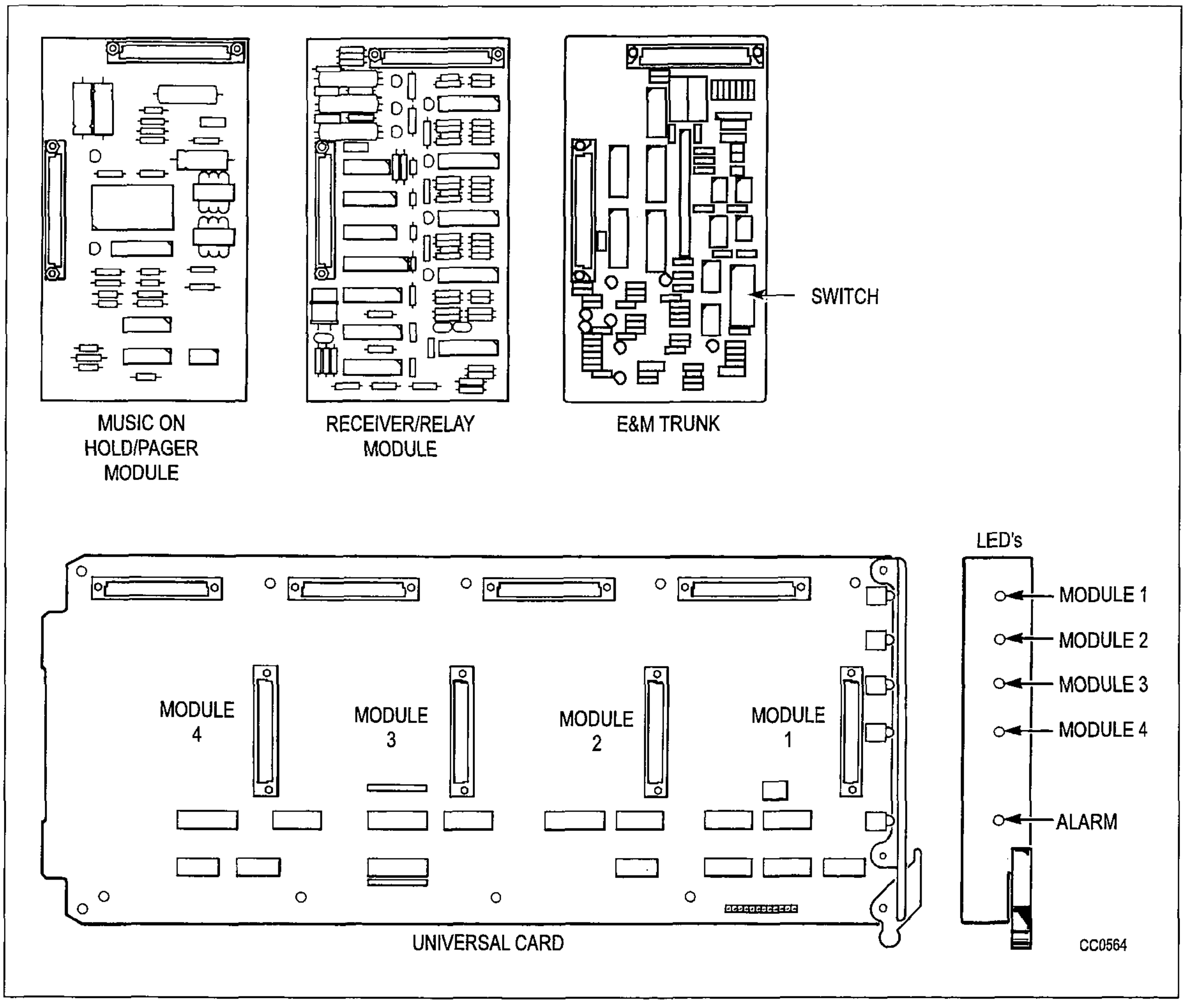


Figure 5-4 Universal Card and Modules

DTMF Receiver/Relay Module

Brief Description

The DTMF Receiver/Relay Module facilitates the reception and decoding of DTMF dialing.

Main Components

Major components of the DTMF Receiver/Relay Module are:

- Mitel filter/codec (four)
- Mitel 8870 DTMF receiver (four)
- Parallel bus interface
- Two general purpose relays.

Facilities

Facilities provided by the DTMF Receiver/Relay Module include:

- Early line split

- Guard time circuit
- Presentation of digits on parallel bus with Data Valid signal.

Circuit Description

There are four receivers on the DTMF module. Each receiver takes its input from the incoming serial PCM audio stream and repeats this data to the outgoing serial PCM stream approximately 125 msec later. A filter/codec converts the data to analog audio which is monitored by a DTMF Receiver chip. When DTMF tones are detected, the loopback of the data to the PCM output stream is disabled (Early Line Split). The DTMF Receiver/Relay Module has a power rating of 2. Two relays are on the module; each is software controlled to provide a contact closure across one tip-ring pair.

Relays

There are two general purpose relays. When each relay closes, it connects a tip and ring pair together.

The relay contacts are rated as follows:

maximum switching voltage: 90 V
maximum carrying current: 0.5 A

Note: This relay contact may be connected only to a secondary circuit that has no direct connection to a primary circuit, and receives its power from a transformer, converter, or equivalent isolation device situated within the equipment.

E&M Trunk Module

Brief Description

The E&M Trunk Module (part number 9109-013-000) plugs into the Universal Card. It provides interface to Type 1 or Type 5 E&M trunks. The module has a power rating of 3.

Facilities

Facilities provided by the E&M Trunk Module include:

- Type 1 and Type 5 signaling
- selectable gain/loss plan for normal or satellite working trunks
- selectable 600 ohm or AT&T Complex Balance Network (350 ohms + 1000 ohms in parallel with 0.21 mF)
- selectable 2- or 4-wire transmission
- on board filter/codec for analog/digital and digital/analog conversions (µlaw).

Operation

The E&M Trunk Module is set for the type of trunk in use by a set of eight DIL switches. The settings are as follows:

Function		Switches 1	2	3	4	5	6	7	8
PABX to Line Gain	3 dB	0	x	x	x	x	x	x	x
	-13 dB	1	x	x	x	x	x	x	x
Line to PABX Gain-	4 dB	x	0	x	x	x	x	x	x
	-11 dB	x	1	x	x	x	x	x	x
Balance	600 ohm	x	x	1	0	x	x	x	x
	Complex	x	x	0	1	x	x	x	x
Transmission	2-wire	x	x	x	x	1	x	x	x
	4-wire	x	x	x	x	0	x	x	x
Signaling	Type 1	x	x	x	x	x	1	x	x
	Type 5	x	x	x	x	x	0	x	x

0= open, 1= closed, x= not applicable
Default setting for North America is 00101100

The E&M Trunk Module applies signals to the M lead and monitors the E lead. In the on-hook condition, the Type 1 interface grounds the M lead; an open presented to the E lead indicates idle, a grounded E lead indicates an incoming call. In the off-hook condition, the Type 1 interface applies -48 volts to the M lead; a ground sent to the E lead indicates an incoming seizure.

Music-on-Hold/Paging Module

Brief Description

The Music-on-Hold/Paging Module (part number 9109-018-000) provides an input for music-on-hold, a paging output, and a relay to switch an external paging amplifier. The module plugs into the Universal Card. The Music-on-Hold/Paging Module has a power rating of 1.

Major Components

Major components of the Music-on-Hold/Paging Module include:

- audio filter/amplitude limiter
- Mitel 8961 filter/codec
- paging driver amplifier
- paging control relay.

Electrical Description

The music input is isolated by a transformer and has an impedance of 600 ohms. The signal should be between 50 and 500 mVrms. High frequencies are attenuated and amplitude limiting is applied as required by FCC rules part 68. Amplitude limiting is applied when the signal exceeds approximately 390 mVrms.

The paging output is isolated by a transformer and has an impedance of less than 200 ohms. The output level into a 600 ohm load is typically -6 dBm (388 mVrms).

The control relay contacts are rated as follows:

maximum switching voltage	90 Vrms
maximum carrying current	0.4 Arms

Note: This relay contact may be connected only to a secondary circuit that has no direct connection to a primary circuit, and receives its power from a transformer, converter, or equivalent isolation device situated within the equipment.

DID Trunk Card

Brief Description

5.5 The DID Trunk Card (part number 9109-031-000) contains six 1-way direct inward dial circuits which provide for direct access to PABX subscriber lines from the public telephone network.

DID Trunk Cards can be used in any digital high-power (upper) slot. The maximum number of these cards is four per system, providing a maximum of 24 ports.

The card measures 157.5 mm x 366.4 mm (6.2 in. x 14.4 in.).

Major Components

Major components in the DID Trunk Card are:

- Mitel 8962 Filter/Codec (one per trunk circuit)
- Feed Reversal relay (one per trunk circuit)
- Alarm LED.

Facilities

The facilities provided by each trunk circuit are:

- Trunk activity LED
- Line protection
- 2-wire / 4-wire conversion (external to internal)
- Analog-to-Digital / Digital-to-Analog conversion (μ–law)
- Immediate, Delay Dial or Wink Start supervision
- Direct Inward Dialing access to PABX subscriber lines
- Conformity with the EIA loss level plan for μ–law compatible PABXs in North America.

Operation

A trunk is idle if the resistance across Tip and Ring is 4000 ohms or more. In idle condition the PABX provides forward battery feed to the line. The Tip is grounded and the Ring is at -48 Volts.

The CO initiates a call to the PABX by terminating Tip and Ring. The supervision circuitry detects the flow of loop current and alerts the system software. The PABX signals that it is ready to receive dialing by briefly applying a battery reversal to the line. Ring is grounded and Tip is at -48 Volts. There are two types of controlled address signaling: Delay Dial and Wink Start.

A Delay Dial signal must start no later than 150 ms after trunk seizure. It is held until the PABX is ready to receive dialing. Minimum hold time is 140 ms.

A Wink Start signal must start at least 100 ms after trunk seizure. It is sent when the PABX is ready to receive dialing and can be held a maximum of 290 ms.

Where the CO does not provide controlled address signaling, the PABX must be prepared to receive dialing 65 ms after trunk seizure.

When the called station or PABX attendant answers, the PABX places battery reversal on the line for the duration of the call. The trunk then returns to the idle state.

A trunk may be busied out by the system software, which then presents an open circuit to the Tip and Ring of both the trunk and trunk card circuit. The trunks default to the busy-out state if system power fails.

Each circuit has a LED on the front panel which lights to indicate the trunk is in use. A seventh LED at the bottom of the panel lights to indicate a failure on the card.

Electrical Description

Line protection comprises high voltage varistors to energy dump ground from Tip and Ring and fusible links incorporated in the battery feed resistors. EMI is controlled by inductors in series with Tip and Ring.

The maximum loop resistance is 1800 ohms. The maximum loop length is 5850 m (19,200 ft) when using 26 AWG wire, 15,240 m (50,000 ft) when using 22 AWG wire.

The card circuitry performs 2-wire to 4-wire conversion, splitting the signal on the trunk into outgoing and incoming speech paths. The analog signal coming from the trunk is converted to Pulse Code Modulation (PCM); the signal to be sent to the trunk is converted from PCM to analog audio. These conversions are performed by a Mitel Codec chip.

Battery feed reversal and busy-out for each trunk are controlled by relays, as shown below.

Condition	Relay 1	Relay 2
Forward Feed (Idle)	ON	OFF
Reverse Feed (Talk)	ON	ON

LS/GS Trunk Card

Brief Description

5.6 The Loop Start/Ground Start Trunk Card, part number 9109-011-001, interfaces six trunk circuits to the system. The card is 158 mm high x 368 mm long (6.2 in. x 14.5 in.). Figure 5-5 illustrates the card and the LS/GS jumper.

Facilities

Facilities provided by the LS/GS Trunk Card include:

- Loop Start or Ground Start selectable by jumper
- M and MM signaling leads available
- Trunk activity indicated by LED (one per trunk)
- Transient suppression on Tip, Ring, and signaling leads
- Alarm LED.

Electrical Description

The Loop Start/Ground Start Trunk Card mounts in any slot and interfaces six trunk circuits to the system. Each trunk circuit is programmed as loop start or ground start by a jumper clip prior to installation.

Each trunk has Tip and Ring leads and M and MM leads for additional signaling, if required. All leads are protected by varistors against transients between line and ground. There are also varistors between Tip and Ring and between M and MM. Each lead is in series with an inductor near the edge connector to reduce electromagnetic interference (EMI).

Each trunk has an LED on the front faceplate of the card that lights to indicate that the circuit is busy. An LED at the bottom of the faceplate lights to indicate a failure on the card.

Operation - Loop Start

To place an outgoing call, the trunk card places a termination across tip and ring. The CO detects the current flow and responds with dial tone. Now the user may begin to dial.

The Trunk Card recognizes an incoming call when it receives ringing voltage or battery reversal from the CO. The Trunk Card will respond by placing a termination across Tip and Ring. The trunk is released when the loop current is broken, either when the near party goes on-hook or the line is physically broken.

Operation - Ground Start

To place an outgoing call, the Trunk Card grounds the Ring lead. The CO responds by grounding the Tip lead. The Trunk then places a termination across Tip and Ring and ungrounds the Ring lead. The CO then sends dial tone to indicate that it is ready to receive dialing.

The Trunk Card recognizes an incoming call when the CO grounds the Tip lead. The CO may also send ringing voltage. The Trunk Card will respond by placing a termination across Tip and Ring. The trunk is released when the loop current is broken, either when one party goes on-hook or the line is physically broken.

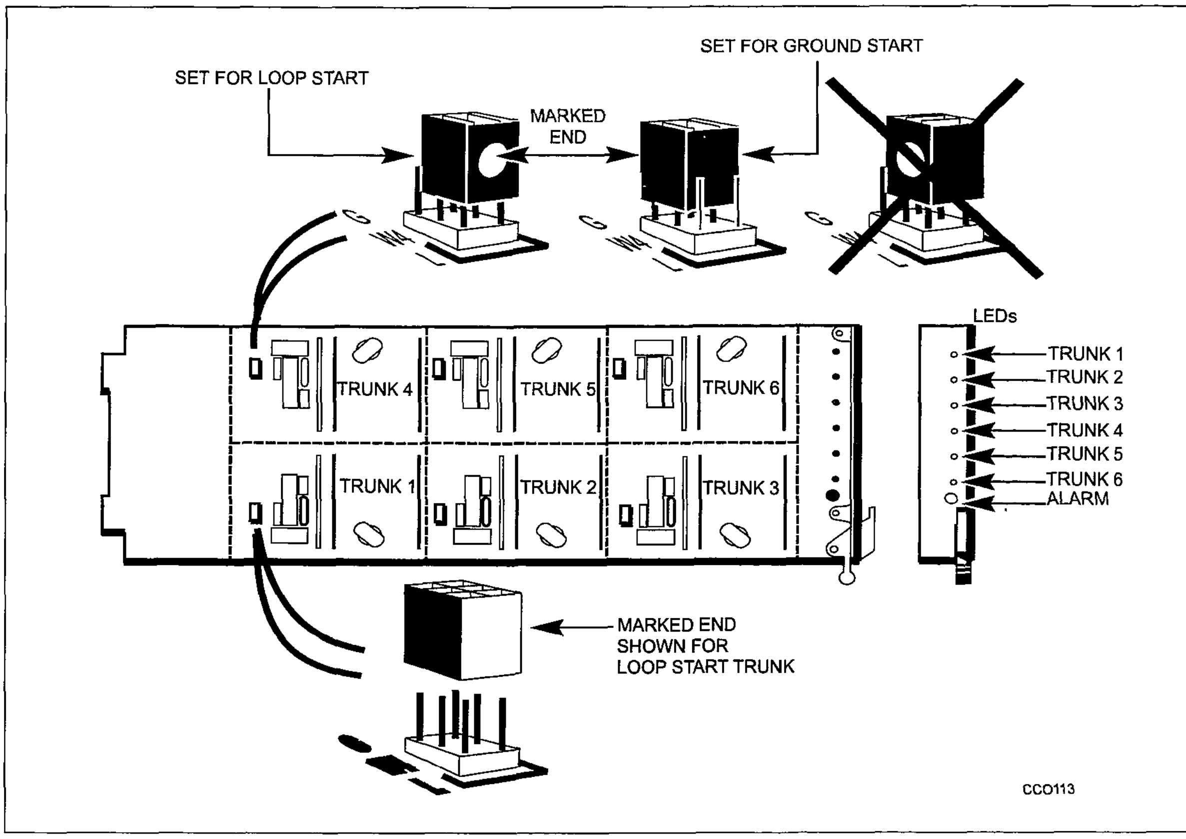


Figure 5-5 Loop Start/Ground Start Card and Jumper Settings

ONS Line Card

WARNING: **ANY CONNECTION OF THIS CARD TO AN OFF PREMISE APPLICATION, AN OUT OF PLANT APPLICATION, OR TO ANY OTHER EXPOSED PLANT APPLICATION MAY RESULT IN A SAFETY HAZARD, AND/OR DEFECTIVE OPERATION, AND/ OR EQUIPMENT DAMAGE.**

Brief Description

5.7 The On-Premises (ONS) Line Card, part number 9109-010-00X, interfaces standard subscriber telephone sets to a PABX in the same building. It contains 12 line circuits and plugs into any slot. The card is 158 mm high x 368 mm long (6.2 in. x 14.5 in.).

There are 13 LEDs on the front face of the card. The top 12 are each connected to a line circuit and light up to show that the circuit is in use. The LED at the bottom of the panel flashes to indicate an alarm (failure) condition.

One ML-only ONS line card, part number 9109-010-003-NA, is shipped with each *SX-200* ML PABX. Only one is allowed per PABX; additional ONS cards must be part number 9109-010-000-SA.

Facilities

Each line circuit provides the following facilities:

- Line protection
- Analog-to-Digital / Digital-to-Analog conversion (μ–law)
- Line circuit status monitoring
- Signaling (ringing, message waiting).

Electrical Description

The following description applies to each line circuit.

Tip and Ring are each protected against transients by a 200 volt varistor to ground. A bridge rectifier provides four protection diodes for the line circuit transistors. The -28 volt line is protected by a 35 volt transorb.

The line circuit performs 2-wire to 4-wire conversion, splitting the signal on the line into outgoing and incoming speech paths. The analog signal coming from the telephone is converted to pulse code modulation (PCM); the signal to be sent to the telephone is converted from PCM to analog audio. These conversions are performed by a Mitel Codec chip.

When the telephone is off-hook, the line circuit status LED on the front panel lights. The line circuit maintains a constant 26 mA current to the telephone while the set is off-hook. Loop length is maximum 600 ohms including the telephone set.

The ONS Line Card supports the Message Waiting feature. A high voltage (-140 Vdc) is applied to the Ring terminal of the line to light a neon lamp on the subscriber's set.

Operation

When a telephone goes off-hook, the line circuit detects the flow of loop current and signals the main processor. The processor responds by connecting a DTMF receiver to the line and sending dial tone to the set. (If the telephone uses pulse dialing, the processor detects the pulses by monitoring the loop current). The user can then dial the desired number.

When a call is directed to an extension, the system applies ringing voltage to the appropriate line and monitors the loop current for an off-hook condition. When the telephone is answered, the ringing voltage is removed.

When a call is ended by one of the sets going on-hook, the call is disconnected and the line returns to its idle state.

OPS Line Card

Brief Description

5.8 The OPS Line Card, part number 9109-040-000, contains six off-premises line circuits. An Off-Premises (OPS) line circuit is used where the line goes outside the building that houses the PABX and the line may be exposed to extraneous high voltages or induced currents (e.g., lightning).

The OPS Line Card can be used in any digital high-power (upper) slot. The maximum number of these cards is four per bay, providing a maximum of 24 ports per bay. The card is 157.5 mm high x 366.4 mm long (6.2 in. x 14.4 in.).

Major Components

Major components for the OPS Line Card are:

- Mitel 8962 Filter/Codec (six)
- 2-wire / 4-wire converter (six)
- Ringing relay (one per circuit)
- Alarm LED.

Facilities

Each line circuit provides the following facilities:

- Line activity LED
- Line protection
- Analog-to-Digital / Digital-to-Analog conversion (μ-law)
- Signaling (ringing).

Operation

Each circuit has a LED on the front panel which lights to indicate that the line is in use. A seventh LED at the bottom of the panel lights to indicate a failure on the card.

The line circuit applies forward battery feed to the line. The Tip is grounded and the Ring is at -48 volts. When the set goes off-hook to place a call, the PABX detects the loop current and responds with dial tone. Dialing may be DTMF or pulses. Dial pulses are debounced by software to assure reliable performance.

When a call is directed to the set, a relay closes and sends ringing voltage to the set. The ringing relay drops out when loop current flow indicates that the telephone has been answered (off-hook condition).

Electrical Description

Line protection comprises high voltage varistors to energy dump ground from Tip and Ring plus fusible links incorporated into the battery feed resistors. EMI is controlled by inductors in series with Tip and Ring.

The maximum loop resistance is 1800 ohms. The maximum loop length is 5850 m (19,200 ft) when using 26 AWG wire, 15,240 m (50,000 ft) when using 22 AWG wire.

The card circuitry performs 2-wire to 4-wire conversion that splits the signal on the line into outgoing and incoming speech paths. The analog signal coming from the line is converted to pulse code modulation (PCM); the signal to be sent to the line is converted from PCM to analog audio. These conversions are performed by a Mitel Codec chip.

The line circuit applies ringing voltage to the appropriate line through a relay and removes it when the telephone is answered. Answer is detected by monitoring the loop current.

Loop current is provided through a pair of 200 ohm resistors. Below 900 ohms loop resistance, active current limiting circuitry limits line power to less than 1.5 watts.

Ringing for each line is controlled by relay 2.

Condition	Relay 1	Relay 2
Idle or Talk	OFF	OFF
Ringing	OFF	ON

Digital Line Card

WARNING: **ANY CONNECTION OF THIS CARD TO AN OFF PREMISE APPLICATION, AN OUT OF PLANT APPLICATION, OR TO ANY OTHER EXPOSED PLANT APPLICATION MAY RESULT IN A SAFETY HAZARD, AND/OR DEFECTIVE OPERATION, AND/ OR EQUIPMENT DAMAGE.**

Brief Description

5.9 The Digital Line Card (DLC), part number 9109-012-00X, provides an interface from the PABX to the following:

- DATASET
- SUPERSET 401+™
- SUPERSET 410™
- SUPERSET 420™
- SUPERSET 430™
- PKM Module
- DNIC Music-on-Hold/Paging Unit (DMP)
- MILINK™ DATASETS
- SUPERCONSOLE 1000™ Attendant Console (must connect to a DLC installed in an upper slot).

The DLC contains 12 asynchronous line circuits (MITEL Digital Network Interface Circuit), and is a low-power digital card which can plug into any slot within a digital bay. The card is 158 mm high x 368 mm long (6.2 in. x 14.5 in.).

There are 13 LEDs on the face of the card. The top 12, one for each line circuit, light when the circuit is in use. The bottom LED on the panel lights to indicate an alarm condition within the card.

Two ML-only Digital line cards, part number 9109-012-002-NA, are shipped with each *SX-200* ML PABX. Only two are allowed per PABX; additional DLC cards must be part number 9109-012-000-SA.

Facilities

Each Digital Network Interface Circuit (DNIC) provides the following facilities:

- Line protection
- Full duplex simultaneous data and voice digital transmission over a single pair of wire
- Line circuit status monitoring
- Signaling and HDLC protocol to its associated DATASET.

Electrical Description

Each Digital Network Interface Circuit (DNIC) connects one of the devices listed in paragraph 5.9 to the common circuitry on the DLC card (and then to the PABX). The common circuitry will be described first, followed by a description of one DNIC.

The common circuitry contains a High-level Data Link Controller (HDLC) which controls the D channel communication between each DNIC and the Main Control Card II (MCC II) within the PABX. This data is passed over one-half of a link to the MCC II. The B1 and B2 channels from the DNICs are multiplexed onto one link between the DLC and the MCC II. The common circuitry includes phase-lock loop circuitry to keep the DNICs in synchronization with the system clock, as well as circuits which prevent the DLC from disrupting the backplane when a card is inserted or removed. The DLC line circuits are arranged in three groups of four; at power-up, each group can be separately sequenced. There are 12 Digital Network Interface Circuits (DNIC) on the DLC card; each is connected to a separate tip-ring pair.

DNIC Description

Each DNIC connects via its tip-ring pair to a proprietary telephone, a DNIC Music-on-Hold/Paging Unit, or a DATASET which also contains a DNIC. The DNICs communicate with each other over the twisted pair at 160 Kb/s (two 64 Kb/s B1 and B2 channels and a D channel). Since the DNIC is a proprietary integrated circuit, each device connected to a Digital Line Card DNIC tip-ring pair must also contain a DNIC. The two DNICs communicate data plus voice simultaneously in full duplex over the single twisted pair between them. The twisted pair also carries the power required by the *SUPERCONSOLE 1000, SUPERSET 401+, SUPERSET 410, SUPERSET 420*, or the *SUPERSET 430* telephone from the DNIC on the DLC. The DATASET is powered from a separate AC power supply.

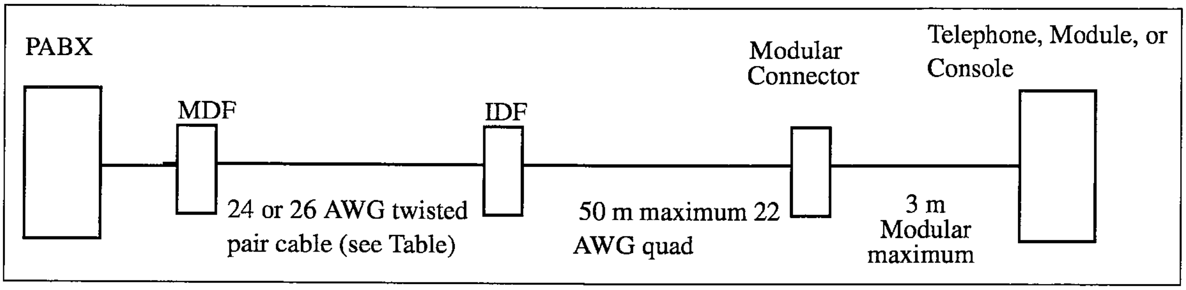
Operation

The Digital Line Card communicates with a DNIC-equipped device using digital transmission techniques: a voice channel, a data channel, and a control channel. It allows simultaneous transmission of voice and data over a single twisted pair of wires. When the DLC is connected to *SUPERSET 410, SUPERSET 420*, or *SUPERSET 430* telephones. Each telephone may be connected to a *MILINK* Data Module which can be connected to a personal computer or similar data device. The telephone's voice operation and the data device's data operation can both function concurrently.

Loop Length Specifications for Connections to a Digital Line Card

The following rules for loop lengths between the Digital Line Card within the PABX and the *SUPERSET* telephones, or *SUPERCONSOLE 1000* Attendant Console must be followed for proper operation of the device:

Maximum loop length (twisted pair) 24 or 26 AWG	see Table
Maximum length of quad cable (22 AWG)	50 m (160 ft)
Modular Line Cord	3 m (10 ft)



Peripheral Device	Maximum Loop Length	
	Without Bridge Tap	With Bridge Tap
SUPERSET 401+	1000 m	1000 m
SUPERSET 410	1000 m	1000 m
SUPERSET 420	1000 m	1000 m
SUPERSET 430	1000 m	1000 m
Dataset 1103	2000 m	1000 m
Dataset 2103	2000 m	1000 m
SUPERCONSOLE 1000 console	1000 m	1000 m

T1 Trunk Card

Brief Description

5.10 The T1 Trunk Card interfaces a single T1 trunk circuit to the PABX. The T1 protocol is used primarily in North America.

Facilities

The T1 Trunk Card provides the following facilities:

- T1 Clock - System Clock Phase Comparator
- Bidirectional T1 to ST-BUS data rate and format conversion
- Line Equalization.

General Description

The T1 interface will transmit and receive 24 8-bit voice/data channels on a 4-wire digital trunk that operates at 1.544 Mb/s. The BCC performs all control functions.

To provide signaling information on the T1 line, data bits are "stolen" from each channel to provide channel associated signaling.

The T1 Trunk Card includes a phase comparator which, through the T1 Clock on the Main Control Card II, keeps the system clock in phase with the incoming frame rate clock. The comparator prevents data losses caused by clock rate differences. (Refer to Main Control Card II in this Section). Phase error is the difference between the clock rate received on the link and the clock rate generated within the system (if the T1 clock is being adjusted to the incoming clock rate). If the difference climbs by greater than 1 in a single reading then the next three readings are filtered out. If the fourth reading has climbed greater than 1 then the link is considered unstable. A maintenance log is generated and the link is no longer used as a network synchronization source.

The system supports two T1 Trunk Cards (slots 5 and 6).

One incoming T1 trunk is selected as the primary timing source; the system locks its PCM clock and all other T1 trunk cards in the system to this incoming data stream.

On-site T1 termination equipment, the Channel Service Unit (CSU) is required. The T1 Trunk Card includes switches to set appropriate line equalization for cable lengths up to 200 meters (655 feet) from the CSU.

An adaptor fitted to the backplane connector provides a 15-pin D-Sub connector for the T1 facility and a 25-pair connector to maintain access to the adjacent odd-numbered card slot. Only one T1 Trunk Card can be installed.

Electrical Description

In the transmit direction, the data from the system PCM link must be converted from the Mitel ST-BUS format of 32 channels at 2.048 MBit/sec to the T1 format of 24 channels at 1.544 MBit/sec. To match the number of data channels, the T1 card skips every fourth channel on the ST-BUS links. The T1 interface circuits retime the output data to 1.544 Mbits/sec and add the framing bits.

In the receive direction, the framing bits are removed, the data is retimed to 2.048 MBit/sec and the channels are mapped onto the system PCM link in the same manner as above, with every fourth channel left empty.

The system runs a loopback test during power-up or reset. A relay is operated to loop the card's transmit line back to its receive line, removing the transmit data from the output line. A 6db attenuation is automatically added to the signal during loopback, to simulate actual transmission. The card remains in loopback until the test is passed.

The relative phase of the T1 and system clocks is determined as follows: An 8 KHz clock output derived from the incoming frame rate is divided by two. This 4KHz signal gates a counter, which counts the number of cycles of the system 2.048 MHz clock occurring during one cycle of the 4 KHz signal. This phase count is sampled every 100 ms. The oscillator on the T1 Clock on the Main Control Card II is adjusted to keep the phase count value constant over a longer term (16 seconds). The T1 circuit data buffer accommodates short-term phase variation (jitter).

Indicators

If any of the data channels on the T1 link are active, the upper front panel LED lights, giving visual indication of channel activity.

If the PABX is receiving a yellow alarm condition, the yellow (NO SYNC) LED lights. If transmitting a yellow alarm condition, and not receiving a yellow alarm condition, then the LED is off.

The lower red LED is the card alarm. If the PABX generates or receives a yellow alarm condition on the T1 link, the ALARM LED flashes.

Physical Description

The card measures approximately 158 X 368 mm (6.2 in. X 14.5 in.). The switches used (SW1) to set up proper equalization on the T1 Trunk card must be set as follows. Lengths are for cable length, not loop length.

Short Cable (under 150 feet)	S1 only - Closed
Medium Length Cable (150 to 450 feet	S2, S3, S4 only - Closed
Long Length Cable (450 to 655 feet)	S5, S6, S7 only - Closed

COV Line Card

WARNING: **ANY CONNECTION OF THIS CARD TO AN OFF PREMISE APPLICATION, AN OUT OF PLANT APPLICATION, OR TO ANY OTHER EXPOSED PLANT APPLICATION MAY RESULT IN A SAFETY HAZARD, AND/OR DEFECTIVE OPERATION, AND/ OR EQUIPMENT DAMAGE.**

Brief General

5.11 The COV Card, part number 9109-020-000, is installed in a digital peripheral bay to interface to voice mail systems that require a COV interface. The card measures 158 mm x 368 mm (6.2 in. x 14.5 in.). It has a profiled edge connector that allows it to be safely inserted or removed from the system with the power on.

The COV Card can be mounted only in a high-power (upper) card slot of a digital bay.

Major Components

The major components of the COV Line Card are:

- Subscriber line interface circuit (SLIC) - one per line
- Backplane interface
- PCM Timer
- 6402 UART
- 8840 Modem

- Line protection circuits
- Line status LEDs - one per line
- Card status LED - one.

Facilities

Facilities provided by the COV Line Card include:

- Amplitude Shift Keyed communication
- Analog/Digital and Digital/Analog conversions (μ-law)
- Battery Feed to power telephones.

Electrical Description

Control information from the backplane is converted to a 32 kHz amplitude shift-keyed (ASK) data stream. The audio information is taken from the 2 Mb/s data link, converted to analog audio and combined with the control information for transmission to the telephone. Conversely, the audio and ASK data signals from the telephone are separated and converted. The ASK data is demodulated and sent to the bay processor. The audio is digitized and transmitted on the data link.

There is only one UART and one modem on the card. The six lines are time-division-multiplexed to the communication circuit. Transmission and reception are simultaneous, but the card receives data from the telephone to which it last transmitted. For example, if the card is transmitting to telephone 2, it is receiving from telephone 1. In the next time slot, it will transmit to telephone 3 and receive from telephone 2.

The COV Card has seven indicators on the front panel. There is an activity LED for each subscriber line. The LED at the bottom of the panel is an alarm indicator for the entire card.

The maximum loop lengths for COV circuits are:

Wire Gauge (AWG)	Max. Loop length
26	1000 m (3300 ft)
24	1500 m (5000 ft)
22	2000 m (6600 ft)

Music-on-Hold/Pager Unit

5.12 The Music-on-Hold/Pager unit interfaces a standard *SX-200* DNIC port to the following external equipment:

- External music source for Music-on-Hold
- External paging amplifier (with or without answerback capability)
- Up to two night bells
- An external alarm.

The unit is powered by the *SX-200* ML PABX and does not require a separate power source. A single 25-pair amphenol connects to the *SX-200* ML PABX via the main distribution frame. A single LED indicator provides basic status information. The unit can be wall-mounted next to the *SX-200* ML PABX.

Each Music-on-Hold/Pager Unit supports a single paging zone. If more than one paging zone is required, then additional Music-on-Hold/Pager Units can be added as required.

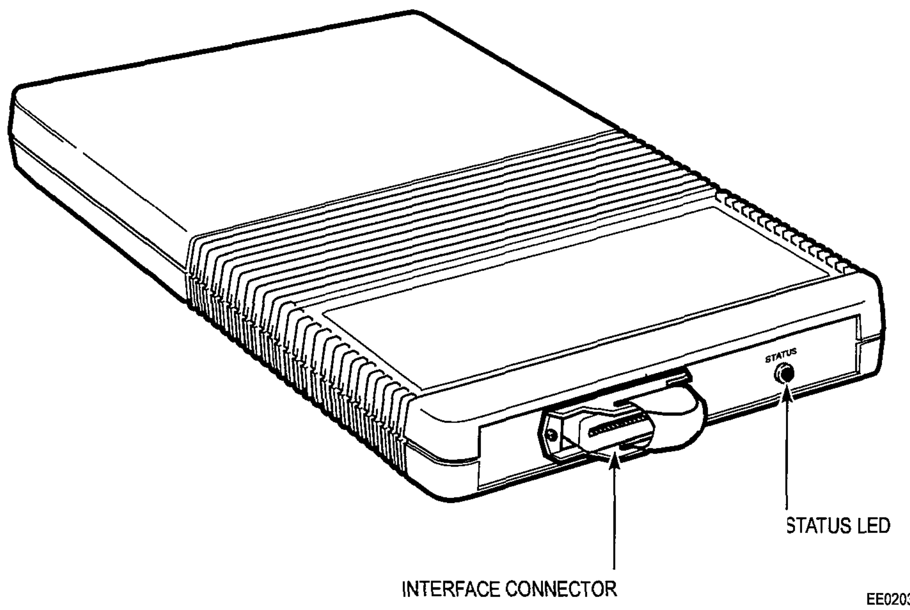


Figure 5-6 Music-on-Hold/Pager Unit

LED Indicator

The LED indicates the following states:

Table 5-1 Music-on-Hold/Pager Unit LED Indicator

Indication	Status
OFF	No power from *SX-200* ML DNIC circuit
Flashing (2 Hz - On 250 ms, Off 250 ms.)	Loss of synchronization with *SX-200* ML PABX, the MOH/ Pager Unit may be faulty
On Solid	MOH/Pager Unit is operating
Winking (2 Hz - ON 50 ms, Off 1 sec.)	MOH/Pager Unit is operating and the paging amplifier is being accessed

Interface Technical Specifications

MOH Input

The MOH input is a transformer-coupled input with impedance of 600 ohms. The gain is fixed at -4 dB A/D. In order to meet signal power limits applied to such interfaces by FCC part 68 and Industry Canada, CS03. The audio level at the digital point will be limited to -15 dBm0 and the frequency response is rolled off (3 dB point) at approximately 3 Khz. The amplitude limiting begins to affect the input signal when it exceeds approximately 100 mVrms; therefore, input signals should be in the range of 10 to 100 mVrms.

Paging Input/Output

The paging interface is a transformer-coupled loop start trunk with an impedance of 600 ohms. The DC termination is activated by a relay when the paging amplifier is being accessed. The gain is fixed at -4 dB A/D and +3 dB D/A.

When used with an "answer back" paging amplifier, this interface must meet signal power limits applied to such interfaces by FCC part 68 and Industry Canada CS03. The audio level at the digital point will be limited to -15 dBm0 and the frequency response is rolled off (3 dB point) at approximately 3 Khz. The amplitude limiting begins to affect the input signal when it exceeds approximately 100 mVrms; therefore, answerback signals should be in the range of 10 to 100 mVrms.

Relays

Relays are provided for Paging, Night Bells, and Alarms. The contacts are rated at 0.1 A at 90 Vac or 0.5 A at 48 Vdc.

Paging Control Relay. Both normally open and normally closed contacts are provided to control the external paging amplifier. These relay contacts are switched whenever the paging amplifier is being accessed.

Night Bell Relays. Two independent, normally open relays are provided to control external night bells.

Alarm Relay. One normally open relay is provided to control external devices such as an alarm.

6 Signaling and Supervision

General

6.1 This Part outlines the signaling and supervision parameters of the PABX.

The standard range of tones are available from the PABX's Digital Signal Processor:

- 12 DTMF sets of tones, as listed in Table 6-1.
- A set of call progress tones as listed in Part 9, which form part of the country's Audible Tone Plan.
- One ringing tone of 20 Hz.

The PABX is capable of accepting and repeating signals from telephone sets which have the parameters shown in Table 6-1, DTMF Tone Parameters and Table 6-2, Dial Pulse Reception Limits.

Where any of the frequencies shown in Part 9 are present at the system input, any other single frequency (200 - 3400 Hz) should be a minimum of 40 dB below the signal frequency. DTMF pulses are registered in the presence of precise dial tone at a level of -10 dBm.

The PABX gives the following output signal conditions:

- Dial Pulse Conditions:

  Pulse Rate : 9 to 11 pps
  Break Interval : 58% to 62%
  Interdigit Time : 800 ms.

- DTMF Dialing Conditions for North America:

  Frequency Deviation : ± 1%
  Tone Duration : greater than 90 ms
  Interdigit Time : greater than 100 ms
  Level, low group : greater than -4 dBm
  Level, high group : greater than - 4 dBm
  Level, DTMF signal : less than -1 dBm
  Level, third Harmonic: better than 40 dB Frequency below DTMF signal
  Twist : 0 dB.

Table 6-1 DTMF Tone Parameters			
	High Frequency (Hz)		
Low Frequency (Hz)	**1209**	**1336**	**1477**
697	1	2	3
770	4	5	6
852	7	8	9
941	*	0	#

Frequency deviation:	± 1.5 %
Signal interval (2 frequency):	40 ms (min)
Per frequency, minimum level:	-17 dBm on line circuit
Twist, maximum (at -10 dBm):	+4 to -8 dB (high frequency relative to low frequency)

Table 6-2 Dial Pulse Reception Limits		
Parameter	**Min**	**Max**
ONS Line:		
Pulse Rate	8 pps	12 pps
Break Duration	58%	64%
Interdigit Time	300 ms	15 s
OPS Line:		
Pulse Rate	8 pps	12 pps
Break Duration	42%	84%
Interdigit Time	300 ms	15 s

7 Transmission

General

7.1 The following descriptions detail some of the transmission characteristics which apply to the PABX.

Frequency Response

7.2 The frequency response relative to 1004 Hz at 0 dBm for different types of interconnections is shown in Table 7-1.

Table 7-1 Frequency Response

		Variation in Attenuation with respect to 1004 Hz (dB)	
Interconnection	**Frequency (Hz)**	**minimum**	**maximum**
Line to Line	60	-20.0	--
or	200	0	-5.0
Line to 2-W Analog Trunk	300	0.5	-1.0
or	3000	0.5	-1.0
2-W Analog Trunk to 2-W	3200	0.5	-1.5
Analog Trunk	3400	0	-3.0
Line to 4-W Analog Trunk	60	-20.0	--
or	200	0	-4.0
2-W Analog Trunk to 4-W Analog Trunk	300	0.4	-0.65
	3000	0.4	-0.65
	3200	0.4	-1.5
	3400	0	-3.0
4-W Analog Trunk to 4-W Analog Trunk	60	-16.0	--
	200	0	-3.0
	300	0.3	-0.3
	3000	0.3	-0.3
	3200	0.3	-1.5
	3400	0	-3.0

Engineering Information

Overload - Digitally Switched Analog Card Types

7.3 The overload levels shown in Table 7-2 define the maximum signal levels which may be applied to an analog interface before clipping occurs. The Input Overload (IOL) defines the maximum analog input signal level which can be applied to a digital interface circuit before clipping of the encoded PCM word occurs. The Output Overload (OOL) defines the maximum analog output signal which can be produced at the output of an interface by the application of a 3 dBm0 digital signal to the input.

Table 7-2 Overload Levels - Digitally Switched Analog Card Types

		Overload Point (dB)	
Interface Type	**Connecting Circuit**	**IOL**	**OOL**
ONS(A)	any circuit	6.0	0
ACO(A)	ONS(D)	0	3.0
ACO(A)	any trunk	3.0	3.0

Quantization Distortion

7.4 The quantization distortion of a connection is a measure of the signal to distortion ratio as the input signal is varied. Note that this parameter does not apply to the analog card types. For 95% of connections the signal to distortion ratio will exceed the limits shown in Table 7-3.

Table 7-3 Signal To Distortion Ratio

1004 Hz Input Level (dBm0)	Signal/Distortion Ratio (dB) 95% of all connections
0 to -30	33
-40	27
-45	22

Intermodulation Distortion

7.5 Intermodulation (harmonic) distortion is measured by using two pairs of equal level tones (851/863 Hz and 1372/1388 Hz), at a total composite input power of -13.0 dBm0. Table 7-4 shows the second and third order products for the different types of connections. For 95% of the connections in each category, intermodulation distortion will exceed the stated limits.

Table 7-4 Intermodulation Requirements (4.8 KB/s)

Connection Type	Second Order Product (dB)	Third Order Product (dB)
Line to Line	40	43
Line to Trunk	45	53
Trunk to Trunk	45	53

Return Loss

7.6 The ERL and SFRL return losses for different types of interconnection or idle states are shown in Table 7-5.

Table 7-5 Return Loss Characteristics							
	Terminal Balance		Through Balance		Transhybrid Loss		Non-Talking
Connection Type	ERL (dB)	SFRL (dB)	ERL (dB)	SFRL (dB)	200 - 3400 Hz	500 - 2500 Hz	(dB)
Line to Line	18	12					
Line to 4-W Trunk	24	14					
4-W Trunk to Line	24	14					
Line to 2-W Trunk	18	12					
2-W Trunk to Line	18	12					
4-W Trunk to 2-W Trunk	28	14					
2-W Trunk to 4-W Trunk	28	14					
4-W Trunk to 4-W Trunk			27	20			
ONS(D) Line					17	19	
2-W Trunk					18	21	
Loop Start LS/GS Trunk (2-W)							6
Ground Start LS/GS Trunk (2-W)							2

Crosstalk

7.7 The crosstalk attenuation, or coupling loss, between any two transmission paths at any frequency between 200 and 3200 Hz is greater than 75 dB for 95% of all connections.

Echo Path Delay

7.8 The round trip echo path delay with a frequency of 1500 Hz does not exceed the stated values for the following types of interconnection:

- Line to Line: 3.0 ms
- Line to Trunk: 3.0 ms
- Trunk to Trunk: 3.0 ms.

Envelope Delay Distortion

7.9 Table 7-6 details the maximum envelope delay distortion for the digital card types. These objectives are met by 95% of all connections. For the analog card types, the envelope delay distortion is less than 200 microseconds between 400 and 3200 Hz, for all connections.

Table 7-6 Envelope Delay Distortion		
Connection Type	**Frequency Band (Hz)**	**Envelope Delay Distortion (microseconds)**
Line to Line	1000 - 3000	less than 280
	400 - 3200	less than 560
Line to Trunk	1000 - 3000	less than 140
	400 - 3200	less than 280
Trunk to Trunk	1000 - 3000	less than 140
	400 - 3200	less than 280

Longitudinal Balance

7.10 All connections (except ONS) meet the longitudinal balance requirements outlined in Table 7-7. Note that these apply to OFF-HOOK circuits only.

Table 7-7 Longitudinal Balance		
Frequency (Hz)	**Longitudinal Balance (dB)**	
	Minimum	**Average**
200	58	63
500	58	63
1000	58	63
3000	53	58

System Impedances

7.11 System impedances are as follows:

- Station - 9109-010: 600 ohms input impedance
  600 ohms DC loop resistance.
- LS/GS Trunk Loop:
  600 ohms input impedance, 1600 ohms loop range.
- LS/GS Trunk T/R to ground Resistance - both in the IDLE state :
  - greater than 30 Kohms (ground start)
  - greater than 10 Mohms (loop start).
- Analog type lines: 600 ohms ac input impedance, nominal
- Analog type trunks: 600/900 ohms ac input impedance, nominal

Idle Channel Noise - C Message

7.12 The idle channel C message noise will not exceed the following values for any type of interconnection:

- Average: 16 dBrnC
- 95% of all interconnections: 20 dBrnC

Idle Channel Noise - 3 kHz Flat

7.13 The idle channel noise for 3 kHz flat noise requirements do not exceed the following values for any type of interconnection:

- 95% of all interconnections: 39 dBrn0
- 50% of all interconnections: 35 dBrn0

8 Loss and Level Plan

General

8.1 This part describes the loss and level plan for North American applications. A large number of interconnections are possible, ranging from interconnections between on-premises (ONS) line circuits, to the interconnection of remotely located satellite PABXs with this PABX. This part describes the principles of the loss and level plans, their application to the PABX, and the arrangements for setting the transmission levels.

North American Loss and Level Plans

8.2 The purpose of a transmission loss and level plan is to provide an acceptable transmission grade of service to all subscribers in the telephone network. At present, two loss plans exist for the public switched network in North America. They are the VIA NET LOSS (VNL) plan and the SWITCHED DIGITAL NETWORK (SDN) plan.

VNL Plan

The VNL plan consists of two parts: a fixed loss portion and a variable loss portion as follows:

- **Fixed Portion.** A minimum fixed amount of loss is introduced into all Toll connections. This loss is equal to 5 dB and is split equally between the two end trunks connected to the Toll network.
- **Variable Portion.** In addition to the fixed portion, a variable loss is introduced into all trunks involved in a connection. This loss, known as VNL, is proportional to the trunk length and its propagation delay. The loss ranges from a minimum of 0.5 dB to a maximum of 3.0 dB, and covers trunk lengths from 0 to about 2900 km (0 to 1800 miles). Trunks in excess of this length employ echo suppressors and are designed to zero loss.

The loss objectives for Toll connections that use the VNL plan range from a minimum of 5.5 dB to a maximum of 8.0 dB between end-to-end CO offices (CL5 to CL5), as illustrated in Table 8-2.

SDN Loss Plan

The Switched Digital Network (SDN) loss plan was developed to meet the needs of the evolving digital public switched network. This plan does not assign losses to intermediate links in a connection. Under the plan, the local area public network (local CO to local CO) is operated at zero loss. This feature eliminates the need to introduce digital padding on intermediate digital trunk links and maintains data transparency throughout the network. Control of echo and noise with this plan is achieved by inserting fixed amounts of loss at the end points where the conversion to analog takes place. A compromise value of 6 dB was selected for line-to-line connections over the Toll network. This loss is inserted in the receive (RX) direction of transmission (D-A) under software control. This is illustrated in Figure 8-2.

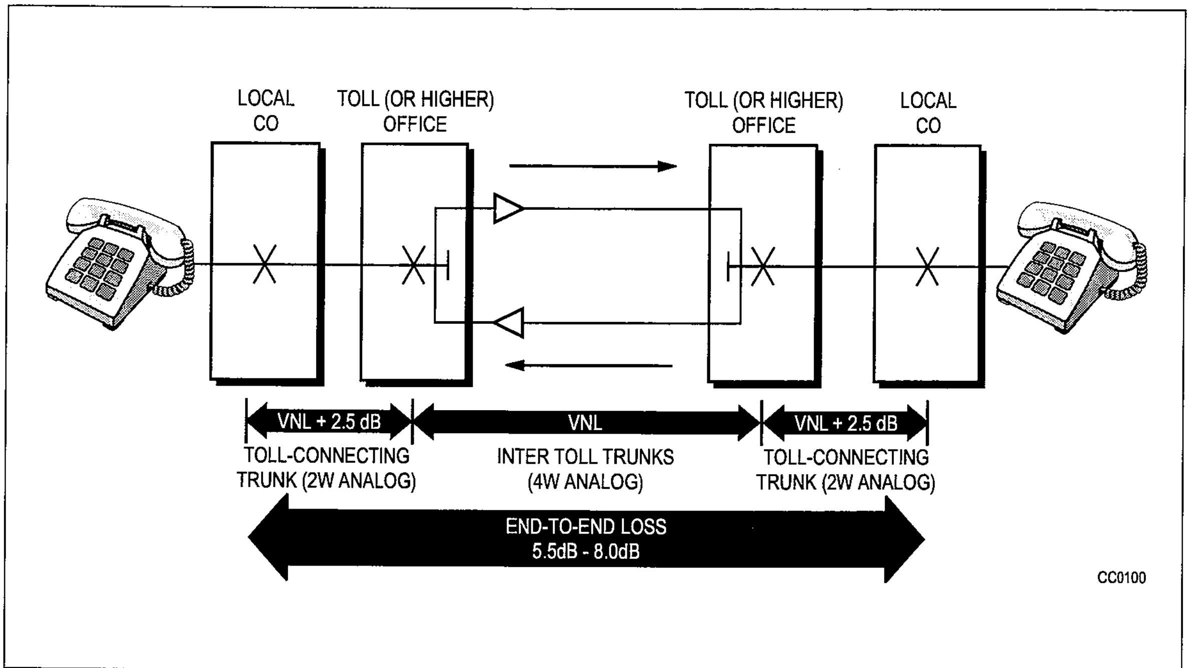


Figure 8-1 Local to Local Central Offices VNL Objectives

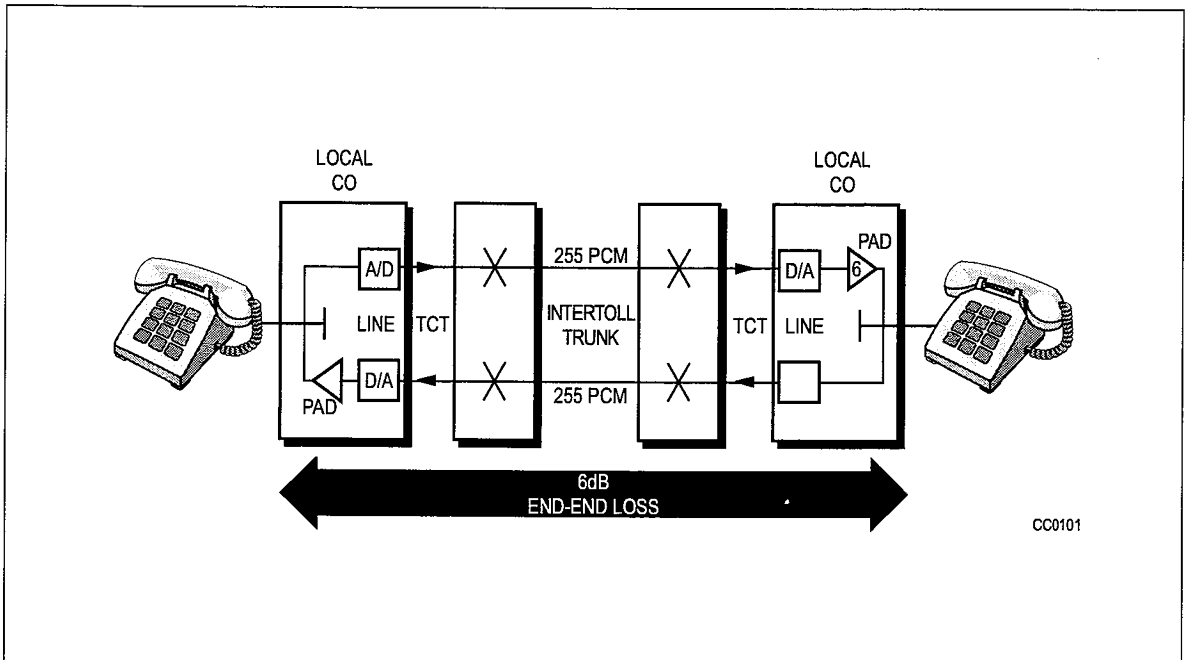


Figure 8-2 Local to Local Central Offices Fixed Loss Plan Objectives

Loss And Level Plan

The loss and level plan used for the PABX is illustrated in Figure 8-3. This layout is not intended to be a typical network, but is drawn to show the different types of trunk and line interfaces which the PABX will accommodate.

Table 8-1 explains the terms used for the different types of peripheral interfaces.

Table 8-1 Interface References

Circuit Ref	Description	Card Type (Part No.)
ONS	An industry-standard telephone set may be connected to the PABX by means of the following card types:	
ONS(D)	ONS Line Card	9109-010
OPSS	An industry-standard telephone set located off premises may be connected via the OPS Line Card.	9109-020
OPSL	An industry-standard telephone set located outside the range of the ONS circuit (over 2 kilometers) may be connected to the PABX via the OPS Line Card.	9109-020
ACO	An Analog CO (ACO) trunk can be connected to the PABX by means of one of the following types of trunk interface cards:	
ACO(D)	LS/GS Trunk Card	9109-011
ACO(D)	DID Trunk Card	9109-031
ATO	An Analog Toll Office (ATO) trunk may be connected to the PABX by means of one of the following trunk interface cards:	
ATO(D)	LS/GS Trunk Card	9109-011
ATO(D)	E&M Trunk Module	9109-013
ATT	An Analog Tie Trunk (ATT) can be connected between this PABX and another PBX, over a 2- or 4-wire analog trunk by means of one of the following trunk cards:	
ATT(D)	E&M Trunk Module	9109-013
ATT(D)	DID Trunk Card	9109-031
DCO	A Digital Central Office Trunk (DCO) connects a Digital CL5 office over digital facilities.	
DTO	A Digital Toll Office Trunk (DTO) connects a Digital PABX to a Digital CL4 or higher office over digital facilities.	
DTT	A Digital Tie Trunk (DTT) connects a Digital PABX to a Digital PABX over digital facilities.	
T1 Trunk	Digital T1 Trunk Card.	9109-021
CTT	A Combination Tie Trunk connects a Digital PABX to a remote channel bank over digital facilities. The remote channel bank usually interfaces to an analog PBX and provides the A/D & D/A conversion.	
sATT sDTT sCTT	This designation refers to a "satellite" tie trunk which connects a Main PBX to a Satellite PBX. The type of trunk cards used to interface are the same as those for the ATT, DTT, and CTT interfaces described above.	

To implement the required loss objectives (Table 8-2) of the plan, combinations of fixed and software selectable analog and digital transmission pads are provided. All analog padding is provided by Mitel 8960 series combined Codec/Filter integrated circuits, over a 7 dB range in 1 dB increments. Padding is provided in both the Transmit (TX) and Receive (RX) directions (see Part C).

Satellite PABX

A satellite PABX (shown in Figure 8-3) is defined as a PABX which has no direct connection to the serving central office for incoming traffic. It has no directory number, and receives all incoming calls through the main PABX over satellite trunks. The satellite PABX is usually in the same local area as its main PABX. Note that the Loss Plan differs for "SATELLITE" trunks in the main PABX from "SATELLITE" trunks in the satellite PABX.

Analog Transmission Pad Arrangements - Digital Interfaces

The analog transmit pad (A/D) setting defines the input level required to produce a 0 dBm0 digital signal, as well as the overload point of the interface (approximately 3 dB above the 0 dBm0 signal level). The Tx pad comprises a fixed portion and a variable portion. The fixed portion is incorporated into the analog interface to the codec/filter. The variable portion (0 to 7 dB) is incorporated into the codec/filter.

The analog receive pad (D/A) defines the output level produced by a digital milliwatt input signal. The Rx pad comprises a fixed portion which is part of the analog interface to the codec/filter, and a variable portion (0 to -7 dB) which is incorporated into the codec/filter. This pad arrangement is illustrated in Figure 8-4.

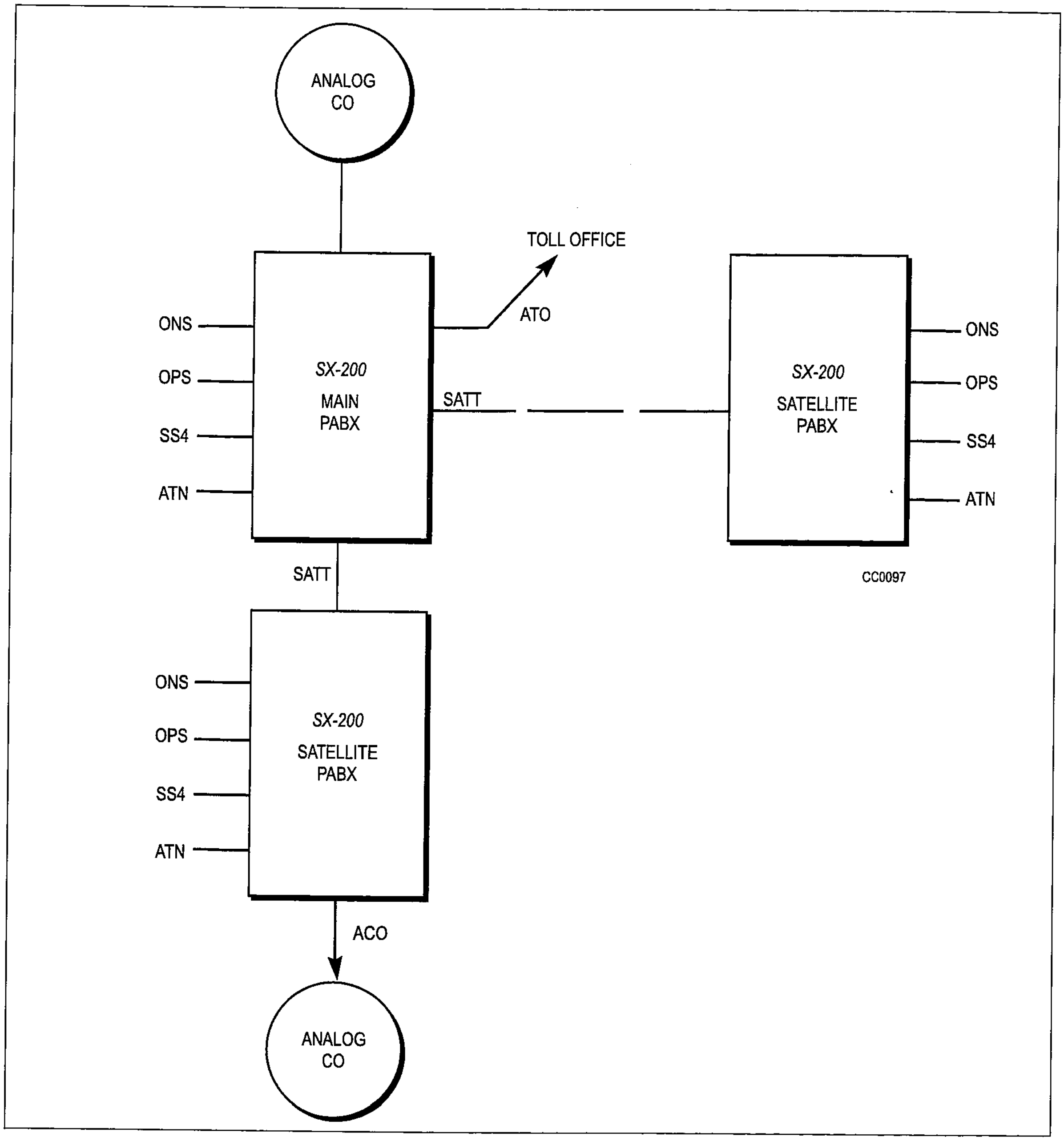


Figure 8-3 Types of Trunk and Line Interfaces

Table 8-2 PABX Loss Plan			
Originating Circuit	**Connecting Circuit**	**Loss (dB) Tx Direction**	**Loss (dB) Rx Direction**
ONS(D)	ONS(D)	-6.0	-6.0
	OPSS	-6.0	-6.0
	OPSL	-3.0	-3.0
	ACO(D)	0	0
	ATT(D)	-3.0	-3.0
	ATO(D)	-6.0	-6.0
	DCO	-3.0	-3.0
	DTO	-3.0	-9.0
	DTT	-3.0	-9.0
	sATT	-3.0	-3.0
	sDTT	-3.0	-3.0
	sCTT	-3.0	-9.0
OPSS	ONS(D)	-6.0	-6.0
	OPSS	-6.0	-6.0
	OPSL	-3.0	-3.0
	ACO(D)	0	0
	ATT(D)	-3.0	-3.0
	ATO(D)	-6.0	-6.0
	DCO	-3	-3
	DTO	-3	-9
	sATT	-3.0	-3.0
	sCTT	-3.0	-9.0
	sDTT	-3	-3
OPSL	ONS(D)	-3.0	-3.0
	OPSL	0	0
	OPSS	-3.0	-3.0
	ACO(D)	0	0
	ATT(D)	-2.0	-2.0
	ATO(D)	-3.0	-3.0
	DCO	0	0
	DTO	0	-6
	DTT	0	-6
	sATT	-2.0	-2.0
	sDTT	-2	-2
	sCTT	0	-6
			Page 1 of 4

Table 8-2 PABX Loss Plan (continued)			
Originating Circuit	**Connecting Circuit**	**Loss (dB) Tx Direction**	**Loss (dB) Rx Direction**
ACO(D)	ONS(D)	0	0
	OPSS	0	0
	OPSL	0	0
	ACO(D)	0	0
	ATT(D)	-2.0	-2.0
	ATO(D)	-3.0	-3.0
	DCO	0	0
	DTO	0	-6
	DTT	0	-6
	sATT	0	0
	sDTT	0	0
	sCTT	3	-3
	ONS(D)	-0.2	-0.2
ATT(D)	ONS(D)	-3.0	-3.0
	OPSS	-3.0	-3.0
	OPSL	-2.0	-2.0
	ATT(D)	0	0
	DCO	-2	-2
	DTO	3	-3
	DTT	3	-3
	sATT	0	0
	ACO(D)	-2.0	-2.0
	ATO(D)	0	0
	sDTT	0/-2	0/-2
	sCTT	3	-3
ATO(D)	ONS(D)	-6.0	-6.0
	OPSS	-6.0	-6.0
	OPSL	-3.0	-3.0
	ACO(D)	-3.0	-3.0
	ATT(D)	0	0
	ATO(D)	0	0
	DCO	-3	-3
	DTO	3	-3
	DTT	3	-3
	sATT	-3.0	-3.0
	sDTT	-3	-3
	sCTT	3	-3
			Page 2 of 4

Engineering Information

Table 8-2 PABX Loss Plan (continued)			
Originating Circuit	**Connecting Circuit**	**Loss (dB) Tx Direction**	**Loss (dB) Rx Direction**
	ONS(D)	-3.0	-3.0
sATT	OPSS	-3.0	-3.0
	OPSL	-2.0	-2.0
	ACO(D)	0	0
	ATT(D)	0	0
	ATO(D)	-3.0	-3.0
	DCO	0	0
	DTO	0	-6
	DTT	0	-6
	sATT	0	0
	sDTT	0	0
	sCTT	0	-6
	ONS(D)	-3	-3
DCO	OPSS	-3	-3
	OPSL	0	0
	ACO(D)	0	0
	ATT(D)	-2	-2
	ATO(D)	-3	-3
	DCO	0	0
	DTO	0	-6
	DTT	0	-6
	sATT	0	0
	sDTT	0	0
	sCTT	0	-6
	ONS(D)	-9	-3
DTO	OPSS	-9	-3
	OPSL	-6	0
	ACO(D)	-6	0
	ATT(D)	-3	3
	ATO(D)	-3	3
	DCO	-6	0
	DTO	0	0
	DTT	0	0
	sATT	-6	0
	sDTT	-6	0
	sCTT	0	0
			Page 3 of 4

Table 8-2 PABX Loss Plan (continued)

Originating Circuit	Connecting Circuit	Loss (dB) Tx Direction	Loss (dB) Rx Direction
DTT	ONS(D)	-9	-3
	OPSS	-9	-3
	OPSL	-6	0
	ACO(D)	-6	0
	ATT(D)	-3	3
	ATO(D)	-3	3
	DCO	-6	0
	DTO	0	0
	DTO	0	0
	DTT	0	0
	sATT	-6	0
	sDTT	-6	0
	sCTT	0	0
sCTT	ONS(D)	-9	-3
	OPSS	-9	-3
	OPSL	-6	0
	ACO(D)	-3	3
	ATT(D)	-3	3
	ATO(D)	-3	3
	DCO	-6	0
	DTO	0	0
	DTO	0	0
	DTT	0	0
	sATT	-6	0
	sDTT	-6	0
	sCTT	0	0
sDTT	ONS(D)	-3	-3
	OPSS	-3	-3
	OPSL	-2	-2
	ACO(D)	0	0
	ATT(D)	0/-2	0/-2
	ATO(D)	-3	-3
	DCO	0	0
	DTO	0	-6
	DTT	0	-6
	sATT	0	0
	sDTT	0	0
	sCTT	0	-6

Page 4 of 4

Engineering Information

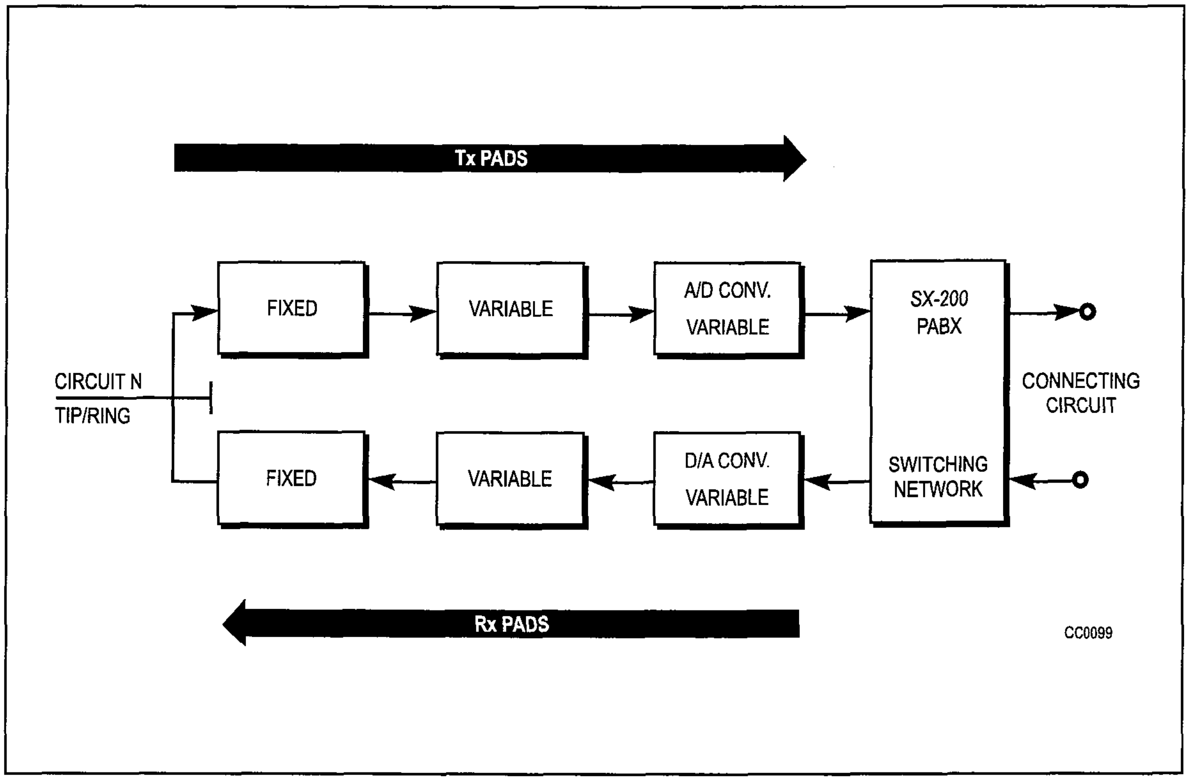


Figure 8-4 Analog Pad Arrangements - Configuration A

Conferencing Loss Plan

8.3 Any system will have at least one 18-port conference circuit (Digital Signal Processor). The conferencing circuit, on a frame-by-frame basis, compares the level of all conferees and sends the loudest signal to all other parties in the conference. The party which is the loudest receives audio from the second loudest party. The conference loss plan is outlined in Table 8-3.

Table 8-3 Conferencing Loss Plan

Originating Circuit	Connecting Circuit	Loss (dB) Tx Direction	Loss (dB) Rx Direction
ONS(D)	ONS(D)	-6.0	-6.0
	OPSS	-6.0	-6.0
	OPSL	-3.0	-3.0
	ACO(D)	-4.0	-4.0
	ATT(D)	-5.0	-5.0
	ATO(D)	-6.0	-6.0
	DCO	-3.0	-3.0
	DTO	-3.0	-9.0
			Page 1 of 5

Table 8-3 Conferencing Loss Plan (continued)

Originating Circuit	Connecting Circuit	Loss (dB) Tx Direction	Loss (dB) Rx Direction
	DTT	-3.0	-9.0
	sATT	-3.0	-3.0
	sDTT	-3.0	-3.0
	sCTT	-3.0	-9.0
	sATT	-3.2	-3.2
	sDTT	-3.2	-3.2
	sCTT	-3.2	-9.2
OPSS	ONS(D)	-6.0	-6.0
	OPSS	-6.0	-6.0
	OPSL	-3.0	-3.0
	ACO(D)	-4.0	-4.0
	ATT(D)	-5.0	-5.0
	ATO(D)	-6.0	-6.0
	DCO	-3.0	-3.0
	DTO	-3.0	-9.0
	DTT	-3.0	-9.0
	sATT	-3.0	-3.0
	sDTT	-3.0	-3.0
	sCTT	-3.0	-9.0
OPSL	ONS(D)	-3.0	-3.0
	OPSL	0	0
	OPSS	-3.0	-3.0
	ACO(D)	-1.0	-1.0
	ATT(D)	-2.0	-2.0
	ATO(D)	-3.0	-3.0
	DCO	0	0
	DTO	0	-6.0
	DTT	0	-6.0
	sATT	0	0
	sDTT	0	0
	sCTT	0	-6.0
ACO(D)	ONS(D)	-4.0	-4.0
	OPSS	-4.0	-4.0
	OPSL	-1.0	-1.0
	ACO(D)	-2.0	-2.0
	ATT(D)	-2.2	-2.2

Page 2 of 5

Engineering Information

Table 8-3 Conferencing Loss Plan (continued)			
Originating Circuit	**Connecting Circuit**	**Loss (dB) Tx Direction**	**Loss (dB) Rx Direction**
	ATO(D)	-4.0	-4.0
	DCO	-1.0	-1.0
	DTO	-1.0	-7.0
	DTT	-1.0	-7.0
	sATT	-1.0	-1.0
	sDTT	-1.0	-1.0
	sCTT	-1.0	-7.0
ATT(D)	ONS(D)	-5.0	-5.0
	OPSS	-5.0	-5.0
	OPSL	-2.0	-2.0
	ATT(D)	-4.0	-4.0
	ACO(D)	-3.0	-3.0
	ATO(D)	-5.0	-5.0
	DCO	-2.0	-2.0
	DTO	-2.0	-8.0
	DTT	-2.0	-8.0
	sATT	-2.0	-2.0
	sDTT	-2.0	-2.0
	sCTT	-2.0	-8.0
ATO(D)	ONS(D)	-6.0	-6.0
	OPSS	-6.0	-6.0
	OPSL	-3.0	-3.0
	ACO(D)	-4.2	-4.2
	ATT(D)	-5.0	-5.0
	ATO(D)	-6.0	-6.0
	DCO	-3.0	-3.0
	DTO	-3.0	-9.0
	DTT	-3.0	-9.0
	sATT	-3.0	-3.0
	sDTT	-3.0	-3.0
	sCTT	-3.0	-9.0
sATT	ONS(D)	-3.0	-3.0
	OPSS	-3.0	-3.0
	OPSL	0	0
	ACO(D)	-1.0	-1.0
	ATT(D)	-2.0	-2.0
			Page 3 of 5

Table 8-3 Conferencing Loss Plan (continued)			
Originating Circuit	**Connecting Circuit**	**Loss (dB) Tx Direction**	**Loss (dB) Rx Direction**
	ATO(D)	-3.0	-3.0
	DCO	0	0
	DTO	0	-6.0
	DTT	0	-6.0
	sATT	0	0
	sDTT	0	0
	sCTT	0	-6.0
sDTT	ONS(D)	-3.0	-3.0
	OPSS	-3.0	-3.0
	OPSL	0	0
	ACO(D)	-1.0	-1.0
	ATT(D)	-2.0	-2.0
	ATO(D)	-3.0	-3.0
	DCO	0	0
	DTO	0	-6.0
	DTT	0	-6.0
	sATT	0	0
	sDTT	0	0
	sCTT	-6.0	-6.0
sCTT	ONS(D)	-9.0	-3.0
	OPSS	-9.0	-3.0
	OPSL	-6.0	0
	ACO(D)	-7.0	-1.0
	ATT(D)	-8.0	-2.0
	ATO(D)	-9.0	-3.0
	DCO	-6.0	0
	DTO	-6.0	-6.0
	DTT	-6.0	-6.0
	sATT	-6.0	0
	sDTT	-6.0	0
	sCTT	-6.0	-6.0
DCO	ONS(D)	-3.0	-3.0
	OPSS	-3.0	-3.0
	OPSL	0	0
	ACO(D)	-1.0	-1.0
	ATT(D)	-2.0	-2.0
			Page 4 of 5

Table 8-3 Conferencing Loss Plan (continued)			
Originating Circuit	**Connecting Circuit**	**Loss (dB) Tx Direction**	**Loss (dB) Rx Direction**
	ATO(D)	-3.0	-3.0
	DCO	0	0
	DTO	0	-6.0
	DTT	0	-6.0
	sATT	0	0
	sDTT	0	0
	sCTT	0	-6.0
DTO	ONS(D)	-9.0	-3.0
	OPSS	-9.0	-3.0
	OPSL	-6.0	0
	ACO(D)	-7.0	-1.0
	ATT(D)	-8.0	-2.0
	ATO(D)	-9.0	-3.0
	DCO	-6.0	0
	DTO	-6.0	-6.0
	DTT	-6.0	-6.0
	sATT	-6.0	0
	sDTT	-6.0	0
	sCTT	-6.0	-6.0
DTT	ONS(D)	-9.0	-3.0
	OPSS	-9.0	-3.0
	OPSL	-6.0	0
	ACO(D)	-7.0	-1.0
	ATT(D)	-8.0	-2.0
	ATO(D)	-9.0	-3.0
	DCO	-6.0	0
	DTO	-6.0	-6.0
	DTT	-6.0	-6.0
	sATT	-6.0	0
	sDTT	-6.0	0
	sCTT	-6.0	-6.0
			Page 5 of 5

Tone Levels

8.4 All of the tones used by the PABX are provided by the Digital Signal Processor on the Main Control Card. The system provides eight tone channels, each of which can generate either a dual or single frequency tone. One channel is used to generate a signal which controls the frequency and amplitude of the ringing generator output. Three channels are used for DTMF tone generation, and the remaining four are used to generate the system call progress tones.

Most call progress and DTMF tones require that the receive gain pad in the line or trunk circuit interface codec be set to a specific value for the duration of the tone. The exception is the camp-on/override tone, which does not require any adjustment of the pads. The interface settings for the call progress tones are outlined in Table 8-4. The DTMF levels and interface settings are outlined in Table 8-5 and Table 8-4 respectively.

Table 8-4 Interface Level (dBm)

Circuit	Dial Tone	Reorder/ Busy	Ringback	Camp-On/Exec. Override	DTMF
ONS(D)	-13.75	-24.0	-19.0	-20	n/a
OPSS	-13.75	-24.0	-19.0	-20.0	n/a
OPSL	-13.75	-24.0	-19.0	-17.0 to -19.0	n/a
ACO(D)	-13.75	-18.0	-13.0	-14.0 to -20.0	-1.0
ATT(D)	-13.75	-18.0	-13.0	-17.0 to -20.0	-1.0
ATO(D)	-13.75	-18.0	-13.0	-17.0 to -20.0	-1.0
DCO	-13.75	-20.0	-15.0	-17.0	-1.0
DTO	-13.75	-20.0	-15.0	-17.0	-1.0
DTT	-13.75	-20.0	-15.0	-17.0	-1.0
sATT	-13.75	-18.0	-13.0	-17.0 to -19.0	-1.0
sDTT	-13.75	-20.0	-15.0	-17.0 to -19.0	-1.0
sCTT	-13.75	-20.0	-15.0	-17.0	-1.0

Table 8-5 DTMF Levels - Trunk Interface Into 600 Ohms

Nominal level - single frequency: -4.0 dBm
Nominal level - frequency pair : -1.0 dBm
Nominal twist : 0.0 dBm

9 Call Progress Tones and Ringing Cadences

This Part describes the different call progress tones and ringing cadences that are available to support PABX requirements.

North America

Tone Plan

9.1 Table 9-1 identifies the tones that are generated by the PABX in North America.

Table 9-1 North America Tone Generation Table

Tone	Frequency 1	Frequency 2	Level With No Interface Gain/Loss
Ringing	20 Hz	---	90 Vrms
Dial tone	350 Hz	440 Hz	-10.75 dBm0
Busy tone	480 Hz	620 Hz	-20.00 dBm0
Ringback tone	440 Hz	480 Hz	-15.00 dBm0
Miscellaneous tone	440 Hz	---	-17.00 dBm0
Modem connect tone	2025 Hz	---	-15.00 dBm0

Note: **1.** Ringing voltage is measured at source.

2. Dial Tone passes through a filter with 7 dB attenuation.

3. Busy Tone has +2 dB added at trunk interface for network connectivity.

4. Ringback Tone has +2 dB added at trunk interface for network connectivity.

5. Miscellaneous Tone (440 Hz at - 17.02 dBm0) is used for the maintenance test tone.

Ringer Cadencing

Table 9-2 identifies the ringer cadencing that is provided by the PABX in North America.

Table 9-2 North America Digital Bay Ringer Cadencing Discriminating Ring (Option 17) enabled and Discriminating Ring Always (Option 18) disabled			
ONS/OPS Internal (standard)	**ONS/OPS External (discriminating)**	***SUPERSET* Telephones Internal (standard)**	***SUPERSET* Telephones External (discriminating)**
.9 on 3.1 off	.350 on .200 off	1 on 3 off	.400 on .200 off
repeating	.350 on 3.1 off	repeating	.400 on 3.0 off
	repeating		repeating

Mexico

Tone Plan

9.2 Table 9-3 identifies the tones that are generated by the PABX in Mexico.

Table 9-3 Mexico Tone Generation Table

Tone	Frequency 1	Frequency 2	Level With No Interface Gain/Loss
Ringing	25 Hz	---	70 Vrms
Dial tone	440 Hz	---	-10.00 dBm0
Busy tone	440 Hz	---	-10.00 dBm0
Ringback tone	440 Hz	---	-10.00 dBm0
Miscellaneous tone	440 Hz	---	-10.00 dBm0
Modem connect tone	2025 Hz	---	-15.00 dBm0

Note: **1.** Ringing voltage is measured at source.

2. Dial Tone passes through a filter with 7 dB attenuation.

3. Busy Tone has +2 dB added at trunk interface for network connectivity.

4. Ringback Tone has +2 dB added at trunk interface for network connectivity.

5. Miscellaneous Tone (440 Hz at - 17.02 dBm0) is used for the maintenance test tone.

Ringer Cadencing

Table 9-4 identifies the ringer cadencing that is provided by the PABX in Mexico.

Table 9-4 Mexico Digital Bay Ringer Cadencing

Discriminating Ring (Option 17) enabled and Discriminating Ring Always (Option 18) disabled

ONS/OPS Internal (standard)	ONS/OPS External (discriminating)	*SUPERSET* Telephones Internal (standard)	*SUPERSET* Telephones External (discriminating)
.9 on 4.1 off	.350 on .200 off	1 on 4 off	.400 on .200 off
repeating	.350 on 4.1 off	repeating	.400 on 4.0 off
	repeating		repeating

Hong Kong / Taiwan

Tone Plan

9.3 Table 9-5 identifies the tones that are generated by the PABX in Hong Kong and Taiwan.

Table 9-5 Hong Kong / Taiwan Tone Generation Table

Tone	Frequency 1	Frequency 2	Level With No Interface Gain/Loss
Ringing	20 Hz	---	90 Vrms
Dial tone	380 Hz	---	-10.00 dBm0
Busy tone	480 Hz	620 Hz	-10.00 dBm0
Ringback tone	440 Hz	480 Hz	-10.00 dBm0
Miscellaneous tone	440 Hz	---	-10.00 dBm0
Modem connect tone	2025 Hz	---	-10.00 dBm0

Note:
1. Ringing voltage is measured at source.
2. Dial Tone passes through a filter with 7 dB attenuation.
3. Busy Tone has +2 dB added at trunk interface for network connectivity.
4. Ringback Tone has +2 dB added at trunk interface for network connectivity.
5. Miscellaneous Tone (440 Hz at - 17.02 dBm0) is used for the maintenance test tone.

Ringer Cadencing

Table 9-6 identifies the ringer cadencing that is provided by the PABX in Hong Kong and Taiwan.

Table 9-6 Hong Kong / Taiwan Digital Bay Ringer Cadencing

Discriminating Ring (Option 17) enabled and Discriminating Ring Always (Option 18) disabled

ONS/OPS Internal (standard)	ONS/OPS External (discriminating)	*SUPERSET* Telephones Internal (standard)	*SUPERSET* Telephones External (discriminating)
.9 on 3.1 off	.350 on .200 off	1 on 3 off	.400 on .200 off
repeating	.350 on 3.1 off	repeating	.400 on 3.0 off
	repeating		repeating

Malaysia

Tone Plan

9.4 Table 9-7 identifies the tones that are generated by the PABX in Malaysia.

Table 9-7 Malaysia Tone Generation Table

Tone	Frequency 1	Frequency 2	Level With No Interface Gain/Loss
Ringing	25 Hz	---	70 Vrms
Dial tone	350 Hz	440 Hz	-9.00 dBm0
Busy tone	480 Hz	620 Hz	-12.00 dBm0
Ringback tone	440 Hz	480 Hz	-12.00 dBm0
Miscellaneous tone	440 Hz	---	-12.00 dBm0
Modem connect tone	2025 Hz	---	-15.00 dBm0

Note:
1. Ringing voltage is measured at source.
2. Dial Tone passes through a filter with 7 dB attenuation.
3. Busy Tone has +2 dB added at trunk interface for network connectivity.
4. Ringback Tone has +2 dB added at trunk interface for network connectivity.
5. Miscellaneous Tone (440 Hz at - 17.02 dBm0) is used for the maintenance test tone.

Ringer Cadencing

Table 9-8 andTable 9-9 identify the ringer cadencing that is provided by the PABX in Malaysia.

Table 9-8 Malaysia Digital Bay Ringer Cadencing

Discriminating Ring (Option 17) enabled and Discriminating Ring Always (Option 18) enabled

ONS/OPS Internal (standard)	ONS/OPS External (discriminating)	*SUPERSET* Telephones Internal (standard)	*SUPERSET* Telephones External (discriminating)
.350 on .200 off	.350 on .200 off	.400 on .200 off	.400 on .200 off
.350 on 2.1 off	.350 on 2.1 off	.400 on 2.0 off	.400 on 2.0 off
repeating	repeating	repeating	repeating

Table 9-9 Malaysia Digital Bay Ringer Cadencing

Discriminating Ring (Option 17) enabled and Discriminating Ring Always (Option 18) disabled

ONS/OPS Internal (standard)	ONS/OPS External (discriminating)	*SUPERSET* Telephones Internal (standard)	*SUPERSET* Telephones External (discriminating)
.9 on 5.1 off	.350 on .200 off	1 on 5 off	.400 on .200 off
repeating	.350 on 2.1 off	repeating	.400 on 2.0 off
	repeating		repeating

10 Traffic Considerations

General

10.1 This part details traffic limitations. Information includes:

- Busy Hour Call Attempt (BHCA)
- System Traffic Capacity
- Grade of Service
- Receiver Provisioning
- Trunk Distribution.

Traffic Limitations

10.2 Traffic capacities are specified on a per line basis in terms of calls per hour, erlangs, and CCS.

A basic system consists of 80 lines and 16 trunks. This configuration has been specified to meet the following heavy traffic characteristics:

Busy Hour Call Attempts	System (BHCA) Mean	99.9 % Peak
Per Second	0.090	0.186
Per Hour	325	671

Bothway Traffic Capacity		
Calls/Hour	Erlang	CCS
426	14.5	522.0

Typical configured system quantities as per ATT 48002 are:

Light - 10 %		Medium - 50 %		Heavy - 90 %	
Lines	Trunks	Lines	Trunks	Lines	Trunks
75	8	75	12	75	19
50	6	50	10	50	15
25	5	25	7	25	10

Grade of Service

10.3 The Grade of Service (GOS) in terms of blocking is outlined in Table 10-1.

Table 10-1 Grade Of Service (GOS)

Link/Resource Blocking	Blocking Probability
Link Blocking:	
Peripheral to Network	< 0.1%
Network to Network	0.0%
Resource Blocking	
Software	< 0.01%
DTMF Receivers, Trunks	provisioning dependent

Receiver Provisioning

10.4 The number of DTMF receivers required to be installed in the PABX depends on various factors, such as the number of lines and trunks installed, the amount of traffic flow estimated for the system and the desired grade of service. Receivers are not required for DNIC devices. To determine the quantity of DTMF receivers required, the following assumptions were made:

- Average receiver holding time for intercom call is 6 seconds
- Average receiver holding time for a trunk call is 17.4 seconds
- Holding time for receivers is exponential
- Call originations are Poisson distribution
- Call holding times are exponential.

The MCC II contains seven receivers. If additional receivers are required, receiver modules can be added to a Universal Card. Each receiver module contains four receivers.

For a given load (heavy, medium and light traffic), the minimum number of required receivers was determined for the following grades of service (ABSBH - Average busy Season Busy Hour):

- **ABSBH = 99.0:** 99.0 % of all receiver requests serviced within 3 seconds
- **ABSBH = 99.99:** 99.99 % of all receiver requests serviced within 3 seconds

The following calculations are used to generate Table 10-2, Table 10-3, and Table 10-4:

Receiver Holding Time (h) = 6 x (% intercom traffic) + 17.4 x (% trunk traffic) Seconds

Receiver Traffic (A) =

$$\frac{\text{(Originating Calls per hour) x (Receiver holding time (sec))}}{3600} \text{ Erlangs}$$

$$\text{Multiple of holding time } (t) = \frac{3\ (s)}{h\ (s)}$$

Probability of delay greater than t (P(>t)) = 1 - ABSBH /100

Using Erlang C formula delay curves and knowing t, A, P(>t), the minimum number of receivers can be obtained.

Because the MCC II contains seven DTMF receivers, an SX-200 ML PABX configured at 96 ports (72 ONS and 24 digital trunks) will not require additional trunks to support heavy traffic.

Table 10-2 Heavy Traffic

No. of Lines	CCS/ Line	Total CCS	In CCS	Out CCS	Intra CCS	In Calls	Out Calls	Intra Calls	Orig Calls	In Trks	Out Trks	2 Way Trks	Receivers Required for ABSBH = 99.0%	Receivers Required for ABSBH = 99.99%
20	6.90	138	61	57	20	40	43	23	66	7	6	9	3	4
40	6.28	251	108	100	43	62	74	49	123	9	8	13	4	6
60	5.94	356	150	140	67	85	103	77	180	11	9	16	4	6
80	5.77	462	190	178	93	108	132	106	238	12	11	19	5	7
100	5.67	567	230	216	120	131	160	136	296	14	13	22	5	8

Table 10-3 Medium Traffic

No. of Lines	CCS/ Line	Total CCS	In CCS	Out CCS	Intra CCS	In Calls	Out Calls	Intra Calls	Orig Calls	In Trks	Out Trks	2 Way Trks	Receivers Required for ABSBH = 99.0%	Receivers Required for ABSBH = 99.99%
20	3.90	78	34	32	12	22	24	14	38	5	5	7	3	4
40	3.42	137	59	55	23	34	40	27	67	6	5	9	3	4
60	3.23	194	81	76	37	46	56	42	98	7	6	11	3	5
80	3.14	251	103	97	51	59	72	58	130	8	8	13	4	6
100	3.09	309	125	118	66	71	87	74	161	9	8	14	4	6

Table 10-4 Light Traffic

No. of Lines	CCS/ Line	Total CCS	In CCS	Out CCS	Intra CCS	In Calls	Out Calls	Intra Calls	Orig Calls	In Trks	Out Trks	2 Way Trks	Receivers Required for ABSBH = 99.0%	Receivers Required for ABSBH = 99.99%
20	1.41	28	34	12	4	7	8	4	12	4	3	5	2	3
40	1.41	56	59	23	10	14	17	11	28	4	4	6	2	4
60	1.41	85	36	33	16	20	24	18	42	5	4	7	3	4
80	1.41	113	46	44	23	26	32	26	58	6	5	8	3	4
100	1.41	141	57	54	30	33	40	34	74	6	5	9	3	5

11 System Characteristics

General

11.1 This section describes the environmental, electrical, and operational characteristics.

Environmental Conditions

11.2 The systems are designed to operate within the environmental conditions outlined in Table 11-1.

Table 11-1 System Environmental Operating Conditions

Specification	Range
Temperature - Cabinet	0° C to 40° C (32° F to 104° F)
- Console	0° C to 30° C (32° F to 86° F)
Relative humidity	20% to 80% noncondensing
Acoustic noise	The system radiates less than 50 dB SPL, "A" weighted, measured 1524 mm (60 in.) from the center of the cabinet.
Maximum Altitude	4000 metres

Heat Dissipation

11.3 A fully configured *SX-200* ML PABX will dissipate heat at approximately 500 BTU/hr.

Shipping and Storage

11.4 The equipment is designed to withstand shipping by truck, rail, air, or sea without damage when packaged in conventional shipping containers of the manufacturer. The range of environmental conditions that the equipment is capable of withstanding in storage is shown in Table 11-2.

Table 11-2 Storage Conditions

Specification	Range
Temperature range	-20° C to 66° C (-4° F to 150° F) for the system -20° C to 60° C (-4° F to 140° F) for the console
Relative humidity	5% to 95% RH at 18° C (64.4° F) non-condensing 10% to 70% RH for the console

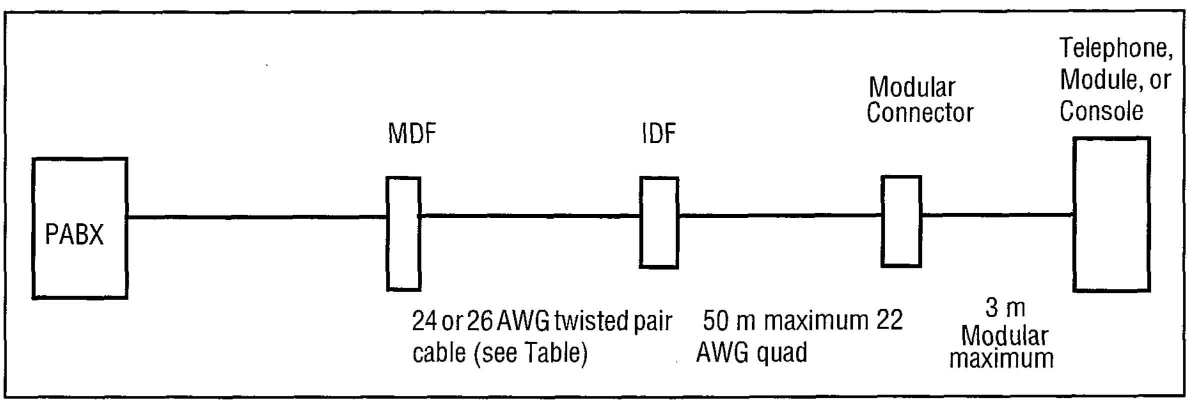
Page 1 of 2

Table 11-2 Storage Conditions (continued)	
Specification	**Range**
Vibration	0.5 g (4.903 m/s^2) (sinusoidal) 5 to 100 Hz 1.5 g (14.7 m/s^2) (sinusoidal) 100 to 500 Hz
Shock	Up to 75 cm (30 in.) drop depending upon package
Mechanical Shock	Up to 61 cm (24 in.) drop on any face or corner
Low pressure	87 mm Hg 15,152 m (50,000 ft)
Temperature shock	-40° C to 21° C (-40° F to 70° F) in 5 minutes 66° C to 21° C (150° F to 70° F) in 5 minutes
	Page 2 of 2

Loop Lengths and Cable Lengths

11.5 The following rules for loop lengths between the Digital Line Card within the PABX and the *SUPERSET 401+, SUPERSET 410, SUPERSET 420,* and *SUPERSET 430* telephones, or the *SUPERCONSOLE 1000* Attendant Console must be followed for proper operation of the device:

Maximum loop length (twisted pair) 24 or 26 AWG	see Table
Maximum length of quad cable (22 AWG)	50 m (160 ft)
Modular Line Cord	3 m (10 ft)



Peripheral Device	Maximum Loop Length	
	Without Bridge Tap	With Bridge Tap *
SUPERSET 401+ *SUPERSET 410*	1000 m	n/a
SUPERSET 420 *SUPERSET 430* Dataset 1101 *SUPERCONSOLE 1000* Attendant Console	1000 m	1000 m
Datasets 1103, 2103, and *MILINK* Data Module	2000 m	1000 m

Note: Peripheral devices connected to Digital Line Cards which have a PN 9109-012-001-NA can not have bridge taps. Digital Line Cards with PN 9109-012-000-SA may have bridge taps.

Loop lengths for various cards are given in Table 11-3:

Table 11-3 Maximum Loop Lengths for Cards

Card Type	Wire Gauge (AWG)	Loop Length
DNIC line card (Digital set)	24	1006 m 3300 ft
	26	1006 m 3300 ft
DNIC Console	24	1006 m 3300 ft
	26	1006 m 3300 ft
ONS Card	22	3560 m 11700 ft
	24	2250 m 7400 ft
	26	1400 m 4600 ft
OPS Card	22	18290 m 60000 ft
	24	11520 m 37800 ft
	26	7225 m 23700 ft
COV Card	22	2000 m 6600 ft
	24	1500 m 5000 ft
	26	1000 m 3300 ft
E&M Trunk	22	2715 m 8900 ft
	24	1708 m 5600 ft
	26	1068 m 3500 ft
DID Trunk Card - CO Trunk resistance	na	2240 ohms
LS/GS Trunk Card - CO Trunk resistance	na	1600 ohms
T1 Trunk - see Note	na	na

Note: These are cable lengths, not loop lengths for 22 gauge (AWG) wire. Set DIP switches on T1 Trunk card for correct equalization depending on cable length between the T1 Trunk and the Channel Service Unit (CSU).

Cable Length	Switch Setting
0 - 45.8 m (0 - 150 ft)	S1 only closed
45.8 - 137.3 m (150 - 450 ft)	S2, S3, S4 closed
137.3 - 200.5 m (450 - 655 ft)	S5, S6, S7 closed

12 Bay Power Supply

12.1 The Bay Power Supply (BPS) is card-mounted and is located in the upper right slot of the cabinet. The electrical power characteristics are summarized in Table 12-1.

Table 12-1 Electrical Input Power Characteristics

	Input	Frequency	Minimum Holdover Time	Input Current
9109-008-000-SA Bay Power Supply	102 Vac to 135 Vac	47 Hz to 63 Hz	40 ms at 120 Vac or 20 ms at 102 Vac (delivering full rated load)	Maximum of: 2.0 Amps at 120 Vac

Electrical Characteristics

12.2 Power is derived from a commercial ac source or an Uninterruptible Power Supply (UPS).

Table 12-2 Bay Power Supply Specifications

	Commercial power	Inverter power	Current
Low range	102 Vrms to 135 Vrms.	102 Vrms to 135 Vrms.	2.0 A rms max.
Frequency	47 Hz to 63 Hz.	47 Hz to 63 Hz.	-
Waveform	Sinusoidal, 5% THD max.	Quasi-square wave, 0.71 duty cycle.	-

Bay Power Supply

12.3 The BPS connects to the backplane through a card-edge connector at the rear of the card; also at the rear is an IEC receptacle which connects to a line cord from the system ac distribution (see Figure 12-1). BPS dimensions are:

Width	5.1 cm	(2.0 in.)
Height	15.7 cm	(6.2 in.)
Depth	36.8 cm	(14.5 in.)

Bay Power Supply output voltages and test point voltages are shown in Table 12-2.

Controls and Indicators

The ON/OFF switch is mounted on the front of the BPS and is used to turn the power on or off to the unit. The upper LED indicates that the BPS is operating, and the lower LED is ON when the ringing amplifier is producing power (flashing in cadence with it).

Input and Output Protection

The input to the converter is protected by a fuse,. and by low voltage protection which shuts off the converter if the input voltage falls below the specified minimum. The converter will not be re-enabled until the input voltage returns to the specified minimum. The input also includes protection which limits the peak inrush current to 20 A.

Each output is protected against short circuits, overloads, and overvoltage. The overload/short circuit protection is self-resetting.

Table 12-3 Bay Power Supply Test Point Voltages

Voltage	Range
+ 5 Vdc	+ 5.07 to +5.23
+12 Vdc	+10.8 to +13.2
-12 Vdc	-13.2 to -10.8
- 5 Vdc	-5.5 to -4.5
-28 Vdc	-30.8 to -23.8
-48 Vdc	-53.76 to -40.8
90 Vac	63.0 to 99.0

Power Fail Sense

The converter has a single alarm signal, PFS (power fail sense), which is driven low when the incoming ac falls below its minimum specified value. At this point there will be approximately 10 ms before the outputs fall out of regulation.

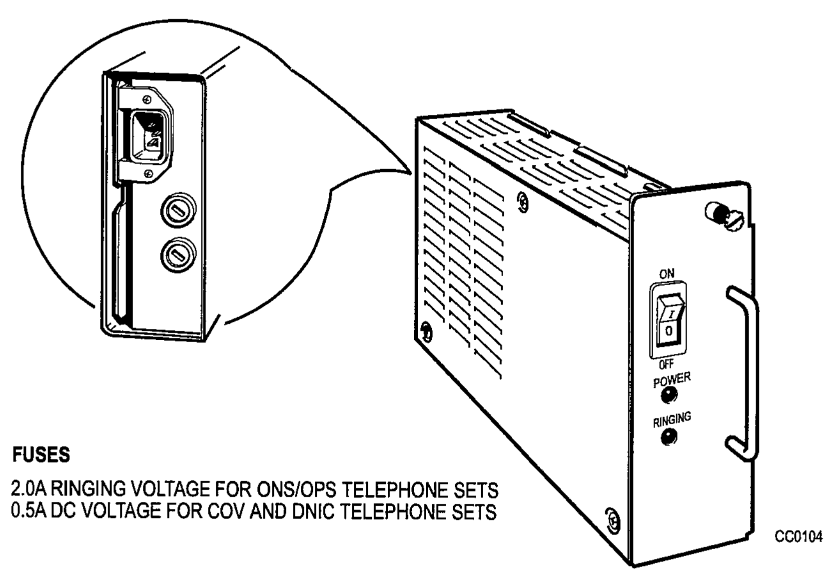


Figure 12-1 Bay Power Supply

13 Reserve Power Supply

The reserve power supply is a stand-alone Uninterruptible Power Supply (UPS) comprising a battery pack, a charger, and an inverter; it is not manufactured by Mitel Corporation. The UPS backup time is dependent upon the unit selected and the capacity of the batteries provided. The unit must meet the specifications provided below.

Please note that compliance to electrical, installation, and building codes is the responsibility of the purchaser of the equipment. Consult local municipal and electrical utility authorities before proceeding with the installation of equipment.

The UPS should be a true Uninterruptible Power Supply which always supplies the output load from its inverter. The UPS must include a reverse transfer switch to automatically bypass itself if it fails. The UPS must be capable of driving rectifier capacitor loads.

Rating	1.5 kVA minimum
Input Voltage	120 Vrms +10%,-15%, 60 Hz 15%
Output Voltage	120 Vrms +10%,-15%, 60 Hz 5%
Output Waveform	Sinewave or Quasi square wave (not square wave)
Transfer time	Less than 30 ms (includes fail detection and transfer time)
Output Receptacle	NEMA 15 A 3-pin grounded
Holdup/Recharge Times	Per customer requirements.

An Uninterruptible Power Supply (UPS) can have an external connection (from an internal relay) which provides a closed contact to remotely indicate status or condition. Conditions which may be indicated include:

- an ALARM condition is present within the UPS
- the UPS is operating from its batteries (probably because commercial ac power has been interrupted).

The relay contact may be connected to a remote alarm or to a “Contact Monitor” line circuit to promptly indicate the condition.

Refer to the manufacturer’s installation manual, which describes conditions that are indicated. Refer to the *Features Description* Practice, for a description of “Contact Monitor” line circuit operation.

14 System Fail Transfer

SFT Operation

14.1 In the event of a major alarm condition on the *SX-200* ML PABX, the MCC II provides Control (relay contact) and power (-48V) signals to an external system fail transfer unit. The following conditions will cause system fail transfer:

- Commercial power failure (if no reserve power supply is used)
- Common control failure.

The system fail transfer relays located on the System Fail Transfer (SFT) Card will connect Central Office (CO) trunks to selected station lines. Calls in progress when SFT occurs will be dropped; however, calls made while in SFT mode will not be dropped when the system returns to normal operation, but will terminate normally at the end of the call.

PABX features are not available while SFT is in effect.

NOTES