MITEL

SX-200 IP Communications Platform

General Information Guide
Release 5.0 UR1

SX-200 IP Communications Platform (ICP)
General Information Guide
Release 5.0 UR1
October 2009

About this document

This guide provides an overview of the SX-200® IP Communications Platform (ICP). It provides details on the features, applications, and services available on the SX-200 ICP. It includes a description of the major call management facilities, the peripheral devices that can be connected to the system, and the hardware configurations that allow you to tailor the system to your needs.

The topics covered in this guide are:

- Overview of SX-200 ICP, its hardware, embedded applications and features
- Supporting applications
- Maintenance

Audience

This guide is intended for:

- End customers
- Sales executives
- Consultants
- Industry analysts
- Media analysts
- Sales engineers
- System engineers

About the SX-200 ICP documentation

The following guides provide information about the Mitel® SX-200 ICP:

- SX-200 ICP Technician's Handbook provides instructions to install, upgrade, maintain, and troubleshoot the Mitel SX-200 ICP.
- SX-200 ICP Technical Documentation in HTML and Folio (NFO)
- SX-200 Safety Instructions
- IMAT Online Help can be printed and provides detail instructions on configuring and managing the SX-200 ICP.

What's new?

Release 5.0 UR1

- The Mailbox Lockout feature enables a user's mailbox to be locked after three unsuccessful login attempts.
- A new embedded voice mail language, English Overlaid, provides numeric prompts ("Press 7 to play...") in place of the default alphabetic prompts ("Press P to play...").

Release 5.0

- Support for Mitel 5540 IP Consoles
- Support for AX controllers

Release 4.0 UR5

- Support for Mitel 5304 IP Phones
- Support for Mitel 5312 and 5324 IP Phones

Release 4.0

New Hardware Features

- New CX controller variation. The original CX controller is now the CXi. The new "switchless" CX is ideal for businesses with existing Layer 2 switches and for those that require more than 16 ports.
- CX and CXi controllers now ship with 512 MB internal CompactFlash.
- New ASU II supports up to 32 ONS phones or up to 8 LS trunks depending on configuration and peripheral cards.
- Support for Quad CIM modules on new CX/CXi controllers (for ASU II support only—Bay functionality coming soon.)
- Support for Mitel 5330 and 5340 IP Phones
- Support for the Applications Processor Card (APC) — a PC on a compact card installed in the CX/CXi controller to host the Managed Application Server.
- Support for Mitel Gigabit Ethernet (GigE) and Wireless LAN (WLAN) Phone Stands

New Software Features

- IP phone capacity for the CX/CXi increased to 100.
- Number of ARS entries increased to 350.
- The Ring Groups feature provides the ability to ring all members of a group simultaneously.
- The Direct Inward Dialing Server application allows Hotels to automatically assign direct dial numbers for their guests.
- New CDE Form 9 subform (Review - Call Forward) provides an onboard Call Forwarding profile editor. (Support for MyAdministrator application has been discontinued.)

- Enhanced Hotel/Motel Subattendant features available using the Mitel 5340 IP Phone.

Release 3.1

- SX-200 ICP MX controller supports up to 238 IP Phones.
- Headset users can now place calls on Hard Hold while Automatic Line Selection is selected.
- Loop Start Trunk Measurement Tool enhancements.
- Telephones now support Set Page, Group Page, and All Set Page across tie/IP trunks.
- Single button transfer to voice mail while on Consultation Hold.
- The 5485 Paging Unit is now accessible to ONS telephones for Paging - PA.

Release 3.0

New Hardware Features

- Mitel Line Interface Module (LIM): Provides users of 5220 and 5224 Dual Mode IP Phones with the ability to make and receive calls on an analog line. The LIM replaces the Mitel 5425 Access Module.
- Support for Application Processor Card (CX)
- Support for Mitel 5212 and 5224 Dual Mode IP Phones
- Support for Telematrix 3000IP Phone

  Note: Although the 3000IP appears in CDE, it will not be available for programming until R3.1.
- SpectraLink NetLink h340 Wireless Telephone

New Software Features

- The Calling Party Number Substitution feature provides calling party identification for outgoing calls for purposes of network identification and call back.
- The Direct Inward Dialing Translation feature enables calls received by dial-in trunks to be routed based on Day, Night1 and Night2 service.
- Support for Embedded PRI with Dual T1/E1 Framer (MX) or T1/E1 Combo (CX).
- New options available on Form 42 (Link Descriptor Options): Protocol, Protocol Variant, Network/User, Unknown Numbering Plan, Bearer capability Voice, CLIR Voice, and Invert D Channel.
- New L2 Switch settings (CX): IGMP Snooping; Rapid Spanning Tree; STP Bridge Priority.
- New System IP settings: DiffServe Code Point; VoiceVLAN ID; Voice VLAN Priority.
- The CX now supports 802.1 p/Q prioritization for voice traffic on the default VLAN (1), or on a separate Voice VLAN. Program matching VLAN ID and priority values on Form 47, Subform 1 (System IP), the DHCP server, and the external L2 switches.
- Line key enhancements:
  - -Maximum number of line appearances is increased from 32 to 64 for DTS, CO,

Logical, Multicall and Key line keys.

- -Automatic line selection is programmable using either the Line Select or Line Preference feature.
- -Call Logging is available for multiple appearances of DTS and CO line keys.
- -Forward Campon feature is now available if the forwarding destination is a Speed Call Key or System Abbreviated Dial Number.
- -Split and Hold features can now be used during conferences with multiple DTS or CO lines (two or more LS trunks).
- -If a user on a line appearance of a CO trunk exits a conference initiated by a user with an ONS set, the conference will continue without interruption.

- Subattendant functionality extended to Mitel 5020, 5220, 5220 Dual Mode, and 5224 Dual Mode IP Phones.
- New Option 16 (IP Set Voice Encryption) available on Form 04 (System Options/System Timers).
- DTMF tones can now be transmitted to 5485 IP Pager Unit for PA Paging.
- 6042 Managed VPN now supports Traffic Shaping to ensure QoS for voice calls. With this feature, smaller sites (fewer than 24 IP Phones) can be interconnected using relatively low-bandwidth cable or DSL links to the Internet. For more information on interconnecting sites over IP trunks, Networking Mitel IP-PBXs.
- Support for additional LS Trunk Line Length and Impedance values in Subform 13 (Trunk Circuit Descriptors Options). Also see Attenuation Levels for Short and Long CO Trunks.
- User mailboxes can now start with a 9. To prevent conflict with the Phonebook feature (Directory Dial by Name), an inter-digit timer is started every time the first digit entered is 9. If no other digits are entered before the timer expires, the user is transferred to the Phonebook.
- The time out behavior (disconnect or forward to operator) can now be selected for each Multi-level Auto Attendant menu node mailbox.
- Mitel Your Assistant now supports the YA Softphone, an IP-based software phone, and YA Lite, a free version of the product.
- The SX-200 ICP now supports disconnect tones for 11 Latin American countries in addition to the default disconnect tone for Canada/USA. The disconnect tone files are selected via System Option 138 (Country Variant For Disconnect Tone Control) in Form 04 (System Options/System Timers).
- MTCE command to reset the Application Processor Card.

Release 2.3

- Asian and Latin American tone plan support

Release 2.2

- Line Interface Module support
- Waiting Lamp indication for queued calls (Mitel Dual Mode 5215 and 5220 IP Phones)

- Analog Main Board (AMB) Hardware up-rev to Rev-C (no new functionality added)
- Echo DSP Filter enhancements
- IP Paging Unit support enhancement
- BCC3 logs and robustness support

Release 2.1

- NetLink e340 and NetLink i640 Wireless Telephones (to replace the discontinued Symbol MiNET Wireless phones)
- Stratum 3 Clock Module (CX)
- support for Category 3 Ethernet cable (CX)
- Mitel Your Assistant
- Line Characterization Tool - milliwatt tone loopback
- Message Flash Notification for Incoming Calls (Mitel 5207 IP Phone)
- support for SMTP Authentication
- embedded voice mail now supports notification on every new message regardless of whether or not notification for previous messages has already been answered.

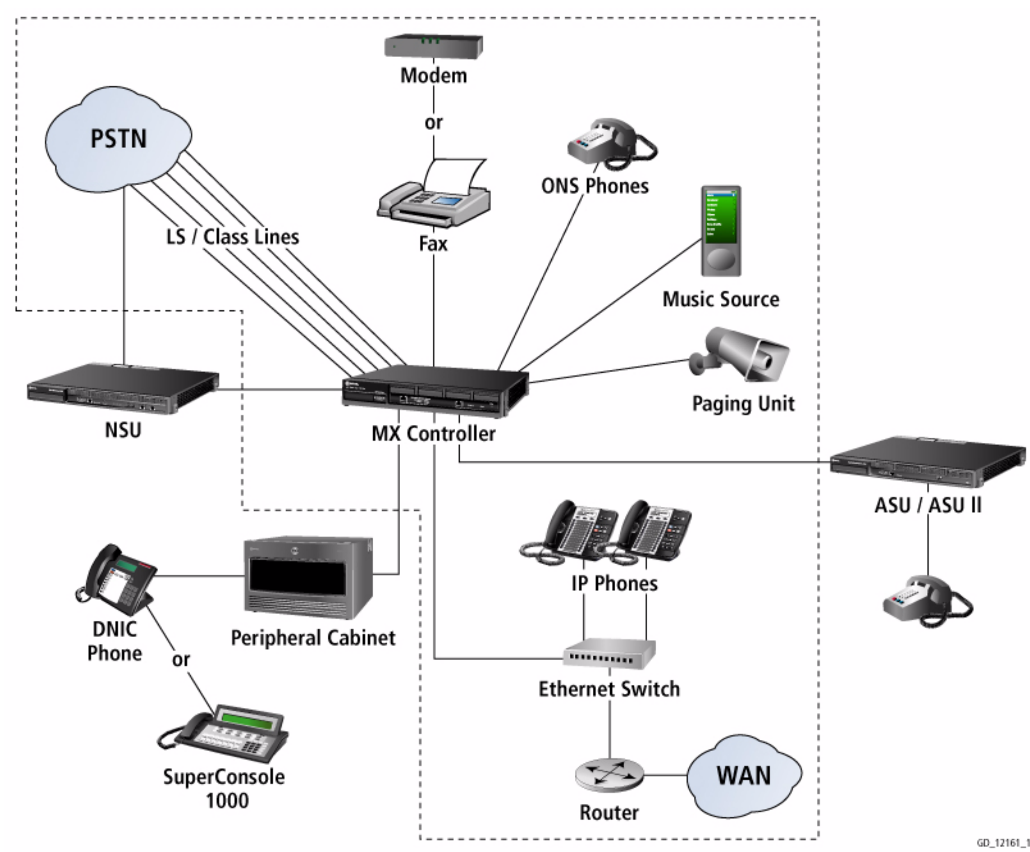
Overview

Mitel SX-200 IP Communications Platform

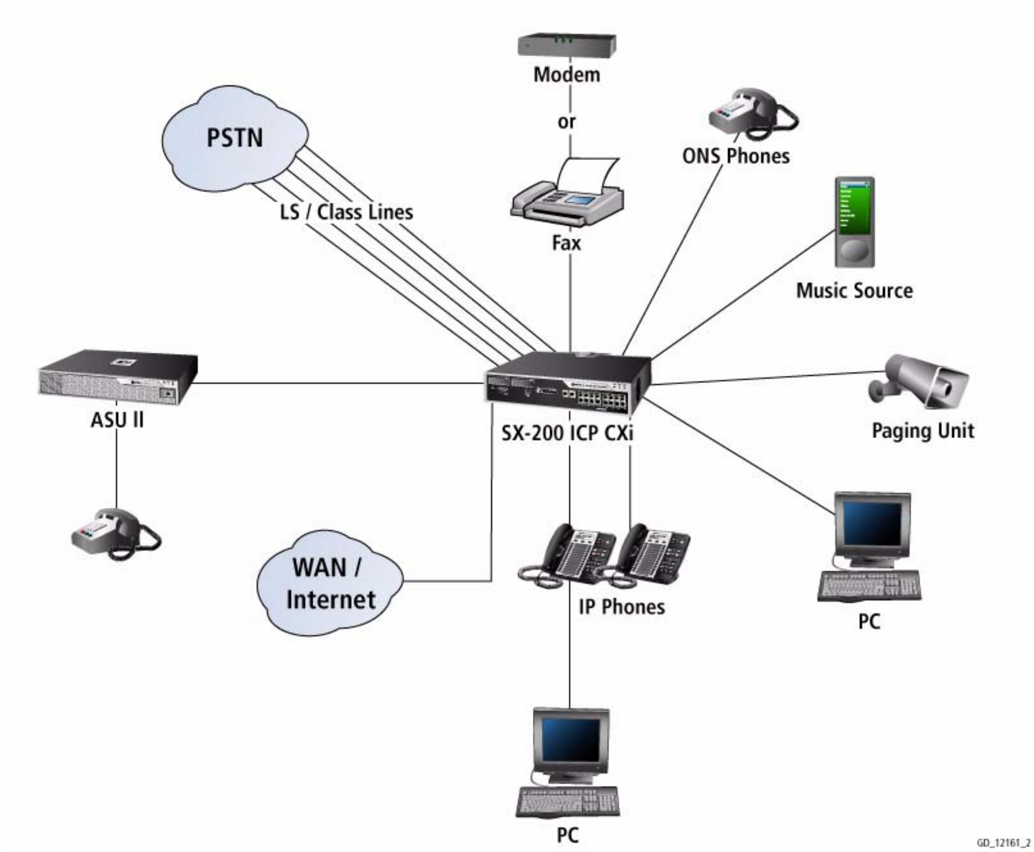
The Mitel SX-200 IP Communications Platform (ICP) delivers superior voice capabilities and features via a low-cost key telephone system that offers Voice over IP (VoIP), LS/CLASS, ONS/CLASS and DNIC solutions. Tailored for small enterprises, the Mitel SX-200 ICP offers the following platforms:

- SX-200 ICP MX
- SX-200 ICP CXi
- SX-200 ICP CX (no internal Layer 2 switch)
- SX-200 ICP AX

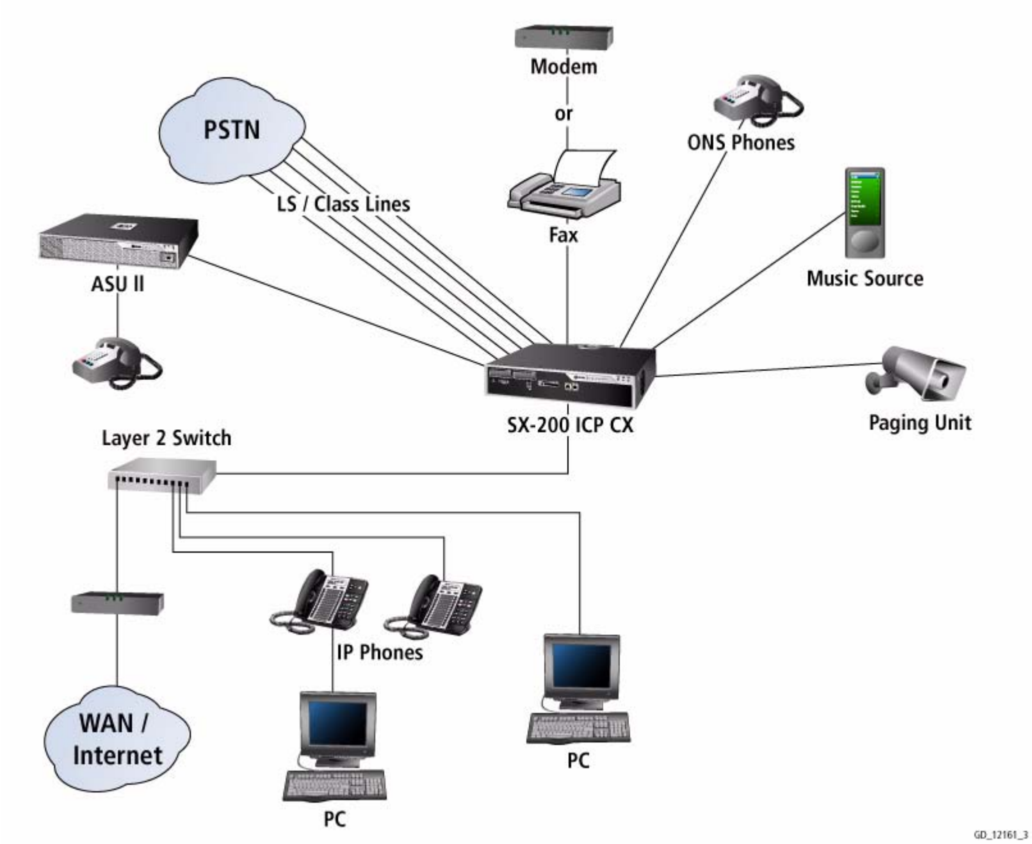
The following diagram illustrates a sample SX-200 MX ICP platform.



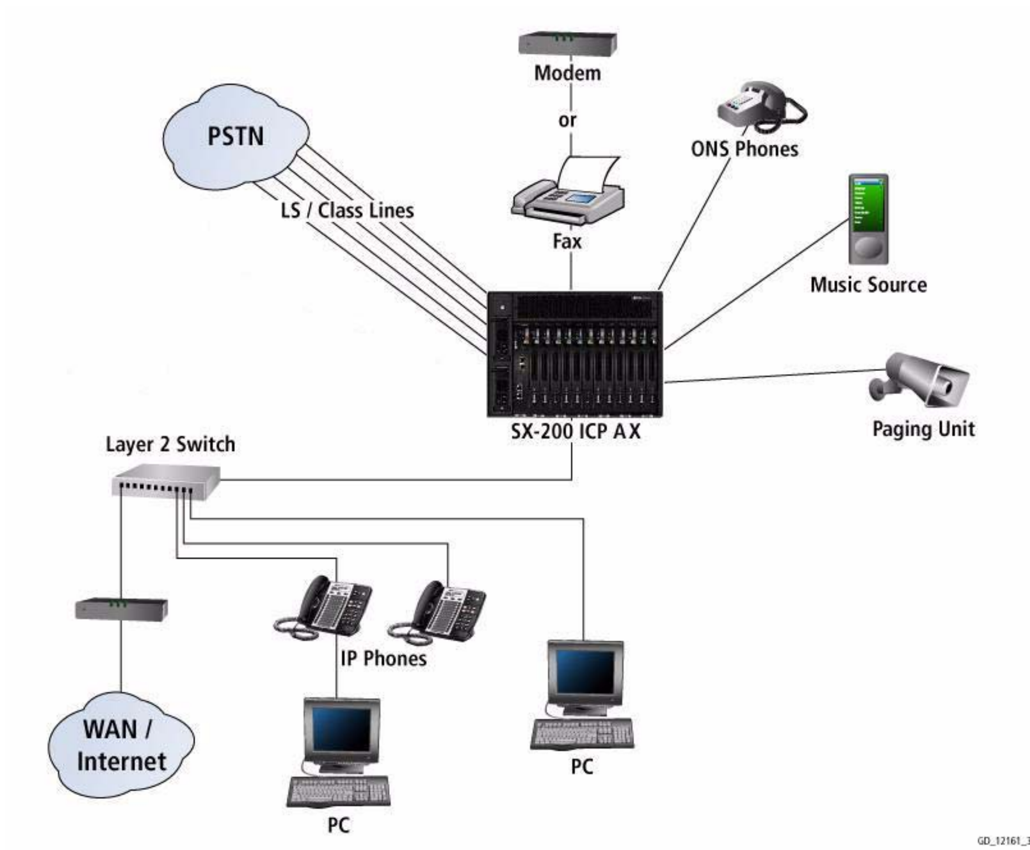
The following diagram illustrates a sample SX-200 CXi ICP platform.



The following diagram illustrates a sample SX-200 CX ICP platform.



The following diagram illustrates a sample SX-200 AX ICP platform.



System

This section describes the SX-200 ICP system architecture and its components.

For more information, refer to:

- “System architecture” on page 11
- “SX-200 ICP Controller” on page 12
- “Network Services Units” on page 23
- “Analog Services Unit” on page 24
- “Peripheral Cabinets” on page 26
- “Peripheral Interface Cards and Modules” on page 27
- “Desktop Devices” on page 29
- “Digital Services Cards and Modules” on page 29
- “Network” on page 32
- “Directed Data I/O” on page 37

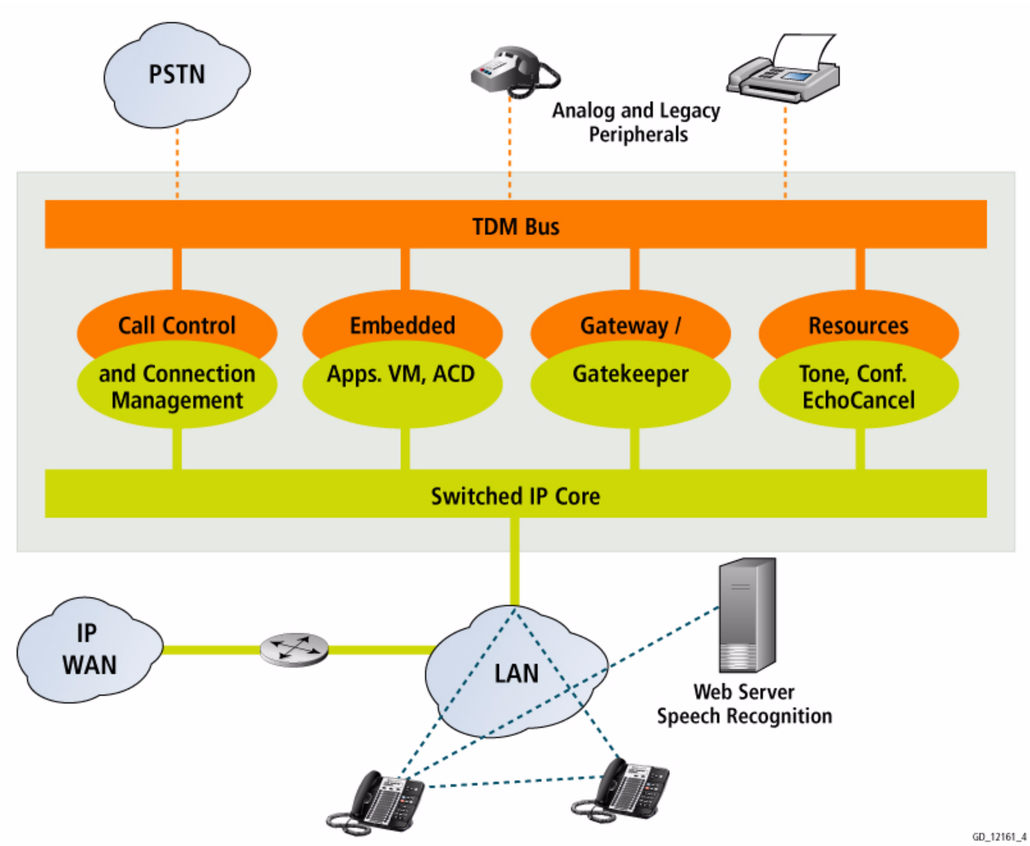
System architecture

The SX-200 ICP is built upon Mitel Data Integrated Voice Applications™ architecture delivering sophisticated call management, applications and desktop solutions for businesses. Mitel delivers a highly scalable, resilient, robust call control that fully utilizes the power of IP while fully supporting the traditional TDM based telephony for legacy devices and PSTN connectivity.

Mitel’s architecture uses the IP network to connect IP telephony devices and provides a supplementary TDM (Time Division Multiplexing) subsystem to switch calls between traditional telephone devices. The SX-200 ICP has the advantage of being able to optimally switch all types of traffic, IP or TDM. The SX-200 ICP provides native call setup, tear down, and signaling between Ethernet IP connected telephones. For traditional telephony, such as POTS and PSTN trunks, call handling is also handled natively by the SX-200 ICP via a conventional TDM circuit-switched subsystem.

This ability to use two different switching techniques simultaneously means that:

- All traffic is switched with minimum conversion between packet and traditional telephony to provide optimum voice quality in all call scenarios.
- Embedded gateway functionality is only required between the IP and non-IP networks optimizing the use of system resources.
- Migration from traditional PBX to IP telephony is seamless and efficient.



SX-200 ICP Controller

The SX-200 ICP Controller provides voice, signaling, central processing, and communications resources for the system. There are four controllers available in Release 5.0:

- CXi Controller
- CX Controller
- AX Controller
- MX Controller

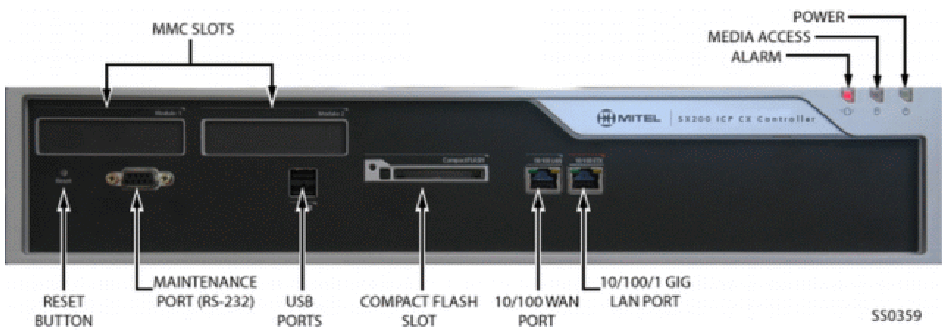
CX/CXi Controllers

CX/CXi controllers have the processing, memory, mass storage, power, and input/output capabilities to support up to 100 IP phones or 150 ONS/DNIC phones, or a combination of both. The original CX controller has been renamed as the CXi. The new CX controller, designed without a Layer 2 switch, is ideal for businesses with existing Layer 2 switches, or for those that require more than 16 ports.

Front panel (CXi)

- RS-232 port (DB-9 connector) for Maintenance terminal
- 10/100Base-TX WAN port
- 10/100/1G LAN interface (CX and CXi)
- Sixteen 10/100Base-TX 802.3af LAN ports (CXi only)
- Two USB 1.1 ports for future use
- Two expansion sites for Dual DSP MMC and T1/E1+DSP+STRATUM 4 Clock MMC
- CompactFlash card reader
- Status LEDs

Front panel (CX)



- RS-232 port (DB-9 connector) Maintenance terminal function (printer function available via LAN port)
- 10/100Base-TX WAN port
- 10/100/1000Base-TX LAN port
- Two USB 1.1 ports for future use
- Expansion sites for Dual DSP and/or T1/E1 Combo modules
- CompactFlash card reader
- Status LEDs

Rear panel (CX and CXi)

- Embedded Analog Main Board (AMB) provides 6 LS (RJ11), 4 ONS (RJ11), with 2 PFT ports, a MOH (3mm) port, a Paging (RJ45) port, 2 generic relays.
- Embedded Analog Option Board (AOB) provides an additional 6 LS (RJ11), 4 ONS (RJ11), and 2 PFT ports. (The LS Trunk and ONS ports on the AOB have the same interface characteristics as the ports on the AMB. The PFT function is associated with ONS1/LS1 and ONS2/LS2 pair.)
- One AC plug site is provided for the 85-265VAC universal power supply

Internal Components

- Analog Main Board
- Analog Option Board
- Hard Drive (Optional)
- Internal CompactFlash card (512 MB)
- External CompactFlash card slot
- 250 Watt Power supply
- Cooling fan
- Option sites for T1/E1/DSP (also known as T1/E1 Combo) and a dual DSP module
- Dual DSP module
- Quad CIM module (for connecting ASUs)

AX Controller

The AX Controller provides support for IP devices and analog devices. It is ideal where a high density of analog devices is required. It can be deployed as a standalone system or included in a network of systems to provide additional analog support.

The AX Controller supports a maximum of 248 IP devices, or a maximum of 288 ONS devices, or a combined maximum of 300 devices.

Note: When installed in a low traffic environment (for example, Hospitality), the AX can support 288 analog sets and 248 IP sets, for a combined total of 536 devices.

The AX Controller provides

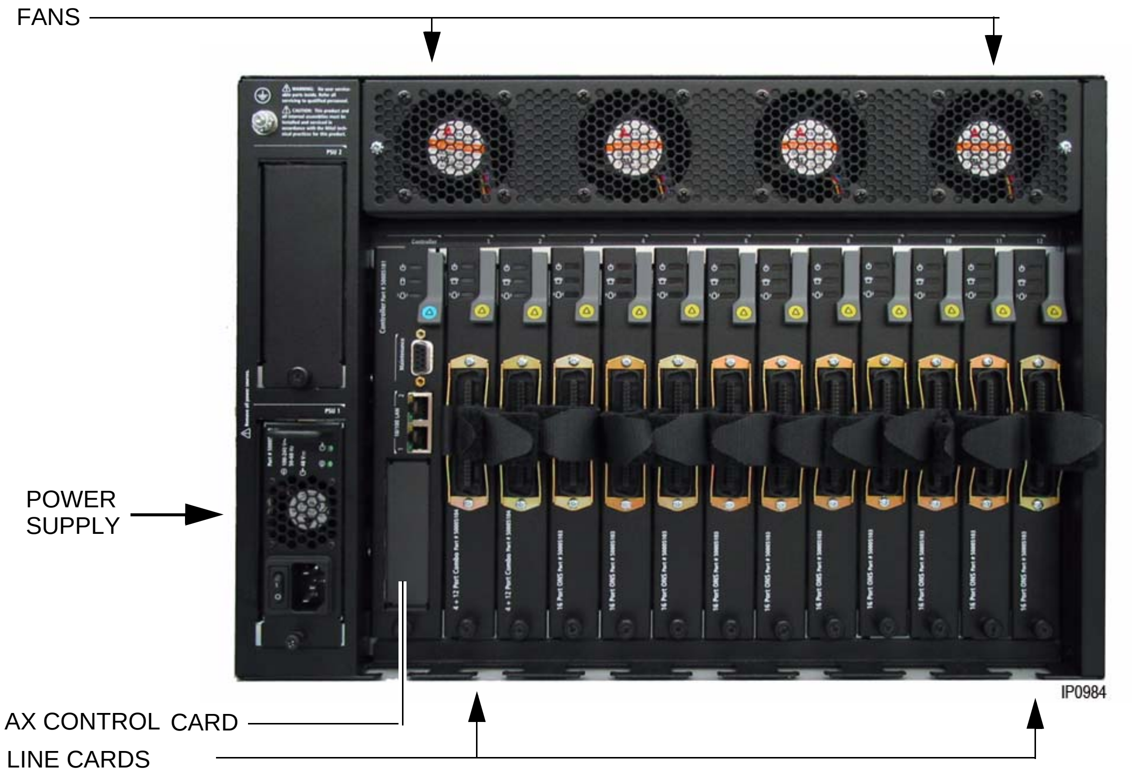
- 12 line card slots to support analog phones and trunks. The following cards are available:
  - 24-port ONS line card
  - 4 + 12 port combo card (4 analog trunks and 12 ONS ports)
- two 10/100 BaseT Ethernet LAN ports (RJ-45 connector).
- one externally accessible expansion slot and one internal expansion slot for up to two of the following optional modules:
  - Dual T1/E1 (external)
  - T1/E1 Combo (external)
  - DSP (internal or external).
- 4 GByte flash card (hosts system files, partitions, and voice mail storage)

Optionally, you can install:

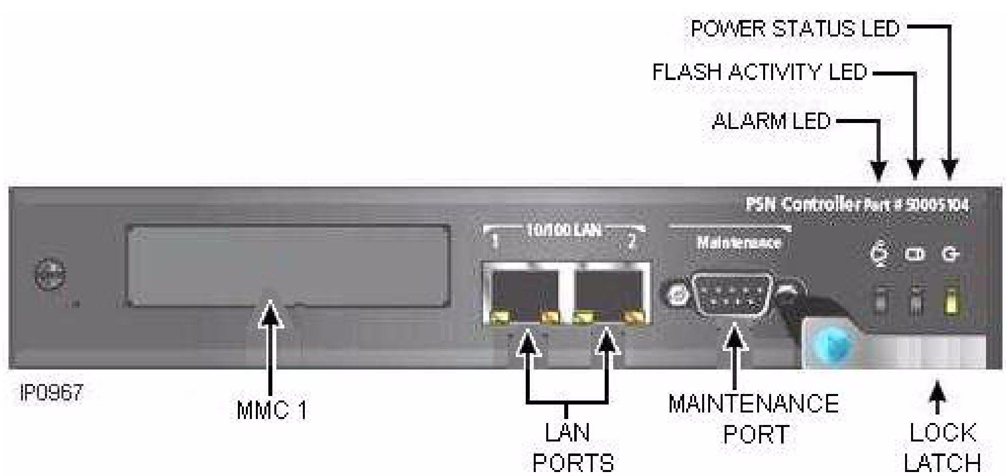
- second AC Power Supply Unit (PSU) for power redundancy
- line cards.

The AX Controller consists of a card chassis, power supply, controller card, and the optional line cards. The power supply, controller card, and line cards are accessed from the rear of the controller.

AX Controller Rear View



AX Controller Controller Card



MX Controller

The MX Controller has the processing, memory, mass storage, power, and input/output capabilities to support up to 248 IP phones or 650 ONS/DNIC phones or a combination of both. It provides 64 channels of Ethernet to Time Division Multiplexing (E2T) and 64 channels of echo cancellation.

The MX Controller is shipped with a Dual DSP Module and embedded Dual CIM for baseline telephony requirements and can be configured with:

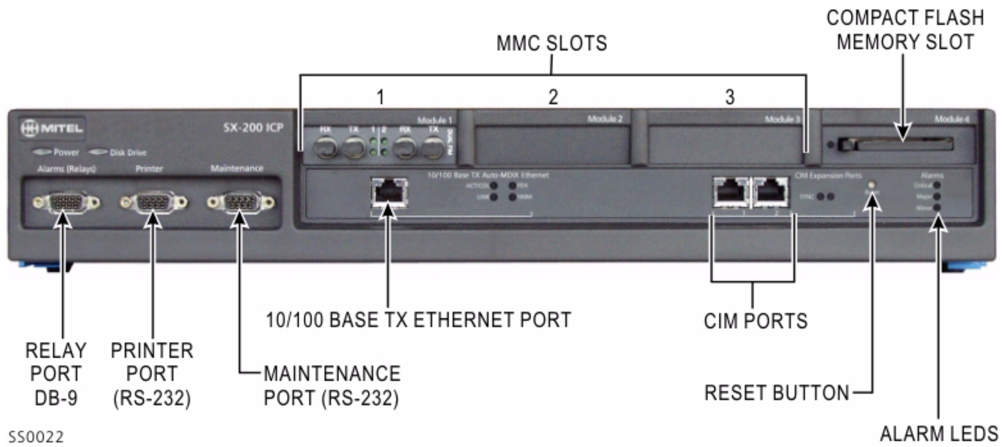
- An additional DSP for G.729 compression (or voice mail ports depending on business configuration that is ordered)
- A Dual FIM for connecting NSUs or Peripheral Cabinets
- One or two Quad CIMs for connecting NSUs or Peripheral Cabinets
- One or two Dual T1/E1 Framers

These modules are shipped separately and must be installed on site.

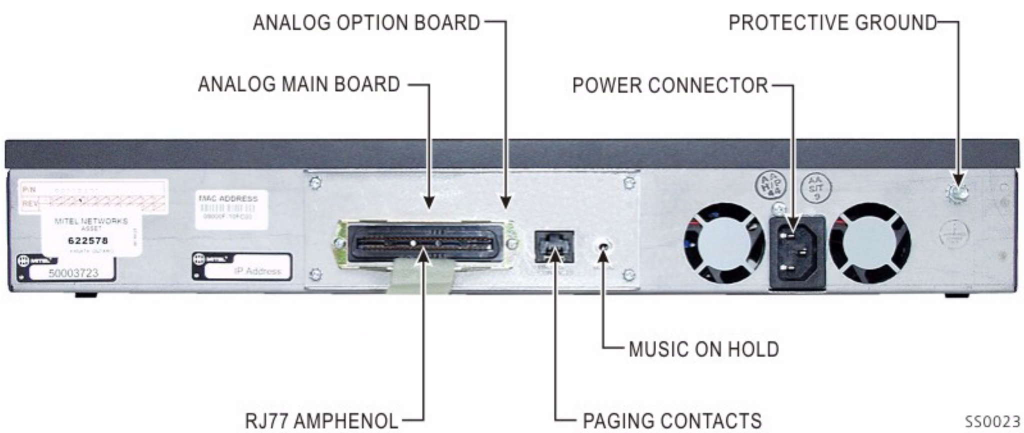
The MX Controller can ship with embedded analog capability provided by one or two optional analog boards:

- Analog Main Board (AMB) provides 2 DNIC ports, 6 Loop CLASS, 2 ONS, 1 Music On Hold, and 1 Paging circuit
- Analog Option Board (AOB) expands on the AMB providing an additional 6 LS CLASS and 2 ONS circuits

Front panel



Rear panel



- Power connector
- Protective ground to ground the chassis
- Music on Hold connector
- Paging/Door Sense Contacts connector
- Amphenol Connector for analog trunks and ONS/DNIC stations

Internal components

- Analog Main Board
- Analog Option Board
- Hard Drive (Optional)
- Internal CompactFlash card
- External CompactFlash card slot
- Stratum Clock (Optional)
- Dual DSP Module/Quad DSP Module (Optional)
- Dual FIM Module (Optional)

- Quad CIM (Optional)
- Power supply
- Cooling fan
- System ID Module

Comparison of SX-200 ICP MX, CX/CXi, and AX controllers

The SX-200 ICP MX, CX/CXi, and AX controllers share the same call control software; however, there are key differences between their hardware.The SX-200 ICP CXi controllers have a Layer 2 switch (managed and powered) built in; the MX, CX, and AX do not. The maximum number of IP desktops for the SX-200 ICP MX is 248; for the SX-200 ICP CX/CXi it is 100, and for the AX it is 248. The MX provides more TDM expansion room than the CX/CXi controller.

For specific details on the differences between these controllers, refer to the following table.

Table 1: Comparison of SX-200 ICP MX, CX/CXi and AX controllers

Area	SX-200 ICP MX controller (200 user)	SX-200 ICP CX/CXi controller (100 user)	SX-200 AX Controller (200 user)
Base system	1 LAN port	1 WAN port 1 10/100/1 GigE LAN port 16 10/100 802.3af LAN ports (CXi only)	2 10/100 LAN ports
	Internal hard drive	512 MB internal Compact Flash	4GB internal Compact Flash
	3 MMC slots	2 MMC slots	2 MMC slots
	Dual DSP MMC (max 8 DSPs)	Dual DSP on mainboard (max 7 DSPs)	Dual and Quad DSPs (max 8 DSPs)
	3 serial ports (maintenance, printer, alarms)	1 serial port for Maintenance Terminal	1 serial port for maintenance
	SYSID module for MOSS	System i-Button for MOSS	System i-Button for MOSS
	Analog mainboard with 2 ONS/2DNIC/6 LS (RJ77), music, pager and night bell	Analog mainboard with 4 ONS/6LS (RJ11), music, pager and night bell/door sensor/alarm	n/a
	Stratum 3 clock	n/a	n/a
	Echo cancellation in hardware	Echo cancellation in software on dual DSP mainboard	Echo cancellation in software on dual DSP mainboard
			Page 1 of 4

Table 1: Comparison of SX-200 ICP MX, CX/CXi and AX controllers (continued)

Area	SX-200 ICP MX controller (200 user)	SX-200 ICP CX/CXi controller (100 user)	SX-200 AX Controller (200 user)
Field Replaceable units	Dual and Quad DSP MCC	Dual and Quad DSP MMC	Dual and Quad DSP MMC
	Dual T1/E1 Framer card	T1 Combo card (T1 link; DSP; Stratum 4 clock)	Dual T1/E1 Framer card T1 Combo card (T1 link; DSP; Stratum 4 clock)
	Internal hard drive	Internal hard drive (optional)	N/A
	Optional Analog board with 2 ONS/6LS	Optional analog board with 4 ONS/6LS	N/A
	Stratum 3 clock	Stratum 3 clock Stratum 4 clock on T1 Combo card	Stratum 4 clock on T1 Combo card
	Dual FIM MMC (optional)	N/A	N/A
	Quad CIM MMC (optional)	Quad CIM MMC (optional)	N/A
Nodes supported	Digital (TDM) Bays	N/A	N/A
	NSU (PRI only)	N/A	N/A
	ASU (24 ONS only)	N/A	N/A
	ASU II (16 ONS, 24 ONSp, or 12 ONS/ 4 LS combo)	ASU II (16 ONS, 24 ONSp, or 12 ONS + 4 LS combo)	N/A
Line and Trunk Interfaces Supported	ONS/CLASS	ONS/CLASS	ONS/CLASS
	OPS	N/A	N/A
	DNIC	N/A	N/A
	IP	IP	IP
	LS/GS	N/A	N/A
	LS/CLASS	LS/CLASS	LS/CLASS
	DID (Analog)	N/A	N/A
	E&M (Analog)	N/A	N/A
	T1/D4 (embedded module in controller or card in peripheral cabinet)	T1/D4 (embedded module)	T1/D4
	PRI (embedded module in controller, card in peripheral cabinet, or NSU)	PRI (embedded module in controller)	PRI
	IP Trunk	IP Trunk	IP Trunk
			Page 2 of 4

Table 1: Comparison of SX-200 ICP MX, CX/CXi and AX controllers (continued)

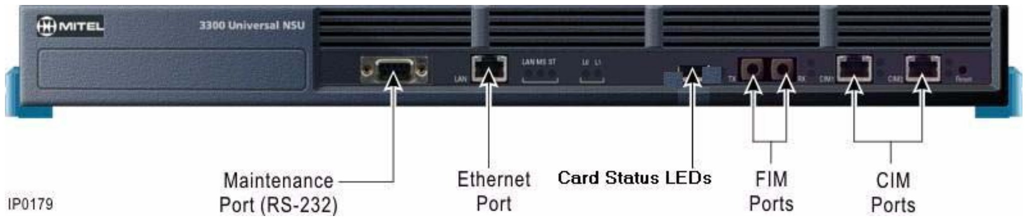
Area	SX-200 ICP MX controller (200 user)	SX-200 ICP CX/CXi controller (100 user)	SX-200 AX Controller (200 user)
Feature Packages supported	ACD	ACD (on IP sets only)	ACD (on IP sets only)
	Hospitality	Hospitality	Hospitality
	Tenanting	Tenanting	Tenanting
	Centralized Voice mail	Tenantingl	Tenanting
	Automated Attendant and Fax Tone Detect	Automated Attendant and Fax Tone Detect	Automated Attendant and Fax Tone Detect
	Mitel Express Messenger™ (version 4.11 and later supported on migration only)	N/A	N/A
	SMTP Client	SMTP Client	SMTP Client
	MiTAI™ (version 11.2 and later)	MiTAI (version 11.2 and later)	MiTAI (version 11.2 and later)
	N/A	Internet Gateway (CXi only)	N/A
	24 Voice Mail ports	16 Voice Mail ports	20 Voice Mail ports
Applications supported	Teleworker Solution (version 3.0 and later)	Teleworker Solution (version 3.0 and later)	Teleworker Solution (version 3.0 and later)
	Mitel Contact Center (6110) (version 4.5 and later)	Mitel Contact Center (6110) (version 4.5 and later)	Mitel Contact Center (6110) (version 4.5 and later)
	Mitel Contact Center Scheduling (6120) (version 4.5 and later)	Mitel Contact Center Scheduling (6120) (version 4.5 and later)	Mitel Contact Center Scheduling (6120) (version 4.5 and later)
	Mitel Real-Time Schedule Adherence (5125)	Mitel Contact Center Scheduling (6120) (version 4.5 and later)	Mitel Contact Center Scheduling (6120) (version 4.5 and later)
	Mitel Multimedia Contact Center (6150)	MItel Multimedia Contact Center (6150)	MItel Multimedia Contact Center (6150)
	Mitel Intelligent Queue (version 2.4, Analogue connection only)	N/A	N/A
	Mitel Speech Server (6500) (version 4.6, DNIC connection only)	N/A	N/A
	Unified Messaging - Standard (Managed Application Server-based)	Unified Messaging - Standard (Managed Application Server-based)	Unified Messaging - Standard (Managed Application Server-based)
	MiTAI SDK	MiTAI SDK	MiTAI SDK
	Mitel Your Assistant	Mitel Your Assistant	Mitel Your Assistant
	NuPoint Messenger™ (Version 8.5 and later)	N/A	N/A
			Page 3 of 4

Table 1: Comparison of SX-200 ICP MX, CX/CXi and AX controllers (continued)

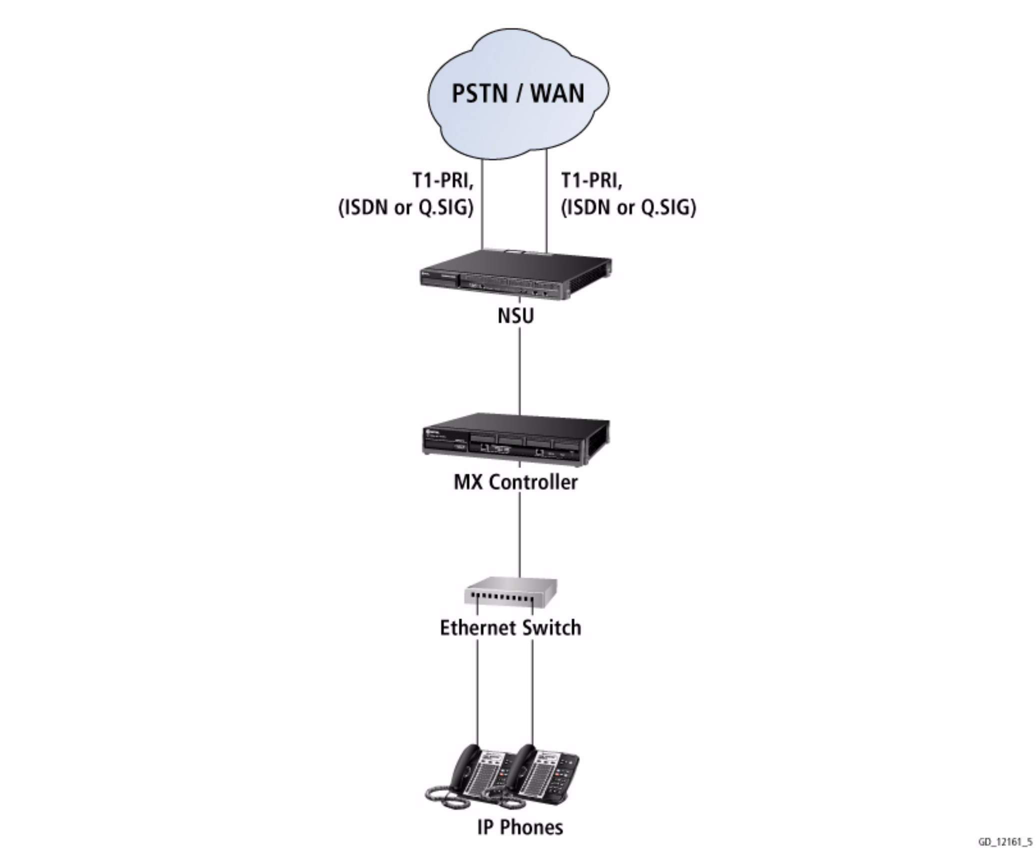
Area	SX-200 ICP MX controller (200 user)	SX-200 ICP CX/CXi controller (100 user)	SX-200 AX Controller (200 user)
Operations and Maintenance	Software on internal flash/hard drive	Software on internal flash/hard drive	Software on internal flash
	Hot-swappable external compact flash (release 2.0 and later)	(embedded module in controller)	4GB Flash Card
	InstallShield for software/database installation	InstallShield for software/database installation	InstallShield for software/database installation
	New install and upgrade with flash	New install and upgrade with flash	New install and upgrade with flash
	Remote upgrade via FTP server	Remote upgrade via FTP server	Remote upgrade via FTP serve
	Default and blank databases	Default and blank databases	Default and blank databases
Call control	Platform independent with additional functionality through bays and support of older sets	Platform independent; does not support older sets	Platform independent; does not support older sets
Voice mail	EMEM ONS Voicemail DNIC Voicemail MEM (card in bays)	EMEM N/A N/A N/A	EMEM N/A N/A N/A
Layer 2 switch	Embedded Layer 2 switch not supported	Embedded Firewall, Internet Gateway (CXi only)	Layer 2 switch is supported
			Page 4 of 4

Network Services Units

The Mitel Network Service Unit (NSU) is used with the MX controller only and provides connectivity to digital trunks for public or private networks. The NSU supports up to two PRI links per unit both using Primary Rate ISDN (4ESS, 5ESS, DMS 100, DMS 250, NI2) protocol.



The NSU connects to an SX-200 ICP Controller through a CIM or FIM connection.

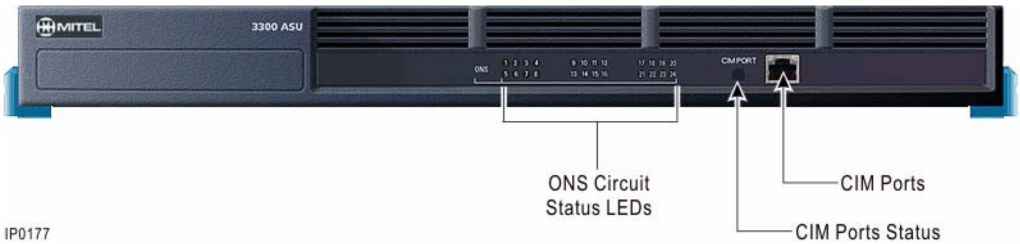


Analog Services Unit

The Analog Services (ASU) unit is used with the MX controller only and provides 24 ONS/CLASS circuits. Up to six ASUs can be connected to an MX Controller. Connection is via a Category 5 Universal Twisted Pair (UTP) cross-over cable through a CIM interface.

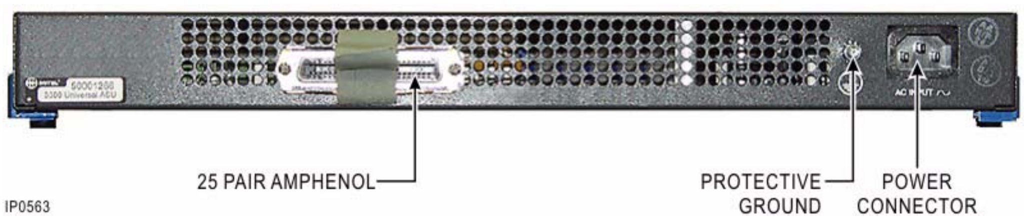
Note: The ASU only supports DTMF telephones (pulse or rotary dial phones are not supported).

Front Panel



- 24 ONS Circuit LEDs indicate the status of the telephone circuits
- 1 CIM circuit LED indicates the status of the CIM link
- RJ-45 connector (CIM connection to the Controller)

Back Panel



- 25 pair D-type connector provides access to the ONS Tip/Ring circuits.
- Standard Male IEC AC input connector for power requirement.

Analog Services Unit II

The ASU II is used with both MX and CX/CXi controllers and supports up to 48 ONS phones or up to 8 LS trunks depending on how the unit is configured with peripheral cards:

The **4 + 12 port combo** card supports:

- 12 On-Premise Station (ONS) Lines for analog phones
- Four Loop Start (LS) trunks for analog connection to a central office
- Four System Fail Transfer (SFT) relays that provide direct connection between an analog telephone and a Loop Start trunk in the event of system or power failure.

The North American version supports Custom Local Access Signalling Services (CLASS) on the ONS circuits. CLASS allows the SX-200 ICP system to pass Calling Line ID digits and CLASS name information to display sets that support Caller ID functionality.

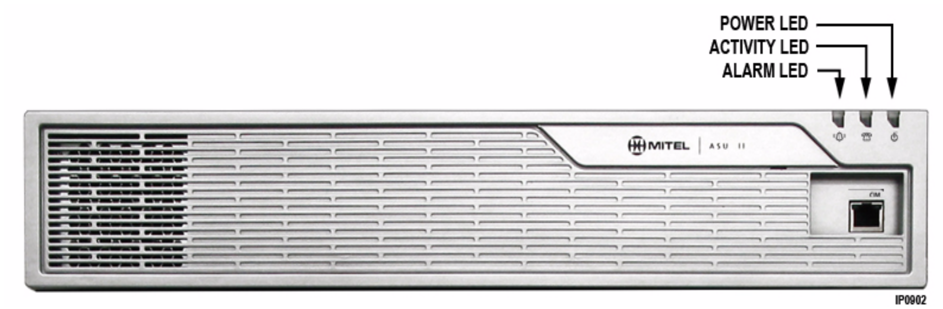
The **16 port ONS** card supports:

- 16 On-Premise Station (ONS) Lines for analog phones

The **24 port ONSp** card supports:

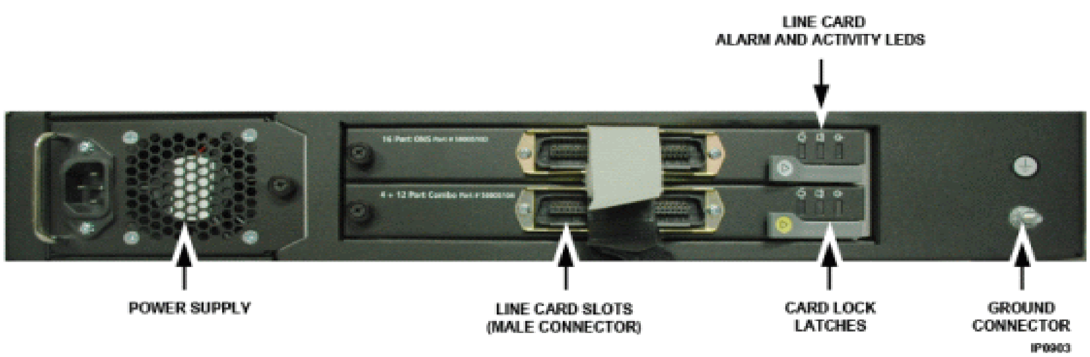
- 24 On-Premise Station (ONS) Lines for analog phones. Circuits on this card have additional electrical protection.

Front Panel



- Alarm, activity, and power LEDs
- CIM status LED
- One CIM port for connecting to the Controller

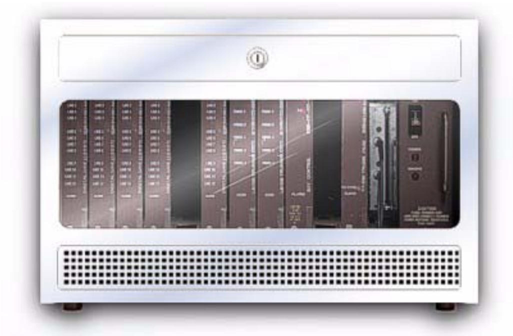
Back Panel



- Power supply and fan
- Two slots for peripheral line cards with amphenol connector (16 port ONS, 24 port ONSp, or 4+ 12 Combo)
- Card identifiers
- Protective ground for grounding the chassis

Peripheral Cabinets

SX-200 Peripheral Cabinets are supported on the SX-200 ICP MX only. Up to seven cabinets can be connected to the controller via a combination of seven CIM and/or four FIM cables. The cabinets can be SX-200 ELx peripheral cabinets, SX-200 LIGHT peripheral cabinets, or a mix of both.



SX-200 PERIPHERAL CABINET

SX-200 LIGHT
PERIPHERAL CABINET

SS0025

SX-200 ELx Cabinet

The SX-200 ELx peripheral cabinet is horizontal and can be mounted in a standard 19" rack, or they can be stacked. The cabinet is plastic and Plexiglas. The door on the cabinet allows the system administrator to see the system status at a glance. The control cabinet and the peripheral bays are linked by fibre or copper cables.

The SX-200 ELx cabinet supports 12 card slots: eight slots support line and trunk cards, and four support the control cards and the FIM or CIM carrier cards.

SX-200 LIGHT Peripheral Cabinet

The SX-200 LIGHT Peripheral Cabinet is vertical. The cabinet contains one Bay Power Supply, one Bay Control Card (with attached Peripheral FIM Carrier plus FIM), and up to eight peripheral interface cards.

Peripheral Bay Power Supplies

The Bay Power Supply (BPS) is card-mounted and is located in the Peripheral cabinet. 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. 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 has a single alarm signal, PFS (power fail sense), which is driven low when the incoming ac falls below its minimum specified value.

Peripheral Interface Cards and Modules

The following cards are installed in the SX-200 ICP peripheral cabinets:

- "Universal Card" on page 27
- "ONS/CLASS Line Card" on page 27
- "Digital Line Card" on page 27
- "LS/GS Trunk Card" on page 28
- "LS/CLASS Trunk Card" on page 28
- "Direct Inward Dial (DID) Trunk Card" on page 28
- "Off-Premise (OPS) Line Card" on page 28
- "Mitel Express Messenger Card" on page 28
- "Peripheral FIM Carrier" on page 28
- "Peripheral FIM Carrier II" on page 28

Universal Card

The Universal Card is a high power card that holds up to four modules. Each module is assigned a power rating. The cumulative ratings of the modules on the Universal Card cannot exceed a value of 10. The modules are as follows:

- Receiver/Relay Module (contains four DTMF receivers and two relays) (power rating = 2)
- Music-on-Hold/Pager Module (contains one music input, one PA paging output) (power rating = 1)
- E&M Trunk Module (contains one E&M trunk) (power rating = 3)

ONS/CLASS Line Card

The ONS/CLASS Line card is a low power card and replaces the ONS Line card. The card has the same functionality as the ONS Line card and if software enabled, offers CLASS functionality.

The ONS/CLASS Line card has 12 DTMF/Rotary line circuits per card. The card accepts up to three industry-standard DTMF/Rotary telephone sets per line circuit. The 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.

Digital Line Card

The Digital Line Card (DLC) is a low power card with 12 Digital Network Interface Circuits (DNIC) per card. The Digital Line Card interfaces DNIC-based peripheral devices to the system through its Digital Network Interface Circuits (DNIC); the DNIC is a proprietary integrated circuit. DNIC devices include Mitel Superset™ 4001, Mitel Superset 4015, Mitel Superset 4025, Mitel Superset 4125, Mitel Superset 4150, Mitel Superset 401+, Mitel Superset 410, Mitel Superset 420, Mitel Superset 430 telephones, Mitel Programmable Key Modules (PKM), DATASETs, SUPERCONSOLE 1000® Attendant Console, DMP Module.

LS/GS Trunk Card

The LS/GS Trunk card is a low power card that contains six loop start or ground start trunks (jumper-selectable) and six message registration inputs. The card may be installed in any digital peripheral slot. Facilities provided by the LS/GS Trunk Card include: Loop Start or Ground Start selectable by jumper, M and MM signaling leads (refer to the feature, Meter Pulse Collection), trunk activity indicated by LED (one per trunk), transient suppression on Tip, Ring, and signaling leads, and an alarm LED.

LS/CLASS Trunk Card

The LS/CLASS Trunk card interfaces eight trunk circuits to the system. LS is the acronym for Loop Start and CLASS is the acronym for Custom Local Area Signaling Services (allows the system to receive calling Line ID digits and CLASS name on incoming CLASS trunks). The LS/CLASS Trunk card can be installed into slots one to eight in a SX-200 rack mount cabinet. The LS/CLASS Trunk card provides loop start operation, forward/reverse current detectors (polarity reversal, answer supervision), alarms and trunk activity indicated by an LED (a single LED for any circuit in use), CLASS signal reception, and transient suppression on Tip and Ring leads.

Direct Inward Dial (DID) Trunk Card

The DID trunk card is a high power card that contains six 1-way Direct Inward Dial circuits. The DID trunk allows incoming trunk calls to dial directly to an extension within the system without attendant intervention.

Off-Premise (OPS) Line Card

The OPS line card is a low power card that interfaces the system to analog extensions which are part of the system, but are located in a different building from the PBX. It contains additional protection circuitry to protect the system from extraneous high voltages or induced currents that may appear on the line. Each OPS card has six circuits.

Mitel Express Messenger Card

The Mitel Express Messenger card is a low-power card. The card uses a DNIC interface that connects directly to the backplane of the cabinet. The card provides either two, four, six, or eight voice mail ports.

Peripheral FIM Carrier

The Peripheral FIM Carrier (PFC) provides the interface between the Bay Control Card and the Fiber Interface Module for the SX-200 LIGHT Peripheral cabinet. It connects to the module position on the Bay Control Card and acts as a carrier for the Fiber Interface Module.

Peripheral FIM Carrier II

The Peripheral FIM Carrier II (PFC II) provides the interface between the backplane of an SX-200 RM peripheral cabinet and its Fiber Interface Module.

Manufacture discontinued or unsupported devices

The following devices have been manufacture discontinued by Mitel. However, they are supported on SX-200 ICP MX controller systems that have been upgraded from the SX-200 EL/ML.

- Superset 3DN and Superset 4DN telephones
- Superset 400 series telephones
- Superset PKM
- MiLINK® Data Module
- LCD Console
- SUPERCONSOLE 1000
- DSS/BLF Interface Unit

The following devices are not supported on the SX-200 ICP system:

- Superset 3 and Superset 4 telephones
- Superset 4 telephones for voice mail
- Modem Interconnect Panel
- DATASET 1102 Rack-mounted Dataset
- DATASET 2102 Rack-mounted Dataset
- DATACABINET 9000 data cabinet
- DATASHELF 9100 datashelf
- ISDN Gateway
- MyAdministrator software application

Desktop Devices

For details on each of these devices supported on SX-200 ICP, refer to “Desktop Devices” on page 39.

Digital Services Cards and Modules

The following Digital Services cards and modules are supported on the SX-200 ICP:

- “T1 Trunk Card” on page 30
- “Bay Control Cards” on page 30
- “PRI Card” on page 30
- “Fiber Interface Modules (FIM and FIM II)” on page 30
- “Dual Fiber Interface Module” on page 31

- "Peripheral Interface Module Carrier Card" on page 31
- "T1/E1 Module" on page 31
- "Dual Link T1/E1 Framer MMC" on page 31
- "T1/E1 Combo MMC" on page 31
- "CIM (Copper Interface Module)" on page 31
- "Quad Copper Interface Module" on page 32
- "DSP Modules" on page 32

T1 Trunk Card

The T1 Trunk Card is a high power card that provides an interface to one 24-channel (D4 format) T1 link. In an SX-200 rack mount cabinet, T1 trunk cards plug into slots 10 and 11 (slots 5 and 6 respectively must then be left vacant). Because of signal cable restrictions in an SX-200 FD cabinet, the T1 card must be positioned in slot 6. With a dual T1 adapter, two T1 trunk cards (in slots 5 and 6) are allowed.

Bay Control Cards

The bay control cards provide control of operations within the cabinet and monitor the lines, trunks and other circuits within the bay. Reports are sent to the 200 ICP Controller via HDLC message links. One bay control card is required in each cabinet.

The bay control cards are BCC II and BCC III. The BCC III supports a DSP module (single), a T1/E1 module, and a FIM II or CIM. The T1/E1 module and the FIM II provide a cost-effective solution for T1 network connectivity for a remote system. The CIM offers extra savings for a co-located system. The DSP module supports the ONS/CLASS Line card and the Record a Call feature. The BCC III cannot be installed in a LIGHT Peripheral Cabinet.

PRI Card

The PRI card provides Primary Rate Access (PRA) to the ISDN service provider. The PRI card, preloaded with software, comes with a T1/E1 module that supports up to two T1 links of ISDN connectivity. The PRI card also requires a Stratum 3 clock in the SX-200 ICP controller. The PRI card (unlike the T1 card) is not classed as a high power card. Because the PRI card is a separate bay, the PRI card is not included in the count for the four high power cards. The PRI card cannot be installed in a LIGHT Peripheral Cabinet.

Fiber Interface Modules (FIM and FIM II)

There are two main types of Fiber Interface Modules: FIM and FIM II. The FIM and FIM II provide a fiber optic based communications link between the SX-200 ICP and a peripheral cabinet or NSU that houses an equivalent FIM. The main difference between the FIM and FIM II is where they are installed. The Fiber Interface Module (FIM) sits on Control FIM Carrier cards and Peripheral FIM Carrier II cards. The FIM II is a module that resides on a PRI card, on a Bay Control Card III (BCC III), or on a Peripheral Interface Module Carrier card. The FIM II on a PRI card is an alternative to the FIM on a control carrier card. The FIM II on the BCC III (in a peripheral cabinet only), is an alternative to the FIM on the Peripheral FIM Carrier II card.

The FIM II on a Peripheral Interface Module Carrier card provides connectivity when the peripheral cabinet does not have a BCC III to hold a FIM II. The FIM II module does not apply to LIGHT Peripheral Cabinets.

Dual Fiber Interface Module

The Dual FIM module, available on the MX controller only, is an optional electrical/optical interface. It connects SX-200 Peripheral Cabinets or NSUs to the controller through fibre optic cable. When transmitting, the module converts electrical signals to optical signals for output over a fiber optic cable. When receiving, it converts optical signals from the cable to electrical signals. The MX can support up to two Dual FIMs installed in MMC slots 1 and 2. The Dual FIM is available as a Field Replaceable Unit (FRU).

Peripheral Interface Module Carrier Card

The Peripheral Interface Module Carrier card is a carrier card for a FIM II or a CIM. The card provides fiber or copper connectivity between a peripheral cabinet and a main control cabinet when the peripheral cabinet has a BCC II instead of a BCC III. Instead of using the Peripheral Interface MMC, if the BCC III was in the peripheral cabinet, the interface modules (FIM II or CIM) on the BCC III would provide the connectivity needed to the main control cabinet.

T1/E1 Module

The T1/E1 module on site 2 of the PRI card provides up to two links of ISDN connectivity. The T1/E1 module on site 2 of the BCC III provides up to two T1 links. The links from the module provide CSU and ESF functionality.

Dual Link T1/E1 Framer MMC

The Dual T1/E1 Framer MMC is available as option for the MX and AX controller. The module has two digital trunk ports, each of which can be configured as a T1 interface (1.544 Mbits/sec) that provides 24 B-channels for T1/D4. The Dual T1/E1 Framer also supports PRI. Up to two modules can be installed in MMC slots 1 and 2. The Dual T1/E1 Framer is available as Field Replaceable Unit (FRU).

T1/E1 Combo MMC

The T1/E1 Combo MMC combines trunking and DSP functionality in a single card for the CX/CXi and AX controllers. The digital trunk port can be configured as a T1 (1.544 Mbits/sec) that provides 24 B-channels for T1/D4. The T1/E1 Combo also supports PRI. The DSP provides resources for CLASS tone generation, Record a Call conferences, DMTF receivers, voice compression, and voice echo cancellation. The module also includes a Stratum 4 clock. The CX/CXi can support a single T1/E1 Combo MMC installed in MMC slot 1 or 2. The AX can support a single T1/E1 Combo MMC installed in MMC slot 1 only. The module is an optional Field Replaceable Unit (FRU).

CIM (Copper Interface Module)

The CIM provides copper connectivity between the peripheral cabinets and the controller. The CIM is very cost effective for a system that is co-located. The CIM supports a distance of up to

30 meters or 100 feet between cabinets. The CIM sits on a Peripheral Interface Module Carrier card in a peripheral cabinet, on site 1 of the BCC III in a peripheral cabinet, or on site 1 of a PRI card in a peripheral cabinet. Unlike the FIM II, the CIM sits close to the faceplate and only has one variant.

The MX controller is equipped with an onboard dual-port Copper Interface Module (CIM). An optional Quad CIM module provides four more ports. Up to two Quad CIMs can be installed in MMC sites 1 and 2.

Quad Copper Interface Module

The MX controller is equipped with an onboard dual-port Copper Interface Module (CIM). An optional Quad CIM module (available for both the MX and the CX/CXi) provides four more ports. Up to two Quad CIMs can be installed in MMC sites 1 and 2. CIM ports can be connected to ASUs, ASU IIs, NSUs, or Peripheral Cabinets. Although a single-port CIM is available for use in the SX-200 Peripheral cabinet, this module cannot be installed in an SX-200 ICP.

DSP Modules

DSP (Digital Signal Processor) modules are available in three configurations: Single, Dual, and Quad. The Single DSP services an SX-200 Peripheral Cabinet and sits on Site 3 of the Bay Control Card III (BCC III). The Dual and Quad reside in the SX-200 ICP controller. The Single DSP module provides the following functionality:

- CLASS tones for the ONS/CLASS Line card. This DSP module has 8 CLASS generator resources that are assigned and released dynamically as they are required. The DSP module on the BCC III must reside in the same bay as the ONS/CLASS Line card.
- Sixteen conference bridges for the Record-a-Call feature. The DSP module provides these bridges to circuits in the same bay. A Record a Call conference is between one internal party, one external party (a trunk call) and the voice mailbox. A Record a Call conference can also be setup between two internal parties and the voice mailbox.
- Sixteen DTMF receivers that can be used system wide. The DSP module replaces the Universal Card with respect to DTMF receivers.

The Dual and Quad DSPs provide the functionality of the Single DSP but in greater quantities plus voice compression resources.

Application Processor Card (CX/CXi only)

The Application Processor Card is a PC on a compact card. Installed in the CX/CXi controller, the Application Processor Card hosts the Managed Application Server (MAS) and associated applications. The Application Processor Card requires a dedicated hard drive.

Network

- “Internet Gateway (CXi only)” on page 36
- “Embedded firewall (CXi only)” on page 36

ISDN (Integrated Services Digital Network)

ISDN support is provided from the ISDN Primary Rate Interface Card (PRI) card in the peripheral cabinet, from the NSU, or on the embedded T1/E1 modules. For more information on the NSU, refer to “Network Services Units” on page 23. For a list of ISDN network services supported by the PRI card and the NSU, refer to “Supported ISDN network services” on page 33.

The Integrated Services Digital Network (ISDN), transmitting voice, data and video at high speeds, accurately and without a modem, has revolutionized communications. ISDN services can be deployed and accessed at enterprise, department and desktop levels by its simple addition to your existing SX-200 network. ISDN proves its worth by its ability to carry voice, data and video imaging on one network.

The SX-200 ICP MX supports Primary Rate Interface (PRI) via the NSU or a PRI card in a bay, or on the embedded T1/E1 module.

ISDN Primary Rate Interface Card

ISDN PRI is becoming the most cost-effective solution for accessing enhanced voice capabilities. All inbound and outbound services that are usually obtained by using different trunk types (such as INWATS, OUTWATS, FX, Tie, and DID) can be accessed with a single ISDN trunk; as a result, the number of system trunks can be reduced by 10 to 15 percent. On outbound calls, the system requests the required service from the Network. The trunk takes on the requested characteristics for the duration of the call.

The PRI card in the peripheral cabinet and the NSU cabinet provides two ISDN links and has the Bearer Capabilities (BC) of Speech (voice) and 3.1 kHz audio. The card also transports the BCs of rate-adapted 56 kbs data and unrestricted 64 kbs data transparently through the system.

Supported ISDN network services

- **Calling Party Number (CPN)** - This number substitutes the calling station number on outgoing calls for purposes of network identification and call back.
- **Calling Name ID (CNID)** - This ID is the incoming calling name delivery per NA DMS100 custom specification or National ISDN-3 (NI3).
- **Calling Line Identification Presentation (CLIP**) - The Calling Party Number can be provided to the ISDN Network for outgoing calls or provided to the PRI card from the ISDN Network for incoming calls. This information is passed onto the system and can be used for database applications such as screen pops and for inclusion in SMDR records.
- **Calling Line Identification Restriction (CLIR) -** This feature allows users to prevent their telephone number from being presented to the called party.
- **Partial PRI Links -** The SX-200 PRI card will support COs that provide this feature.
- **Direct Dial-In (DDI) -** DDI is an ISDN option that allows direct access to a line behind a system through a unique directory number. This allows the dialed digits of an incoming

ISDN call to be presented to the system. All ISDN trunks are treated as Dial-In trunks; the CO always sends digits to the system.

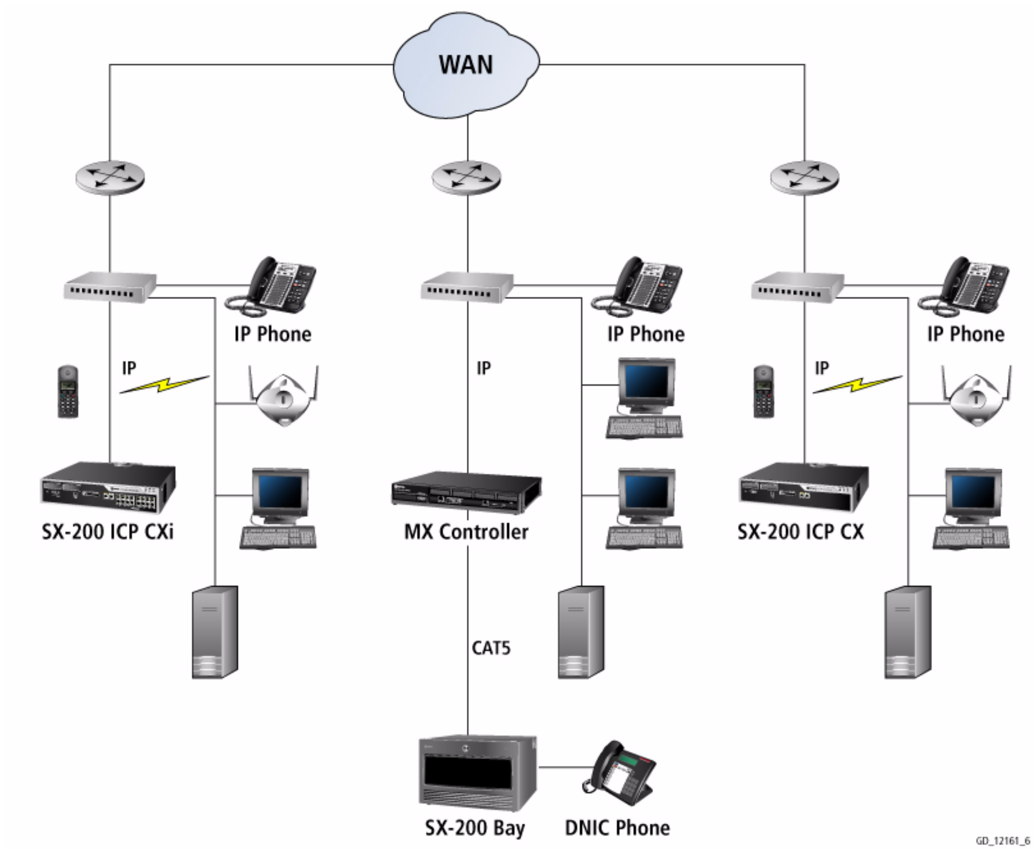
- **Call-By-Call Service Selection (CBC) -** This feature allows telephone users to select the ISDN network services that they wish to use on a per call basis.
- **DID Calling Party Number Forwarding -** Outgoing CPN delivers the calling party's DID number to the Network when the call has been identified as a call from a device with an associated DID number instead of delivering the main directory number associated with the system.
- **Equal Access to Interexchange Carriers -** The system provides a carrier access code which identifies to the Central Office which Interexchange Carrier is to receive the call. The system outpulses a digit string which includes a carrier access code, followed by an identification number, followed by the called number.
- **Min/Max** (PRI card and NSU only) - This feature allows a customer to control incoming and outgoing call traffic. Minimums are assigned to ensure that a particular type of call (such as INWATS) always has a set number of lines available. Maximums are assigned to limit certain types of calls, i.e., OUTWATS. This ensures that resources are not used up by a single type of call. Different Min/Max databases can be created for different times of the day or for special occasions such as telethons or infomercials.
- **Auto Min/Max** (PRI card and NSU only) - This feature provides user programmable time-of-day automatic control of Min/Max parameters.
- **NFAS** (Non-Facility Associated signaling) (PRI card and NSU only) - NFAS allows you to use a single D-channel to handle the signaling requirements for a group of PRI links that all use the same Protocol. This feature eliminates the need to purchase a D-channel for each link. NFAS is mainly for North America.
- **D-Channel Backup** (PRI card and NSU only) **-** This feature is used for signaling to establish and maintain the circuit, and to send user data. D-channel Backup provides an alternate D-channel for calls related to NFAS. If the active D-channel fails, the system switches to the backup D-channel to support call processing. This functionality is mainly for North America. NFAS is required in order to program D-channel Backup.
- **Q.SIG** (PRI card and NSU only) - This feature provides the ability to connect Q.SIG compatible PBXs from different vendors together to form a private network and to connect the SX-200 ICP to the SX-2000® LIGHT system or any other Q.SIG compatible PBX. Q.SIG features that are supported include Calling Name for incoming calls, Message Waiting Indication, Call Transfer, Call Diversion, and Path Replacement.
- **Remote LAN Access** - This feature provides LAN access to the wide area network (WAN) for both incoming and outgoing calls through LAN servers (routers or bridges).
- **Multiple Variants and Configurations** - This feature provides the ability to run multiple protocol variants and program multiple configurations on the two links of the PRI Gateway through the IMAT application. The option to run multiple variants allows you to connect the PRI Gateway to two different CO switches. The option to run multiple configurations allows you to program Network-side on one link of the PRI Gateway and program User-side on the other link of the PRI Gateway. For more information on programming multiple variants and configurations, refer to the IMAT online Help.

IP Networking

IP Networking provides customers with a new option for networking systems together. Instead of leasing dedicated voice circuits, customers can route voice traffic over the existing LAN/WAN infrastructure.

IP Networking for the SX-200 ICP (MX, CXi, and AX) is supported. Controllers that are geographically separated can be seamlessly networked to share information and services in a transparent and cost efficient manner. IP Networking can be used as the primary communication between controllers or as a backup to TDM networking. The IP Networking feature supports both G.711 and G.729 encoding. Connections with up to 100 other network nodes and a total of 24 channels are supported from any one node.

The following diagram illustrates a sample SX-200 ICP network.



The IP Networking call signaling supported is based on the Q.SIG feature set of the SX-200 ICP and consists of:

- Basic Call: incoming, outgoing
- Call Failures: Busy, Reorder, Alternative route
- Transfers: Supervised and Unsupervised with recall
- Forwarding: Always, Busy and No Answer

- Call Offer: Camp on
- Networked Voice mail
- Networked Attendant
- Message Waiting Indicators
- Network Voice mail Softkeys

Ethernet WAN and LAN interfaces

In addition to standard telephony interfaces, the SX-200 ICP CXi includes a complete range of Ethernet interfaces. It has a WAN interface for connection to the Internet at 10 or 100 Megabits per second. The WAN interface can obtain an IP address by DHCP or PPPoE, or be programmed with a static IP address and default gateway.

Note: IP Trunking is not supported over the CXi WAN interface. All IP Trunks must be via the LAN interfaces (AX, CXi and MX).

The 200 ICP CXi also has a 16-port, 10/100 Layer 2 switch for connection to an Ethernet LAN. Each of the 16 ports provides power to IP devices in compliance with the 'Power over Ethernet' specifications in IEEE 802.3af.

The 200 ICP AX has a two-port, 10/100 Layer 2 switch for connection to an Ethernet LAN.

The CXi also has an unpowered 1 Gigabit Ethernet port for connection to another switch on the LAN. By connecting another switch, it is possible to increase the connection capacity for the CXi from 16 to 100, with 16 connected to the onboard switch and 84 connected to the offboard switch(es).

The internal switch uses a traffic prioritization scheme based on IEEE 802.1 p/Q VLAN prioritization standards. This ensures the quality of voice calls by routing packets with priority value 6 (from IP phones) ahead of packets with priority value 0 (from PCs and other IP devices).

Internet Gateway (CXi only)

The primary function of the Internet Gateway is to link the internal and external networks. In this role, it performs many-to-one NAT (Network Address Translation), converting private IP addresses on the LAN to a single public IP address on the WAN interface. NAT redirect, or "IP port forwarding," is included as a programmable feature, enabling external traffic to reach internal services or machines. The Internet Gateway also provides firewall functionality, logging unknown packets and then either dropping or rejecting them, or allowing the packets to pass through to the internal network. If VPN tunnels are in use, the Internet Gateway can perform IPSec and PPTP pass-through, and can function as a PPTP server.

Embedded firewall (CXi only)

The firewall examines all packets destined for the internal network. Unless they are addressed to a specific TCP or UDP port programmed on the firewall, the packets are declared "unknown" and then either dropped or rejected. Unknown packets are also logged.

The firewall can be configured to allow external access to internal resources by programming port forwarding (NAT redirect).

Directed Data I/O

You can specify the location of system printer ports, designate printout types, and route data to various outputs. The SX-200 ICP supports the following output data functions:

SMDR	Maintenance Logs	Traffic Measurement
CDE Data Print	Hotel/Motel Wakeup	Hotel/Motel Audits
ACD Real Time Events	ACD Agent Summary	Mitel Applications Interface (MAI)
ACD Group Summary	ACD Monitor Print	

Applications requiring bi-directional data, such as ACD Monitors, Hotel Motel Front Desk Terminals and PBX-PMS Interface, are also supported using IP sockets in the SX-200 ICP. IP-enabled devices or applications can connect to the sockets via Telnet. A third-party RS232-to-IP serial port converter is required to connect serial devices to the network.

The system can support seven different printers.

Desktop Devices

Mitel's IP desktop portfolio provides network applications supported by the SX-200 ICP. The portfolio includes:

- "IP Phones" on page 41
- "Accessories for IP Phones" on page 50
- "Conference Phones" on page 55
- "Digital Phones" on page 56
- "Accessories for Superset 4000 Series Digital Phones" on page 59
- "Music-On-Hold/Pager Unit (DMP)" on page 62
- "Power Accessories" on page 63

The following table compares the phones supported by the SX-200 ICP MX, the SX-200 ICP CX/CXi and the SX-200 ICP AX.

Table 2: Comparison of Phones supported by MX, CX/CXi and AX

Phone/Device	SX-200 ICP MX	SX-200 ICP CX/CXi	SX-200 ICP AX
Superset 3DN/4DN Telephones (migration upgrade only)	Y	N	N
Superset 400 Series Telephones (migration upgrade only)	Y	N	N
Superset 4001 Telephone (migration upgrade only)	Y	N	N
Superset 4015 Telephone	Y	N	N
Superset 4025/4125 Telephones	Y	N	N
Superset 4150 Telephone	Y	N	N
Mitel 5001 IP Phone	Y	N	N
Mitel 5010 IP Phone	Y	Y	N
Mitel 5020 IP Phone	Y	Y	N
Mitel 5201 IP Phone	Y	Y	N
Mitel 5207 IP Phone	Y	Y	N
Mitel 5212 IP Phone	Y	Y	N
Mitel 5215 IP Phone	Y	Y	N
Mitel 5215 IP Phone (Dual Mode)	Y	Y	N
Mitel 5220 IP Phone	Y	Y	N
Mitel 5220 IP Phone (Dual Mode)	Y	Y	N
Mitel 5224 IP Phone	Y	Y	N
			Page 1 of 2

Table 2: Comparison of Phones supported by MX, CX/CXi and AX (continued)

Phone/Device	SX-200 ICP MX	SX-200 ICP CX/CXi	SX-200 ICP AX
Mitel 5304 IP Phone	Y	Y	Y
Mitel 5312 IP Phone	Y	Y	Y
Mitel 5324 IP Phone	Y	Y	Y
Mitel 5330 IP Phone	Y	Y	Y
Mitel 5340 IP Phone	Y	Y	Y
NetLink i640 Wireless Telephone	Y	Y	N
NetLink e340/h340 Wireless Telephone	Y	Y	N
Symbol MiNet Wireless IP Phones	Y	N	N
SUPERCONSOLE 1000	Y	N	N
5540 IP Console	Y	Y	Y
Mitel Programmable Key Module (PKM) 12	Y	N	N
Mitel Programmable Key Module (PKM) 48	Y	N	Y
Superset DSS Module (migration upgrade only)	Y	N	N
Mitel IP Programmable Key Module (PKM) 12	Y	Y	Y
Mitel IP Programmable Key Module (PKM) 48	Y	Y	Y
5303 Conference Phone	Y	Y	N
5310 IP Conference Unit	Y	Y	Y
Gigabit Ethernet Phone Stand	Y	Y	N
Wireless LAN Phone Stand	Y	Y	N
Dataset 1103	Y	N	N
Dataset 2103	Y	N	N
Analog Phones	Y	Y	Y
			Page 2 of 2

IP Phones

The SX-200 ICP supports the following Mitel IP Phones:

- "Mitel 5304 IP Phone" on page 42
- "Mitel 5312 IP Phone" on page 43
- "Mitel 5324 IP Phone" on page 44
- "Mitel 5312 IP Phone" on page 43
- "Mitel 5340 IP Phone" on page 46
- "NetLink i640 Wireless Telephone" on page 47
- "NetLink e340/h340" on page 48
- "Telematrix 3000IP" on page 49

Mitel 5304 IP Phone

The Mitel 5304 IP Phone is a two-line, dual port telephone that provides voice communication over an IP network. Features of the newly designed telephone include:

- Support for SIP and MiNET protocols
- 2-line x 20-character white, backlit, graphics display with contrast control and auto-dimming
- 8 programmable multi-function keys (for speed dialing, line appearances, feature access)
- 2 Lines with LED indicators
- Dual port (10 / 100 Mb Switched Ethernet)
- Calling Line ID Support
- Volume/Contrast Up/Down keys, Speed calling, Call forward, Call hold (Place / Retrieve), Call transfer, Conference Call setup, Page Send / Receive, Voice mail access, Last number Redial
- Hearing Aid Compatible (HAC) handset, Wall mountable (optional)
- Multiple powering options (802.3af compliant)
- Designed for power conservation: reduces power consumption for overall energy savings
- Small footprint (4” x 7.5” or 10cm x 20cm)

5304 IP Phone

Mitel 5312 IP Phone

The Mitel 5312 IP Phone is a full-feature, dual port, dual mode telephone that provides voice communication over an IP network. It features a back-lit LCD display screen, on-hook dialing and off-hook voice announce with handsfree answerback, and a large ringing and message indicator. It also offers 12 programmable personal keys for speed calling or one-touch feature access to a variety of other features. Ten fixed feature keys provide convenient access to features such as Conferencing, Redial, and many customizable user settings. The 5312 IP Phone supports Mitel Call Control (MiNet) and SIP protocols. The 5312 IP Phone can be used as a Teleworker phone.

- Additional Personal Keys and a LCD display
- Handsfree speakerphone operation (full duplex)
- The Message Light in the top right hand corner has separate message and ringing indicators.
- MUTE key replaces Microphone key
- MiNet message encryption support

5312 IP Phone

Mitel 5324 IP Phone

The Mitel 5324 IP Phone is a full-feature, dual port, dual mode telephone that provides voice communication over an IP network. It features a back-lit LCD display screen for display-assisted access to features, on-hook dialing and off-hook voice announce with handsfree answerback, and a large ringing and message indicator. It also offers 24 programmable personal keys for speed calling or one-touch feature access to a variety of other features. Ten fixed feature keys provide convenient access to features such as Conferencing, Redial, and many customizable user settings. The 5324 IP Phone supports Mitel Call Control (MiNet) and SIP protocols.

The 5324 IP Phone also supports modules such as the Line Interface Module, 5310 IP Conference Unit, and the 12 and 48 Button Programmable Key Modules. The 5324 IP Phone can be used as an ACD Agent Phone or a Teleworker phone.

- Additional Personal Keys and a LCD display
- The Message Light in the top right hand corner has separate message and ringing indicators.
- MUTE key replaces Microphone key
- MiNet message encryption support

5324 IP Phone

Mitel 5330 IP Phone

The Mitel 5330 IP Phone is a full-feature, dual port, dual mode enterprise-class telephone that provides voice communication over an IP network. It features a large graphics display (160 x 320) and self-labeling keys. The 5330 IP Phone offers 24 programmable multi-function keys, for one-touch feature access. Ten fixed feature keys provide convenient access to features such as Conferencing, Redial, and to many customizable user settings as well as navigational keys to access various screens and applications. 5330 IP Phones also support the 5310 IP Conference Unit.

Ideal for executives and managers, the 5330 IP Phone can be used as an ACD Agent or Supervisor phone, as well as a Teleworker phone.

5330 IP Phone

Mitel 5340 IP Phone

The Mitel 5340 IP Phone is a full-feature, dual port, dual mode enterprise-class telephone that provides voice communication over an IP network. It features a large graphics display (160 x 320) and self-labeling keys. The 5340 IP Phone offers 48 programmable multi-function keys, for one-touch feature access. Ten fixed feature keys provide convenient access to features such as Conferencing, Redial, and to many customizable user settings as well as navigational keys to access various screens and applications. With the appropriate programming, the 5340 IP Phone provides the same wide range of Hotel/Motel subattendant features as the Superset 4150 DNIC phone. 5340 IP Phones also support the 5310 IP Conference Unit.

Ideal for executives and managers, the 5340 IP Phone can be used as an ACD Agent or Supervisor phone, as well as a Teleworker phone.

5340 IP Phone

NetLink i640 Wireless Telephone

The NetLink i640 Wireless Telephone is the industry's most durable handset for workplace applications. Only SpectraLink combines innovative design, advanced manufacturing, and rigorous test processes to ensure handset durability. The six-ounce NetLink i640 is extremely simple to use, requires minimal training, and is durable enough to withstand the rigors of workplace use. Push-to-talk functionality is also available for broadcast communication between employees, eliminating the need for two-way radios or walkie talkies. The large earpiece seals out background noise and provides comfort for frequent or long calls.

NetLink e340/h340

The NetLink e340/h340 Wireless Telephones support a broad range of enterprise applications and are ideally suited for the general office, finance, or hospitality environments. This compact handset offers a rich set of features including a high-resolution graphic display, menu-driven functions, and messaging capability — all within a lightweight, ergonomic design. The NetLink e340/h340 phones provide exceptional voice quality and mobility at an affordable price. The h340 phone is similar to the e340 but provides additional durability and a backlit keypad, making it ideal for health care applications.

Telematrix 3000IP

The dual-mode TeleMatrix 3000IP Phone is a multi-line telephone that can operate either as a standalone device connected to a SIP (Session Initiation Protocol) service provider or as part of a Mitel ICP system using the MiNet protocol. The TeleMatrix 3000IP Phone is sold and supported by TeleMatrix, Inc.

Features of the telephone include

- Twelve keys, each with a built-in status indicator
- Seven fixed-function keys: Cancel, Hold, Redial, Transfer/Conference, Message, Speaker, Mute
- Automatic selection of prime line or ringing line (the bottom line key)
- Key selection of non-prime line
- Handset and ringer volume control
- Hands-free speakerphone operation (half duplex)
- Handset, speaker, and ringer volume controls (Up Arrow and Down Arrow)
- Message Waiting lamp
- Dual Ethernet port to provide connectivity to the LAN for both your telephone and computer
- Adjustable tilt mechanism
- The Message key can be used to call an embedded voice mail system, regardless of whether any messages are waiting in the user's voice mailbox. This functionality can be extended to a centralized voice mail system by programming the Message Key Routing For This Tenant option.
- Powered from an Ethernet connection compliant with 802.3af.

For more information about Telematrix 3000IP, refer to the technical documentation available at www.telematrixusa.com.

Accessories for IP Phones

- "Cordless Handset and Headset" on page 51
- "Mitel IP Programmable Key Modules" on page 52
- "Line Interface Module" on page 53
- "Mitel IP Paging Unit" on page 54

Cordless Handset and Headset

The Cordless Handset and Cordless Headset offer corridor mobility for Mitel 5330 and 5340 IP phone users. The Cordless Handset and Headset allow the user move freely within their office or adjacent offices (up to 300 feet from their desk) while still communicating from their desk phone.

Both cordless devices connect to an IP telephone through the cordless module, which attaches to the back of the phone. The cordless headset rests and recharges in a headset cradle that attaches to the side of the phone. The cordless handset recharges in the handset cradle.

The Cordless Devices Application provides access to the configuration settings and information screens that apply to the cordless module and accessories.

Features of the cordless accessories include:

- LED Indicators on the Cordless Module, Handset and Headset indicate both connectivity and charging status
- Eight hours talk time
- 43 Hours standby time
- Operating range of up to 300 feet (100 metres) in a typical office environment
- Out of communications range warning tone
- Support for two cordless devices (Handset and Headset) per Cordless Module
- DECT-based design: DECT 6.0 cordless technology provides higher quality voice transmission, density, and is less susceptible to interference compared to Bluetooth.

Cordless Handset and Headset

Mitel IP Programmable Key Modules

IP Programmable Key Modules (PKMs) add programmable keys to the 5324 IP phone. An IP PKM Interface Module installs in the back of the 5324 IP Phone to allow the 12- or 48-button IP PKM to connect to the IP phone without using an additional LAN port.

PKM keys can be programmed as feature keys, speed call keys, Direct Station Select keys, or line appearance keys. Each key has a Line Status Indicator that works the same way as those on the associated telephone. The keys can be programmed through the telephone.

Note: Up to two 48-button PKMs can be attached for a total of 96 additional keys.

12-Button PKM

48-Button PKM

Line Interface Module

Mitel's Line Interface Module for the 5324 phone:

- enables incoming and outgoing analog PSTN calls directly from an IP phone
- supports survivability (failover) in the event IP connection is lost
- provides Emergency dialing support, for phones such that emergency calls connect through the analog PSTN connection

The patented Line Interface Module enhances the Teleworker solution, and extends Mitel's resiliency strategy for IP communications from the core of the network to the desktop.

The Line Interface Module allows users to make and receive calls in the event of interrupted ethernet service or catastrophic LAN or ICP failure and provides automatic failover of voice communication without interruption of voice services.

The Line Interface Module provides a means to connect a 5324 IP phone directly to an analog line (PSTN trunk) in order to make and receive calls. It is a convenient solution that enables remote workers to have local breakout capability and can be used to access local emergency services. The Line Interface Module provides convergence reassurance for IP phones in the event of a service disruption or an IP network (LAN) failure.

The module allows a user to select a PSTN line to make and receive phone calls on an analog line without disconnecting from the IP connection.

Support for system failure switchover mode. If the Ethernet connection to the phone fails, the phone connects to an analog line to allow the business to still make and receive calls.

When the module is installed in the phone, users are able to access local emergency services

The Line Interface Module has two different modes of operation: LIM Mode and Failover Only Mode. These modes are determined by the System Administrator.

- LIM Mode (recommended for Teleworker/Remote configurations) allows the user to select an external analog line via a line key programmed on the 5324 IP phone. The analog line can be used at any time.
- Failover Mode, where you can use the Line Interface Module line only when the IP connection has failed. In Failover Mode, if the phone does not receive a response to 'keep alive' messages, the phone assumes the Ethernet link is down and automatically switches to analog mode.

Mitel IP Paging Unit

The Mitel IP Paging Unit is an optional module that provides paging functionality on the SX-200 ICP. DTMF tones can be transmitted to the IP Paging Unit for PA Paging.

The IP Paging Unit is installed as a stand-alone or a wall-mounted unit. Two LEDs provide basic status information. The unit connects to the LAN using an RJ-45 cable and is powered by a 24 VDC power adapter. Each IP Paging Unit supports one paging zone.

A third party remote paging amplifier (not included) connects to the paging unit and is powered separately.

IP0223

Conference Phones

Mitel 5310 IP Conference Unit

The Mitel 5310 IP Conference Unit is a full duplex, high-quality, conference unit that uses acoustic beam-forming technology for superior performance. The 5310 IP Conference Unit connects to a 5324, 5330, or 5340 IP Phone to provide full conferencing and telephony functionality. This eliminates the requirement for an additional LAN port.

The conference unit features:

- Acoustic beam-forming technology that controls near end, far end, and double talk, and also locates direction of speech
- Visual confirmation that the Conference Saucer has picked up the speaker's voice
- Module and soft keys for Conference Controller Application for the 5324, 5330 and 5340 IP Phones.

Mitel 5310 IP Board Room Conference Unit and 5324 IP Phone

Digital Phones

Mitel offers the following Superset 4000 series digital business telephones:

- Superset 4015 multi-line telephone set with a LCD display
- Superset 4025 multi-line telephone set with an enhanced LCD display

Superset 4025 Telephone

The Mitel Superset 4025 connects to a DNI card in the Peripheral unit. It is a multiline, digital telephone with

- 20-character alpha-numeric liquid crystal display (LCD) with contrast control
- Three softkeys for feature access
- 14 line keys, each with a built-in line status indicator
- Eight fixed-function keys: SuperKey, Cancel, Hold, Redial, Transfer/Conference, Message, Microphone, and Speaker
- Automatic selection of prime line
- Key selection of non-prime line
- Handsfree operation (half-duplex)
- Handset, speaker, and ringer volume controls
- Ringer pitch control
- Message Waiting lamp

The Superset 4025 supports PKM IMs for connection to additional devices.

Mitel Superset 4025

5540 IP Console

The 5540 IP Console offers easy-to-read liquid crystal display, hardkeys for the most often performed functions, and softkeys for situation-dependent features, all contained in a compact package.

The Console can also be used as an economical option for a department secretary handling calls for a group of people, a maintenance console for troubleshooting and a programming console for customer data entry.

Attendant/Secretarial. The console's four-line, 80-character liquid crystal display (LCD) shows time and date, and call status information including the names of callers within your organization, call source and destination, and number of calls waiting to be answered. The 14 hardkeys are dedicated to standard attendant activities - answering calls, putting calls on hold, blocking calls, paging, releasing calls to their destination or hanging up, canceling dialed digits, checking the status of trunk groups, and performing Attendant functions such as setting time and date, and switching to night service. Ten softkeys control access to the attendant features through blank keys on the console. The name of the feature associated with a particular key is shown on the screen only when it is available for use.

Maintenance. All maintenance activities - system level functions (such as setting time and date), reporting functions such as configuration, alarm status, and the display and clearance of device errors), maintenance log functions, and traffic measurement - can be done through a Console. When the Console is being used as a maintenance console, the softkeys displayed are the ones available on a maintenance terminal. Maintenance access is password controlled.

Customer Data Entry. All Customer Data Entry (CDE) - initial system installation, moves, adds and changes, and system expansion - can be done through the Console. When the Console is being used as a CDE Console, programming is done by softkeys. As in maintenance, access to CDE is password-controlled.

The SX-200 ICP supports the 5540 IP Console which features

- Up to nine line appearances
- Eight call hold positions
- English, French, and Spanish operation
- Two headset jacks

The 5540 IP Attendant Console has a tilt display and two blank firmkeys for extra features. The console supports up to two PKM 48 devices. The 5540 IP Console does not have printer port, but printing can be requested if the printer is connected to the same network as the console.

5540 IP Attendant Console

Accessories for Superset 4000 Series Digital Phones

Mitel offers the following accessories for the 4000 Series phones:

- “Mitel Programmable Key Module 12” on page 59
- “Mitel Programmable Key Module 48” on page 60
- “Mitel PKM Interface Module” on page 60
- “Mitel Analog Interface Module” on page 61

Mitel Programmable Key Module 12

The Mitel Programmable Key Module (PKM) 12 provides 12 additional personal keys for Superset 4025 Digital Phones. They can be programmed as feature keys, speedcall keys, Direct Station Select keys, or line appearance keys. Each key has a Line Status Indicator that works the same way as those on the associated telephone. The keys can be programmed through the telephone.

The PKM 12 connects to a Superset 4025 Digital Phone through a Mitel PKM Interface Module (IM). The PKM IM is installed separately at the base of the telephone and is only compatible with Superset 4025 Digital Phones.

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Mitel Programmable Key Module 48

The Mitel Programmable Key Module (PKM) 48 provides additional feature keys for Superset 4025. They can be programmed as feature keys, speedcall keys, Direct Station Select keys, or line appearance keys. Each key has a Line Status Indicator that works the same way as those on the associated telephone. The keys can be programmed through the telephone.

The PKM 48 connects to a Superset 4025 Digital Phone through a Mitel PKM Interface Module (IM). The PKM IM is installed separately at the base of the telephone. A second PKM 48 can connect to the first to provide 48 additional feature keys for a total of 96 additional keys.

Mitel PKM Interface Module

A Mitel PKM Interface Module (IM) is installed in a Superset 4125 telephone to connect a PKM 12 or up to two PKM 48 devices.

Mitel Analog Interface Module

The Mitel Analog Interface Module (AIM) allows the connection of one Mitel IP Programmable Key Module 12 or up to two Mitel IP Programmable Key Module 48s and one two-wire analog device such as an analog telephone, fax machine, or modem (to a maximum loop length of 50 feet and a ringer load of up to 2 REN). This interface module allows simultaneous use of both the Superset 4000 series telephone and the analog peripheral.

Superset 4125 telephones require a power adaptor for the interface module connection. A power adaptor must be ordered if the PKM 48 or PKM 12 is to be installed on a Superset 4125 telephone.

Message Waiting is not supported by sets connected to an AIM. Devices with a Z-type REN are not to be connected to an AIM. The analog device must signal using DTMF tones; devices that use dial pulse signaling are not supported.

Music-On-Hold/Pager Unit (DMP)

The SX-200 ICP provides built-in Music-On-Hold (MOH) and Pager capability with the MX controller only. For extra functionality, such as multiple MOH sources, you can install a Music-On-Hold/Pager Unit which interfaces a standard SX-200 ICP 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 ICP system and does not require a separate power source. A single 25 pair amphenol connects to the SX-200 ICP via the main distribution frame. A single LED indicator provides basic status information. The unit can be wall-mounted next to the SX-200 ICP.

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.

Music-On-Hold/Pager Unit

Power Accessories

Power accessories are required for the following:

- “SX-200 ICP” on page 63
- “IP Phones” on page 63
- “Peripheral Cabinets” on page 63

SX-200 ICP

The SX-200 ICP requires an Uninterruptible Power Supply (UPS); a reserve power supply for the control cabinet and digital peripheral cabinets comprising of a battery pack, a charger, and an inverter. The UPS backup time is dependent upon the unit selected and the capacity of the batteries provided. The unit must be able to provide 115 Vac at 15 A. The unit must provide the following power for each type of controller:

- CX and MX controllers: 60 W
- CXi controller: 250 W (assuming that all 16 ports are feeding power to the telephones)

Marketing and sales literature, available from authorized representatives, identifies several uninterruptible power supplies (UPS) that are compatible with the SX-200 ICP system.

IP Phones

Power is provided to the IP Phones by

- an external supply (48-volt PoE brick) can be used by 5300 series IP Phones
- a multi-port Ethernet Inline Power Module (such as the PowerDsine 24PT Inline Power Unit)
- Layer 2 switches with integral power feed (built into the CXi controller)

Peripheral Cabinets

Power is provided to the Peripheral cabinets by the

- Bay Power Supply
- System Fail Transfer

Bay Power Supply

The Bay Power Supply unit is a rack-mounted AC-to-DC converter that furnishes the required operating voltages for circuit cards in the 96-port digital bays. The supply also contains a ringing voltage generator.

System Fail Transfer

The SFT is an optional, stand-alone, wall-mounted device that connects to the system’s peripheral cabinet or main distribution frame (MDF). The device supplements the SFT capabilities provided internally on the analog card within the SX-200 ICP Controller. Both the stand-alone device and the internal SFT allow preselected DTMF or rotary telephones to be connected directly to CO trunks in the event of system failure in the system.

When the system goes into SFT mode, the SFT unit connects up to six internal POTS telephone extensions directly to the CO, bypassing the system completely. Each SFT can control six circuits, and up to four SFTs can be daisy-chained together for each zone, providing security for 24 internal extensions.

The SFT switches to SFT mode under the following conditions:

- Failure of the system power converter
- Failure of the system main control
- Interruption of the system AC power
- Loss of the fiber link between the main control and peripheral cabinets.

Power Supply: All power for the SFT unit is provided from the -48 Vbat source on the system. A source of -12 V powers the electronic circuitry on the card. This supply is derived from the -48 V input and powers all the SFT circuitry except the transfer relays. The relays are powered by a transistor-regulated -41V source, also derived from the -48 Vbat input. Thus, in the event of Vbat varying between the standard -42.5 V to -56.5 V, the current drain remains constant.

Transfer Relays: Each circuit in the SFT uses a four form C relay to transfer between normal and SFT modes of operation.

Loop Detector: When a transfer relay enters SFT mode, the loop detector connects in series with the loop between the extension and CO trunk facility. This circuit prevents the extension from returning to normal operating mode before an SFT mode call is completed. When the SFT mode call is completed, the extension is returned to normal operating mode.

SFT Control Leads: The transfer control sensor on the SFT senses a loop closure across the SFT and SFT return (SFTR) leads. When a loop closure is sensed, the power to the relays is removed, the relays are released, and all circuits enter the transferred state.

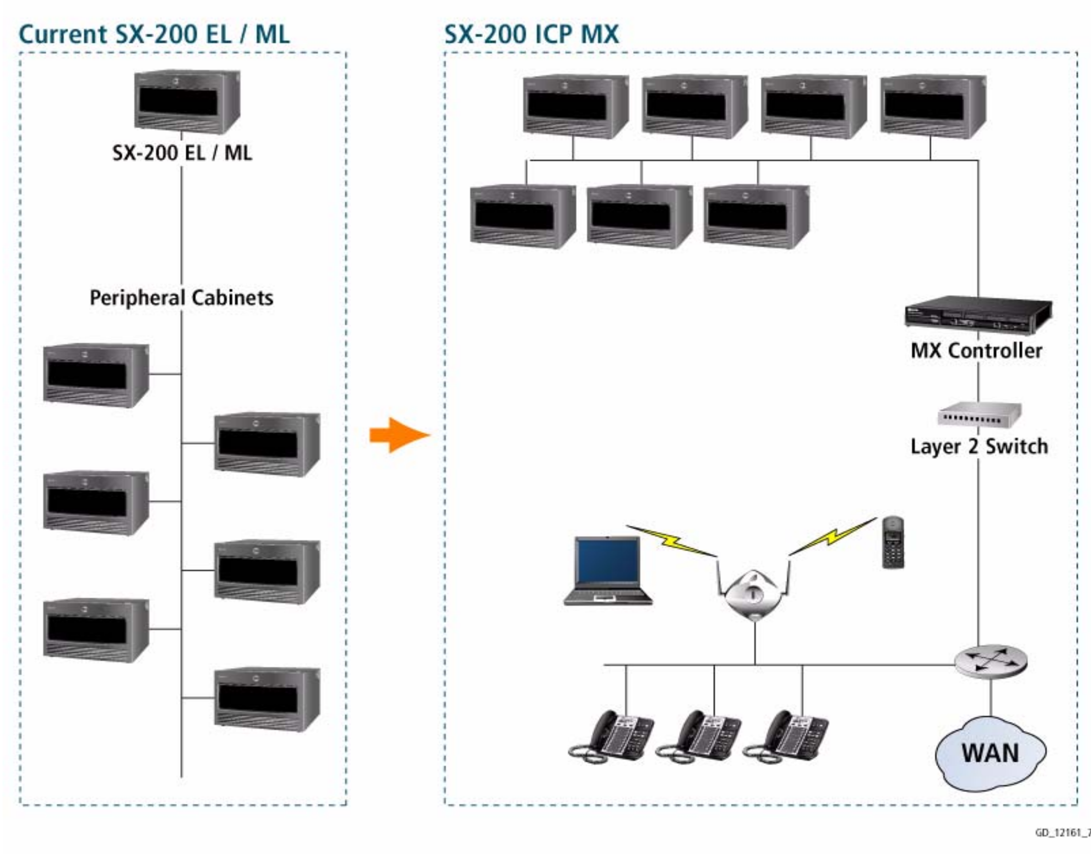
Power Consumption: The total current drain for the SFT is typically 80 mA.

Power Dissipation: The power dissipation, in watts, is shown below.

Power Supply	TYP. (Watts)	TYP+20%
-48Vbat	3.18	3.81
@Vbat=-56 V	3.71	4.45

Migration

Customers can migrate from an SX-200 EL/ML to an SX-200 ICP with an MX controller running Release 2.0 software.



Note that the SX-200 ICP MX controller provides support for certain feature packages and phones only following a migration from an SX-200 EL/ML system. However, if you purchase an SX-200 ICP MX controller, these particular packages and phones are not supported. For more information on packages and phones supported following a migration, refer to “Comparison of Phones supported by MX, CX/CXi and AX” on page 39 and the individual phone descriptions in “Desktop Devices” on page 39.

Management and Maintenance

The SX-200 ICP is managed and maintained via

- “Customer Data Entry (CDE)” on page 67
- “Maintenance Terminal” on page 69

Customer Data Entry (CDE)

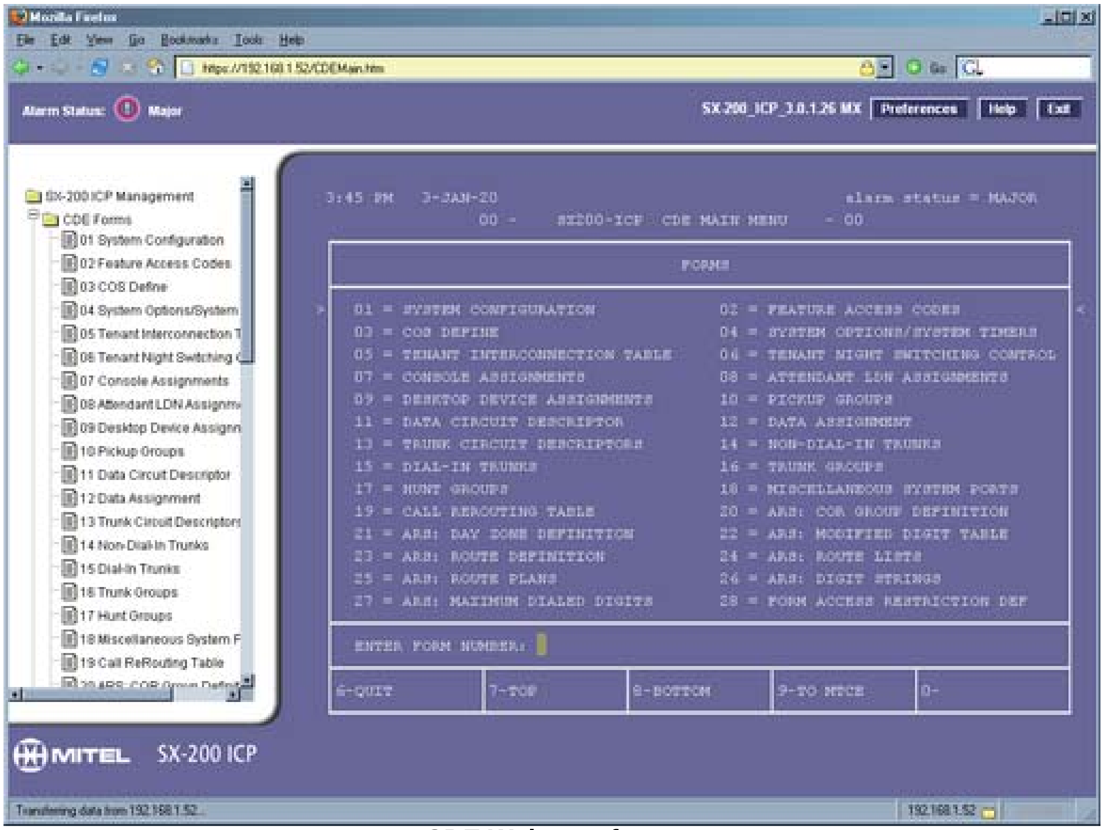
After the SX-200 ICP is powered up, the system is ready for programming. Customer data entry is accomplished from:

- ASCII CRT terminal (VT100™ compatible): used for local programming only; remote programming requires a PC. The terminal connects via an RS-232 connection to the Maintenance connector on the SX-200 ICP controller.
- a PC with the following:
  - Windows® 95, 98, NT, XP or 2000 Professional
  - for serial connections, a VT100 emulator or other communications program
  - for remote connections, the Mitel Telnet client (supplied with software), or any other secure Telnet client that supports SSL/TLS
  - a serial port and serial cable
  - a Network Interface Card (NIC) and Ethernet cable
- Attendant console: for on-site customer data entry on the SX-200 ICP MX only

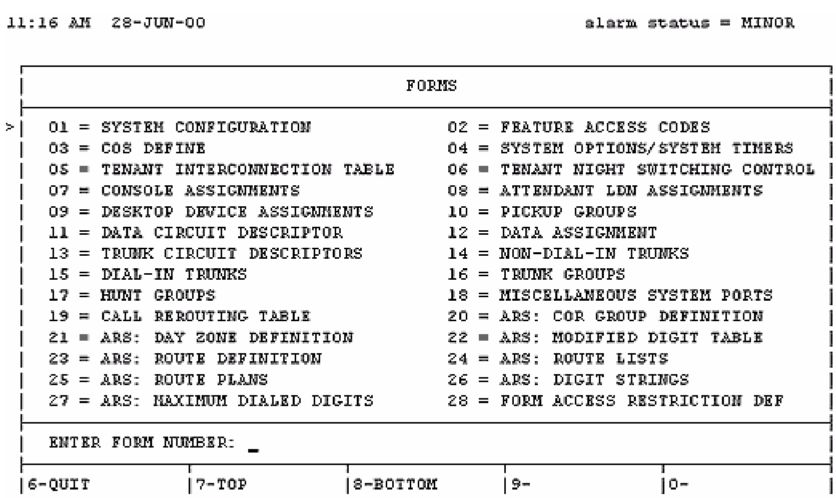
The console LCD guides the installer or maintainer through the data entry procedure by displaying a series of prompts and listing the required steps to be performed. The console displays four lines of 80 characters each. The two top lines display the steps to be taken; the two bottom lines display the prompts that define the 10 function keys on the system console.

All CDE programming is done through a series of English language programming forms, with each form made up of a number of data fields. System features, operation, calling restrictions and toll control are customized to your needs by entering the appropriate data or enabling the appropriate options in the CDE forms. All CDE data is entered using the softkeys on the console, or the keyboard on the terminal and PC.

To ensure the system always meets your changing requirements, authorized personnel can modify the system’s CDE entries at any time by altering the entries in the appropriate forms.



CDE Web Interface



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Maintenance Terminal CDE Forms Display

Maintenance Terminal

The objectives of the maintenance functions are to isolate a fault to a replaceable card or module. Maintenance functions can be performed from the attendant console, an RS-232 terminal, or a PC via Secure Telnet. A connector for a remote maintenance terminal is provided on the rear panel of the cabinet.

Information presented on the maintenance terminal includes

- System date, system time
- Current system alarm level
- System identification number

Information presented on the four-line console LCD display includes

- System date, system time
- Command entry line

When the console is used for maintenance, the maintenance output data is displayed on the LCD.

Alarm Indication

The system has four alarm levels: no alarm, minor, major, and critical. Minor alarms indicate problems affecting a portion of the system, such as failure of a line or trunk circuit. Major alarms indicate problems causing a system-wide degradation of service. Critical alarms indicate serious problems that cause automatic activation of System Fail Transfer. The system maintainers can adjust the alarm thresholds to suit the customer's requirements. The thresholds represent the alarm level trip points; that is, the precise divisions between the alarm levels. The thresholds are simple percentages, indicating availability: the number of working devices is compared to the number of programmed devices. The Critical Alarm threshold is not a percentage, but is a precise numerical value. When the number of available devices falls below this number, a critical alarm is raised. The system can be programmed to send a set of generated logs to an email Address or FTP Server.

Alarm LEDs

The SX-200 ICP controller has LEDs for Minor, Major, and Critical alarms. All peripheral cards such as lines, trunks and receivers in Peripheral cabinets have one red alarm LED. The system lights the LED if the card fails a diagnostic test, or if the card is installed in a wrong or non programmed card slot.

Alarm Status Display

The maintainer can display current system alarm levels for the entire system or for separate categories. The categories are

- Lines
- Trunks
- DTMF receivers

Configuration Report

The configuration report allows the maintainer to display the system configuration showing the location of major devices down to the level of the modules installed on cards in the peripheral bays.

Copy Database

The maintainer can make a backup copy of the system database onto a storage medium on a PC.

Customer Data Entry (CDE) Backup and Restore

This feature allows customer data to be dumped onto a storage medium on a PC, and also allows new generic software to be loaded into the system from a PC.

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 ICP Maintenance connector. You can log in remotely via a secure telnet session or use a direct plug in via a serial connection. Backup copies of the database can be stored on a PC or an FTP server.

Database Storage on Loss of Power

The customer data entry database is stored in the internal flash card in the SX-200 ICP. Customer data entry information can also be kept on a remote PC for retrieval in case of major system failures.

Device Error Analysis Statistics

The SX-200 ICP record errors detected during the operation of datasets, HDLC links, and T1 trunks. Maintenance personnel can generate statistical reports based on this data.

Device Status Report

The maintainer can display the status of any peripheral circuit or circuits by entering a command at the maintenance terminal. The information displayed includes: circuit location, circuit type, call processing state and maintenance state.

Diagnostic Log Files

A file of the major occurrences in the diagnostic system is maintained in internal flash memory. This file can be directed to the RS-232 maintenance terminal, the attendant console, a printer, an e-mail address or an FTP server.

Remote Maintenance Administration and Test (RMATS) Access

RMATS allows personnel at a central maintenance center to access the SX-200 system and retrieve maintenance data or make programming changes.

Remote Printing of CDE Reports

CDE reports are generated and captured on the remote terminal so the user can obtain a softcopy of the reports and print them.

Remote Software Download

Remote software download allows a system software upgrade to be performed from a PC. System software to be loaded onto the maintenance PC can be obtained by subscribing to the Mitel subscription service. System Option 109 - Remote Software Download must have been purchased and enabled.

Remove from Service, Return to Service

The maintainer can remove a line, trunk, or receiver circuit from service for maintenance. Removing a circuit from service makes it inaccessible to call processing; it remains so until the maintainer returns it to service.

Show, Set Date

The maintainer can show and set the system date from the maintenance terminal.

Show, Set System Time

The maintainer can show and set the system time from the maintenance terminal.

Superset Firmware Download

The Superset 4025, Superset 4125, and the Superset 4150 telephones are outfitted with flash ram that contains firmware. The firmware can be upgraded using the firmware download maintenance commands. This makes it possible to download new firmware in the field to add new functionality to the Superset telephones.

System Logging Facility

The SX-200 ICP systems keep a system event log. Each time the maintenance state of a device changes, or a major event occurs such as a card installed in the wrong slot, the system generates a log report. These log reports can be read, printed, or deleted from the maintenance terminal or console.

Applications

In the SX-200 ICP system, groups of features have been combined into feature packages designed to meet the specialized needs of small to medium size enterprises. In addition to these feature packages, you can purchase a set of applications that provide superior voice capabilities.

For information, refer to

- “Feature packages” on page 73.
- “Embedded applications” on page 76.
- “Applications” on page 81.

Feature packages

- “ACD TELEMARKETER” on page 73
- “Centralized Voice Mail” on page 74
- “Hotel/Motel” on page 74
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ACD TELEMARKETER

The ACD TELEMARKETER® application is a purchasable option.

The ACD TELEMARKETER application is an advanced Automatic Call Distribution (ACD) system that is fully integrated with the SX-200 ICP system, and designed with the power and performance needed to ensure satisfaction in the most demanding call center environments. For maximum efficiency, all ACD personnel use Mitel 5000-series IP telephones, Superset 4015, Superset 4025, Superset 4125, Superset 4150, Superset 430, Superset 420, and Superset 410 telephones programmed with special displays and softkeys.

The heart of the ACD TELEMARKETER feature is the ACD PATH, an innovative call routing design that guides incoming calls through the system. The ACD TELEMARKETER feature also uses predictive overflow to keep call queueing time to a minimum.

The ACD TELEMARKETER feature includes real-time displays via standard asynchronous datasets and ASCII terminals. Thirteen displays encompass every area of the ACD operation.

The purchasable option, Maximum ACD Agents, enables the maximum number of ACD agents that can be logged in concurrently. This maximum number is from 0 through 100, in increments of 5.

Centralized Voice Mail

The Centralized Voice mail feature allows a network of IP PBXs (SX-200 ICP, 3300 ICP and SX-200 IP Nodes) to share a single voice mail facility with Message Waiting Indication at all network sites. The Centralized Voice mail feature works with any voice mail interface (for example ONS or DNIC). The Centralized Voice mail feature can also be installed on another SX-200 ICP or a Mitel 3300 ICP and accessed over IP Trunks. The Centralized Voice mail feature is a purchasable option.

When a call is forwarded to voice mail from an extension on the same PBX to which the voice mail device is connected, the caller's extension number, the forwarding extension number, and the call forward reason are passed to the voice mail system. When the forwarding extension is on another PBX, Centralized Voice mail is used to pass this information between PBXs to the voice mail system.

The transfer of information is done over tie trunks between each PBX.

Hotel/Motel

The Hotel/Motel application is a purchasable option.

Hotel/Motel features speed up guest check in and check out, and allow you to manage your rooms efficiently, wake up guests on request, control guest telephone privileges, recover the cost of guest calls (SMDR supported), and notify guests of their messages. These functions are all handled by the Attendant or front desk clerk using the SUPERCONSOLE 1000 Attendant Console or the Front Desk Terminal.

The Front Desk Terminal interfaces to the SX-200 ICP through sockets. If efficient billing is in place, the Front Desk Terminal provides a low-cost alternative to a Property Management System (PMS) for smaller Hotel/Motel operators (in the 40 - 90 room size). It is ideal for fast check in and out, guest location, and housekeeping functions.

For computerized control and monitoring of Hotel/Motel functions, the system can interface to a property management system (PMS).

The Hotel/Motel feature package integrates standard system features with custom hotel/motel features. The system can also interface with a property management system (PMS). The Hotel/Motel and property management features are purchasable options.

The SX-200 ICP can operate with either the Hotel/Motel Package or the Property Management System package, but not both packages on the same system.

Mitel Express Messenger

Mitel Express Messenger delivers affordable, easy-to-use voice mail and auto attendant capabilities on a single card that plugs into an Mitel SX-200 peripheral cabinet.

Each card provides up to eight voice mail ports. More than one Express Messenger may be installed in a system; however, each Express Messenger will operate independently. For example, multiple Express Messenger systems could be installed to provide voice mail support to several tenants. Express Messenger typically supports 10 to 25 users per port, depending on the usage of Express Messenger.

Key features of Express Messenger include:

- Message waiting indicators on Superset and Mitel IP telephones
- Mnemonic prompts, such as "Press P for Play, D to discard," provide an intuitive navigation system for users
- Ability to record a name and personal greeting to each mailbox user
- Ability for each user to record a personal greeting set for a specific number of days (with automatic expiration)
- Ability to record calls
- Password protected mailboxes
- Unlimited message length
- Ability to save messages as well as set parameters for automatically purging saved messages
- Message erase, reply, forward, rewind/hold/ fast forward
- Ability for a message to receive priority placement within mailbox
- Callers with the ability to review, re-record and append their message before sending
- Call forwarding

Property Management System

A Property Management System (PMS) provides a center for managing a hotel business. The PMS system can provide reservation control, centralized accounting and billing, and call logging.

IP-enabled PMS applications can communicate with the SX-200 ICP via a Telnet connection. Applications that require a serial interface must use a third-party Serial-to-IP port converter to connect to the network.

When information about a guest is changed at the PMS system, messages are sent to the PBX via the PMS. Similarly when information about any guest is changed on the PBX, messages are sent to the PMS system. In summary, the PMS is an intermediary for passing messages from the PBX to the PMS system and from the PMS system to the PBX.

Station Message Detail Recording

Station Message Detail Recording (SMDR) or “call detail recording” is an integral part of the system. It generates a descriptive call record for every incoming and outgoing trunk call made via the system. These call records can be routed to an RS-232 port for processing or printing. They allow the customer to evaluate the use of the system’s trunks and determine whether the quantity and type of trunks are the most economical mix for the traffic being handled by the system. In addition, the customer can analyze the use of the trunk network by corporate personnel. Misuse can then be corrected through modifications to the toll control assignment. The SMDR feature package includes Trunk SMDR, Data SMDR and ACD TELEMARKETER Reporting System SMDR.

Tenanting

Economy of scale makes sharing system services practical. Using the tenanting features, up to 25 small businesses, or departments of a larger business, can share the services of an SX-200 ICP system. Logically, the system can be divided into up to 25 separate PBXs, each providing its tenant with customized features and services.

Consoles, night bells, Music-on-Hold, 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 system 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 display telephones as subattendant positions. The main console position could be handled by these telephones, using line buttons to receive tenant recalls.

Traffic Measurement and IP Trunk Performance

Traffic Measurement involves collecting data about the system (measurement) and interpreting this data (analysis) to optimize system performance. Once traffic measurement has been started in the PBX, it continues automatically until changed or stopped. Traffic measurement produces a single report for the system. The report includes all tenants, if a tenant service is provided. Statistics collection is performed to measure the performance of IP trunks in the system.

Embedded applications

The following applications are embedded in the SX-200 ICP system.

- “Embedded Voice Mail” on page 77

The following application interfaces are supported on the SX-200 ICP system:

- Mitel Application Interface (MAI) Package: This purchaseable option allows MITEL computer-based applications to access the system features. MAI is used in conjunction with an external host computer connected via IP.

Embedded Voice Mail

The SX-200 ICP includes an integrated highly-featured voice mail system. Up to 24 ports (MX controller) or 16 ports (CX/CXi controller) are available for voice mail calls with support for a maximum of 750 mailboxes and five hours of storage time with an internal compact flash (256 Mb). The storage time can be increased by replacing the internal flash with a hard drive. The features provided by the voice mail system are described in Table 3 on page 77.

Table 3: Voice Mail features

Feature	Description
Personal Greetings/Name	Each mailbox user can record subscriber name and a personal greeting.
Message Prologue	Informs subscribers when they access their mailbox how many new or saved messages they have (if any).
Temporary Greeting	Each subscriber can record a personal greeting set for a specific number of days (with automatic expiration).
Password Protected Mailboxes	Access to subscriber mailboxes requires a password. Password length system-wide can be from three to six digits. (Default is four digits.) Callers have three chances to enter a valid password before they are disconnected.
Message Envelope	Played prior to beginning of each message, containing priority type, date, and time (including caller identification for internal and external calls). Mailboxes can be individually configured to play the envelope only in response to a key press – i.e., at the request of the subscriber.
Message Length	Unlimited message length with a 5-minute continuation prompt. Minimum message length is two seconds
Saved Messages	A subscriber may save messages. They are automatically purged from the system after 15 days (or as reprogrammed) or you can specify that saved messages are never deleted. New messages are never purged automatically. The saved messages are played in last-in first played order
Message Review	Allows immediate replay of a message, including message envelope (timestamp, calling party information).
Message Erase	Allows immediate deletion of a message from the system. The message cannot be subsequently restored; deletion is immediate and permanent.
Message Reply	Allows immediate reply to a message received from another internal mailbox subscriber.
Message Forward	Allows messages to be forwarded to other subscribers and distribution lists with or without a pre-pended comment.
Message Rewind/Hold/Fast Forward	Allows subscribers to rewind, fast forward, or pause messages for several seconds.
Message Keep/Skip	Allows subscribers while listening to a message to advance to the next new message (if any). Each new message played is marked as “saved.”
Multi-Level Auto Attendant	Allows a hierarchical menu to be programmed on the auto attendant providing callers with better self-service access to the person or department they are calling. In an MLAA system, callers reaching the Auto Attendant are routed from the main menu through to one or more additional sub menus until their call is answered. You can program up to 10 multi-level menus, each with its own greeting and prompts.
	Page 1 of 4

Table 3: Voice Mail features (continued)

Feature	Description
Urgent Messages	The message receives priority placement in the listener's mailbox.
Private Messages	The message cannot be forwarded to another subscriber's mailbox.
Certified Messages	On internal calls, the sender is notified when the recipient has read the message.
Message Record/Send Actions	Callers have the ability to pause during recording, review, re-record, and append to a message before sending it. A message can also be cancelled prior to sending.
Message Addressing	Subscribers can address messages to multiple recipients and hear the recipient's name played back to confirm valid entry of mailbox numbers.
Forward Voice Mail to E-Mail	Allows users to forward voice messages, including Record-a-Call messages, to an E-mail address. Users can choose to manually forward voice messages, or automatically forward all voice messages.
Memo	Subscribers have single-digit access to send a message to their own mailbox, for future reminders and memo-type messaging.
Standard Unified Messaging	Enables the SX-200 ICP to manage e-mail messages using SMTP or IMAP. For more information, refer to “Unified Messaging - Standard” on page 85.
Message Notification	The subscriber is notified that they have received a message by the message light on their phone (MWI), and optionally by setting the notification type to one of the following options, which causes the voice mail system to call: • the mailbox’s associated extension number, for analog phone extensions or phones without a message light (prompts called party to log into their mailbox). • an outside number (prompts called party to log into their mailbox). • a message pager (plays an audio message indicating messages are waiting). • a tone-only pager (simply hangs up after a far connection is made). • a digital pager (plays DTMF digits corresponding to a system-wide callback number along with the specific mailbox number). The system administrator may change notification options. The mailbox owner may also modify them if the system administrator grants permission. In addition to the notification type, the phone number and schedule are configurable. The schedule determines whether paging occurs: • around the clock, regardless of the business schedule. • only during open business hours. • only during closed business hours. • never (disabled until the schedule is changed to one of the three previous schedule options). Finally, a mailbox may be configured to do non-MWI notification only in response to urgent messages (as opposed to all messages). By default, a busy or no answer condition detected on a notification call results in two additional retries occurring at 15-minute intervals. All notification results are posted to the system log file.
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Table 3: Voice Mail features (continued)

Feature	Description
Distribution List, Broadcast Message	Distribution lists can be set up for global (system-wide) use to make it easier to send messages to a group of people. Users can also set up distribution lists for their personal use. The system provides forty-nine programmable global lists numbered 001 to 049. A fiftieth list (the Broadcast list) numbered 000 contains all programmed extension-type mailboxes. The system creates the list automatically; it cannot be modified. All mailbox owners can use the global lists but only the administrator can change the programmable ones. Personal distribution lists are numbered 050 to 059. Global distribution lists can be created and managed by telephone or via CDE. Personal lists can be created and managed by telephone only - see the Voice Mail User Guide for details.
New mailbox Tutorial	The system guides the user through the steps required for initial configuration of mailbox, including specification of a (non-default) passcode and recording of a personal greeting and name.
Mailbox Types	The following mailbox types are available: **Extension** - the auto-attendant transfers a caller to the mailbox's associated extension. If the called party is busy or does not answer, the caller is prompted to leave a message in the mailbox. The extension mailbox may be linked to other mailboxes for transfer only (dual mailboxes). This permits the caller to transfer to other mailboxes in the same department. **Message-Only** - the auto-attendant does not attempt a transfer but immediately prompts the caller to leave a message in the mailbox. **Transfer-Only** - the auto-attendant transfers a caller to the mailbox's associated extension but does not take a message if the called party is busy or does not answer. **MENU (Menu Tree)** - allows you to set up a hierarchical menu structure for MLAA operation. **Information-Only** - the auto-attendant only plays the mailbox greeting; no transfer or prompt to leave a message occurs. **Administrator** - for accessing administrative functions such as greetings recording. The Hospitality Features provide two types of mailboxes for hotel/motel applications: guest mailbox and front desk mailbox. A guest mailbox provides a guest with basic voice mail functionality. A front desk mailbox allows the front desk attendant to administer the guest mailboxes.
Property Management System (PMS)	A Voice Mail feature that allows the hospitality industry to connect their Hotel PMS systems to the voice mail application via a serial interface. This serial connection allows the PMS to notify voice mail when a user checks in or checks out. Based on this information the voice mail system either checks in or out a mailbox for the guest.
Softkey Integration	Users with Mitel telephones can press softkeys instead of dialing codes to select Mitel Express Messenger menu options. For example, to listen to message, a user can press the Play Message softkey instead of dialing the digit 7.
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Table 3: Voice Mail features (continued)

Feature	Description
Dual Mailboxes	A transfer-only mailbox can be linked to the same extension as an existing extension-type mailbox. This enables, for example, a single mailbox for a sales department and the sales manager.
Personal Contacts	Personal Contacts allow users to store alternate numbers where callers can contact them instead of leaving a message. Callers are prompted in the greeting to press a key to have their call transferred to the alternate number—they are never told the number. Users can program up to ten (10) Personal Contacts.
Distribution Lists	A Distribution List allows mailbox subscribers to send messages to several people at one time. There are two types of distribution lists: personal lists and global lists. Personal lists are set up by individual subscribers for their own use. Global lists are for use by all subscribers and are set up using the Distributions List Form. Only the system administrator can set up or change the global lists. Up to 49 global lists (001-049) can be created. A fiftieth list (000) is already set up to broadcast messages to every local mailbox. Users can create up to 10 personal lists (050-059). Each distribution list can have up to 750 contacts.
RAD Greetings	This feature provides the ability to play recorded greetings through an embedded voice mail port (RAD port), eliminating the need for external tape machines or other audio-playing devices. RADs are commonly used to automatically answer incoming calls and deliver pre-recorded messages such as “All of our representatives are busy helping other callers, please continue to hold to maintain your call priority.” When the RAD message finishes playing, the caller usually hears music-on-hold while waiting for an agent to become available. RAD messages may also give the caller information, which answers their questions, thus resulting in a 'good’ abandoned call. They may also provide advertising or promotional information to callers while they’re waiting for someone to take their call.
Record a Call Option	Allows users and ACD agents to record telephone conversations to be reviewed later. The message is saved in Voice Mail. Recorded calls can be replayed to ensure accurate information was derived from the conversation or perhaps to monitor harassing telephone calls. When a user activates this feature, it is accomplished in silence.
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Applications

The SX-200 supports the following applications:

- “Mitel Teleworker Solution” on page 81
- “Mitel Customer Interaction Solutions (formerly 6100)” on page 82
- “Mitel Call Accounting” on page 83
- “Mitel Speech-Enabled Applications (formerly 6500)” on page 83
- “Unified Messaging - Standard” on page 85
- “MiTAI” on page 86
- “Mitel NuPoint Messenger IP” on page 86

Mitel Teleworker Solution

The Mitel Teleworker Solution enables businesses to easily enjoy the benefits of teleworking through a low-cost, "plug and work" solution that extends the corporate network to virtually any location. Businesses can now benefit from reduced overhead and increased employee retention while users can be more flexible and productive in how they work.

The Teleworker Solution is easily implemented using standard Mitel IP phones (5212, 5224, 5312, 5324, 5330, 5340) Navigator, and Your Assistant softphones.

The Teleworker Solution provides:

- Flexibility and familiarity through using a standard Mitel IP Phone
- Transparent access to corporate voice and data services
- High levels of security
- Seamless integration with Windows and Macintosh clients for access to the corporate data network
- Scalability with support for large numbers of remote workers
- Plug-and-work simplicity

Mitel Customer Interaction Solutions (formerly 6100)

Mitel Customer Interaction Solutions combine robust communications platforms, Automated Call Distribution (ACD), and a modular suite of feature-rich, web-based applications for streamlining contact center management, and enabling advanced multimedia customer contacts. The Mitel Customer Interaction Solutions portfolio includes:

- **Mitel Contact Center Management** is a browser-based application that provides real-time and historical monitoring as well as agent forecasting.
- **Mitel Contact Center Management Enterprise Node** is an add-on product to Contact Center Management that provides multi-switch (remote and/or co-located) enterprise-wide historical reporting and real-time monitoring.
- **Mitel Interactive Contact Center** is an application that allows you to control agent and queue states instantly and easily via the Contact Center real-time display. Interactive Contact Center integrates with Contact Center and Mitel 6150 Multimedia Contact Center to provide virtual queuing.
- **Mitel Contact Center Scheduling** is an application that integrates with the Contact Center Forecasting functionality to provide automatic agent scheduling, based on business rules and required skills.
- **Mitel Schedule Adherence** is an add-on product to Contact Center Scheduling that allows you to see what agents are doing in relation to what is scheduled and quickly identify areas of non-adherence.
- **Mitel Agent Portal** is an application that displays caller information on agent desktops via a number of different applications that can be configured to 'pop' or display automatically on any desktop.
- **Mitel Multimedia Contact Center** is an advanced contact distribution package that integrates with Microsoft® Exchange 2003 to route emails, chats and faxes to the longest idle agents in MS Outlook.
- **Mitel Intelligent Queue** is an a browser-based recorded announcement solution that provides standard recorded announcements, intelligent messaging capabilities, routing and callback.

Mitel Contact Center Solutions are described in detail in the Customer Interaction Solutions General Information Guide.

Mitel Call Accounting

Call Accounting allows you to track your telephone system costs and summarize them in reports so you can manage your telephone expenses and activity effectively. Call Accounting allows you to identify unauthorized telephone calls and determine whether you are using the most cost-effective carriers for your trunks.

Call Accounting:

- tracks and reports call activity and costs
- tracks toll fraud via customer-defined specifications
- integrates with Mitel Contact Center Solutions portfolio to provide unified administration setup and use as well as access to all that portfolio's database, templates and reporting and data mining tools
- integrates with Mitel OPS Manager for multi-node data collection

The optional Subscriber Services application enables you to create customized telephone rate plans to cost subscriber calls.

The optional Traffic Analysis application analyzes trunk traffic to maximize service levels and decrease costs.

Mitel Speech-Enabled Applications (formerly 6500)

The Mitel Speech Server provides a unique architecture that brings powerful speech recognition capabilities to a wide range of telephony solutions, applications, and standalone products. Mitel Speech Server technology supports conversational speech recognition, recognizing entire sentences and not simply single words.

The Speech Server is a speaker independent, flexible vocabulary technology. This means that users don't have to train the system to understand their voice nor must they remember a fixed set of commands. The core Speech Server platform supports speaker authentication for unsuppressed security, barge in capability to allow power users to quickly navigate through applications with Calling Line ID for superior integration with PBX and voice over IP platforms

The suite of applications that the Speech Server offers includes:

- Auto Attendant
- Unified Messaging Attendant

The Speech Server Attendant allows users to place calls to people quickly and efficiently by speaking their names. In addition to placing calls by name, users can say a department name or telephone number or query the system for the phone numbers of people or departments. An online tutorial introduces users to the system features, and voiced-based help is available at any time.

The Speech Server Attendant is a Windows 2000-based system that works with Mitel SX-200 PBXs, SX-2000 PBXs and 3300 ICP systems, as well as Nortel Networks, Avaya, Siemens, NEC and analog PBX platforms.

With this application, users can

- Place a call to any number in the company directory by stating a name, extension, or department
- Navigate through multi-level menus using voice commands
- Call into the system from their home phone or cell phone and place calls to external numbers that are programmed in the company directory, provided the users have been assigned the required system privileges
- Program their own list of frequently called numbers and then place calls to those numbers through using speech commands (registered users only)

Additional functionality includes

- forwarding an incoming call to another number (Mobility option)
- integration with Outlook Contacts (Personal Directory)
- requires Active Directory 2000/2003 and Active Directory software option

Speech Server Unified Messaging

Unified Messaging provides unified messaging features in addition to the Speech Server Attendant functionality. Unified messaging stores e-mails, voice mails, and faxes in one location (Exchange 2000/2003 Server) and allows users to access and manage these messages anywhere, from a phone or desktop. If the required software options are enabled, users can also manage their appointments, meetings, and tasks, using speech commands.

With this application, users can

- Tell the system to organize and play messages based on caller name, date, type of message, and priority, so users don’t have to scroll through messages sequentially to find a particular message.
- Dial calls by simply saying the contact’s name (through access to Outlook's contact list).
- Check for urgent messages using the phone.

Extra functionality provides you with the ability to

- Manage appointments, meeting requests, or tasks using the telephone and speech commands (Calendar and Task Management option)
- Forward a fax or email to another fax number using the telephone (Fax Integration Option)

Speech Server Options

The following applications are available as additional purchasable options on Auto Attendant or Unified Communications:

- Mobility
- Calendar and Task Management
- Fax Integration

For complete details, refer to the Mitel Speech Server documentation.

Mobility

The Mobility option provides users with the ability to redirect the Auto Attendant calls made to their default number to one of their other programmed numbers, or to a temporary number. While Mobility is enabled, all calls made to the user's default number (by stating only the user’s name) are redirected to the "Reach me at" number. However, if a caller specifically requests the user’s cell phone, pager, fax, or home phone number (for example, by stating “Bill Smith on his cell”), then the call is directed to the requested number and is not redirected.

Calendar & Task Management

Calendar & Task Management provides Unified Messaging users with access to their Calendar and Task lists using spoken commands. Users can also review their message lists, and create, modify, or delete appointments, meetings or tasks.

Fax Integration

Fax Integration allows Unified Messaging users to integrate with a third party fax server. This option allows users to view their faxes in the Outlook In-box, program notification, read a fax header using text-to-speech and forward their faxes to another fax machine. The faxes are stored on the fax server and the Exchange 2000/2003 Server (a no charge option).

Note: Mitel does not provide the third-party fax.The third-party fax that is supported is Right Fax.

Unified Messaging - Standard

The Unified Messaging - Standard feature package enables the SX-200 ICP to manage e-mail messages using SMTP (Simple Mail Transfer Protocol) and/or IMAP (Internet Message Access Protocol). SMTP is a standard component of the feature package; IMAP requires the purchase of a Mitel 6000 MAS with the optional Unified Messaging Blade. In order to use this package, you must enable embedded voice mail for forwarding of voice mail to e-mail. The SX-200 ICP offers the following SMTP client features:

- Voice mail to e-mail: allows users to forward voice messages, including Record a Call messages, to e-mail. Users can manually forward individual messages or they can configure the system to automatically forward all messages. Users can also specify whether messages are saved or deleted once they are forwarded.

- Notification of E911 calls: when a user dials E911, a distribution list (maximum three users) will be sent in an e-mail message with the subject "E911." The body of the e-mail includes the caller's name, extension number and location (if programmed). Also, for accountability and potential liability purposes, a log with the same information shall be generated when the e-mail is sent.
- Notification of Alarms: if the SX-200 ICP system detects an Alarm, the system will create an alarm log that can be sent in an e-mail message to three different addresses with the subject "Alarm Notification." The body of the e-mail can include - minor, major, or critical alarms.
- On-demand Maintenance Logs: used to send higher level maintenance logs directly to Mitel Technical support which saves site troubleshooting time.

For more information on SMTP and IMAP support, refer to *SX-200 ICP Technical Documentation in Folio (NFO).*

MiTAI

Mitel Telephony Applications Interface (MiTAI) is an Applications Interface (API) that allows third-party-developed CTI applications to interface with Mitel's call control. A developer's toolkit plus run-time software is also available, which enables developers to create computer telephony applications. For additional information refer to the *MiSN Third-party Developers Program* page at Mitel OnLine.

Mitel NuPoint Messenger IP

NuPoint Messenger IP® is a powerful, server-based voice processing system that provides call processing along with voice messaging and paging support. Users can access their voice mails remotely and can be notified by telephone or pager when a voice message is left for them.

NuPoint Messenger IP offers inbound caller, attendant, hospitality and mailbox user messaging features as well as digital networking. All of these features and functions are accessible from a touch-tone telephone. NuPoint Messenger IP also offers complete desktop control of voice messages from a default email client or web browser. In addition, NuPoint Messenger IP provides applications and interfaces that administrators can use to administer the NuPoint server onsite or remotely as well as to create their own applications to suit their company's specific needs.

Some examples of NPM applications include:

- Paging a mailbox owner when a new voice mail message arrives
- Allowing callers to not only leave a voice mail message, but input their call back number which is then displayed on the mailbox owners pager.
- Scheduling automatic wake-up calls to any telephone at any date and time
- Recording a voice message and having it automatically distributed to thousands of people
- Delivering new, unplayed voice messages to an on- or off-system telephone number of choice

- Routing callers to predetermined destinations based on time of day, day of week, or day of year
- Property Management Integration and custom Hotel prompts

NuPoint Messenger IP supports integration to multiple PBXs through:

- Enhanced-SMDI (ESMDI)
- MWI information transfer between SX-2000 and 3300 ICP PBXs
- Dual MWI support for two separate PBX integrations

NuPoint Messenger IP now supports SMS notification to cellular phones. SMS notification text-messages users when they receive new voice messages.

NuPoint Messenger IP supports telephone softkeys that allow users to control voicemail functions through context-sensitive keys on the telephone. Softkey support is only usable by NuPoint Messenger IP systems that are integrated to the 3300 ICP using an IP integration.

Note that if the 200 ICP is networked to other PBXs (3300 ICP or SX2000), softkeys are supported on those PBXs only if MSDN networking is in use; T1/D4 is not supported.

For more information about NuPoint Messenger IP, refer to *NuPoint Messenger IP General Information Guide* available at www.mitel.com.

Mitel Your Assistant

Mitel provides full video and collaboration features via Mitel Your Assistant®. Your Assistant enables customers to effectively manage telephony and data communication with features that enhance productivity such as PC-based call handling, visual conference call management, secure chat, call annotation timing and recording.

Your Assistant is available in multiple versions: Your Assistant, Your Assistant Softphone, and Your Assistant Video/Data Collaboration Module.

Your Assistant

Your Assistant provides the following:

- **Simplified Call Management:** The Your Assistant desktop control panel offers intuitive visual point and click access to the call management features of the 3300 ICP. Ad hoc conference calls can be managed by dragging and dropping the name of a participant into the conference. Your Assistant remembers the most frequently dialed numbers and makes them easily accessible from a centralized drop-down menu.
- **Data and Telephony Presence and Availability:** Your Assistant maximizes successful communication by indicating if people are on the phone, away from their desk, available for secure instant chat or wanting data collaboration.
- **Corporate Secure Incorporated Instant Messaging (IM):** IM and file sharing features offer security as well as cohesive teamwork. Users can initiate single or multi-party chats

at the click of a mouse and share documents by dragging and dropping files into chat sessions.

- **Versatile Call Forward Options:** Allows users to set up multiple call forward profiles. Your Assistant also supports real-time call forwarding to other extensions, external phone numbers and voice mail via a simple interface.
- **Knowledge Management:** Allows users to associate files in various formats (Microsoft Word, Excel, PowerPoint and PDF) and Microsoft Outlook emails to a contact in their Corporate Contacts list and their PIM (Microsoft Outlook is supported). When a contact calls, their associated items are made available to the user for quick access.
- **Caller Line ID-based Routing:** Allows user to set up automatic call handling policies based on rules applied to specific caller line IDs. For example, users can forward selected calls to voice mail at specific times of the day. This allows users to take important calls while routing all other calls to voice mail.
- **Directory Integration:** Supports Corporate Directory for the 3300 ICP and LDAP database interface. If the user selects the Corporate Directory option, their Corporate Contacts list is populated with data from the 3300 ICP telephone directory. If the user selects the LDAP database interface option, it enables integration with additional PIMs and databases that support LDAP. The LDAP interface utility within Your Assistant maps the data fields in the external database to fields within Your Assistant.
- **Web Window:** A smaller browser window is provided as a shutter within the Your Assistant main window. This browser window is used to display timely notification of relevant information. It can be used to broadcast important messages to users within an enterprise.
- **Federated Servers:** Your Assistant servers in multiple locations can share IM and presence information between servers.These federated servers allow Your Assistant users in one office to view the presence and availability of Your Assistant users in another office in the same network.
- **Centralized Call Logging:** The Your Assistant server can log incoming calls for Your Assistant clients while the client software is not running. When Your Assistant is started, the Your Assistant server updates the client with all the cached call log information since the last client session and displays it in both the Call History and Call Log window.
- **Outlook Synchronization:** Your Assistant (release 3.1 and later) provides synchronization of contact data between Your Assistant and Outlook for contacts imported into the Your Assistant Personal Contacts list. Synchronization flows from Outlook to Your Assistant only which means that only the changes made within Outlook are reflected in Your Assistant.

Your Assistant Softphone

Your Assistant Softphone includes all of the features listed above as well as these additional features:

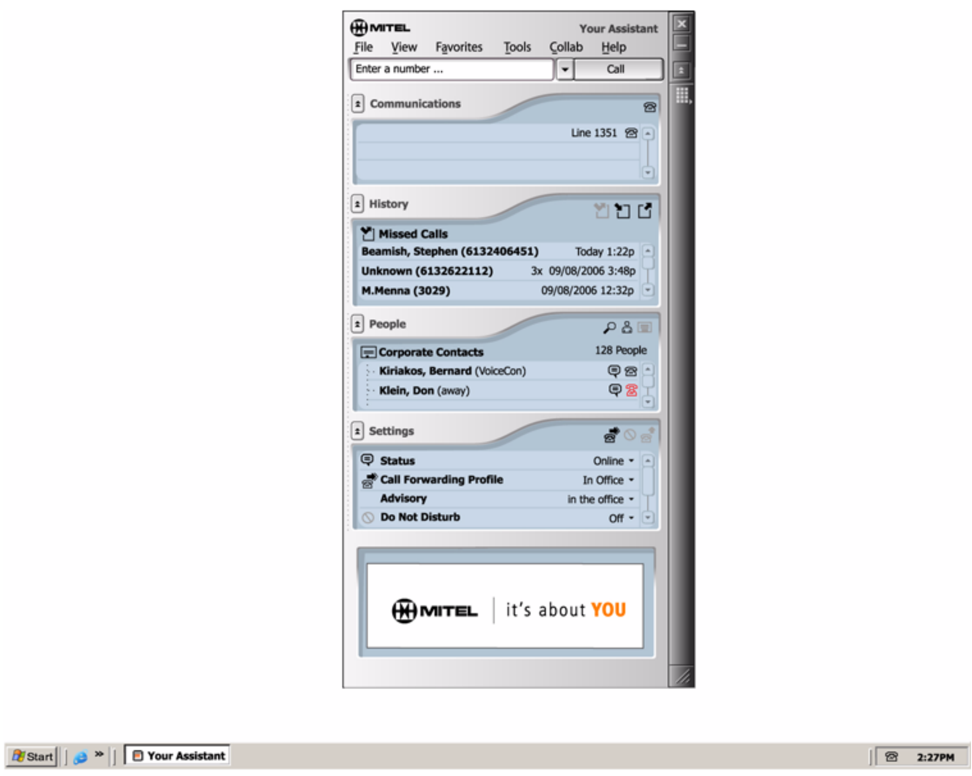
- **Embedded PC Softphone:** provides a embedded IP-based software telephone from a remote PC or laptop. This allows the remote user connected to the enterprise via a secure network connection to appear as though they are at their desk.
- **Record calls:** allows user to record calls and save them to their PC.

Your Assistant Video and Data Collaboration Module

Your Assistant Video and Data Collaboration Module allows users to easily escalate a voice call into a video and/or data conference at any time. The module include all of the features listed above (with or without softphone). Collaboration sessions can be scheduled with a meet-me URL to start a conference or created during a call for on-demand collaboration. This multimedia collaboration module offers internal and external users application sharing and co-browsing, remote desktop control and multi-party desktop video conferencing. Any user in an organization that has a Your Assistant license can create on-the-fly collaboration and conferencing sessions with other colleagues within the organization or with the public. Your Assistant eliminates the need to pay for hosted services and calling a web hosting provider to book time. The Your Assistant Video and Data Collaboration Module offers the following key features:

- Sharing of PowerPoint presentations, documents, applications, desktop regions and entire desktops
- Annotation and white boarding capability
- Video conferencing (using USB web camera). There is a 10 party maximum in collaboration session and a 20 party maximum in video only session.
- Setting up web conference from within the Your Assistant GUI while on a call (Voice First)
- Pre-scheduling collaboration sessions within Your Assistant

The following is an example of the Your Assistant interface.



Features

The SX-200 ICP offers a broad range of features. For feature descriptions, refer to:

- “SX-200 ICP Features Supported” on page 91
- “SX-200 Bay Services Supported on the MX Controller” on page 103
- “SX-200 ICP Feature Descriptions” on page 104

SX-200 ICP Features Supported

Table 4: SX-200 ICP Features Supported

Feature	CX/CXi, MX controller and AX Controllers
Abbreviated Dial	x
Access Codes-Global Find	x
Account Codes	x
Account Codes - Verified	x
Account Codes - Verified (Special DISA)	x
Add Held	x
Analog Networking	x
Attendant Abbreviated Dial Number Entry	x
Attendant Access (Dial 0)	x
Attendant Advisory Message Setup	x
Attendant Alarm Readout	x
Attendant Automatic Overflow	x
Attendant Bell Off	x
Attendant Busy Override	x
Attendant Callback-Busy No Answer	x
Attendant Call Forward Setup and Cancel	x
Attendant Call Selection	x
Attendant Call Splitting and Swapping	x
Attendant Calls Forwarded On No Answer	x
Attendant Conference	x
Attendant Console Display Language	x
Attendant Console Handset and Headset Receiver Volume Control	x
Attendant Console Last Call Retrieve	x
	Page 1 of 12

Table 4: SX-200 ICP Features Supported (continued)

Feature	CX/CXi, MX controller and AX Controllers
Attendant Console LCD Display	x
Attendant Console LDN Keys	x
Attendant Console Lockout	x
Attendant Console Macro Keys	x
Attendant Console Set Paging- Directed, Group, or All Set	x
Attendant Date and Time Setup	x
Attendant Default Call Positions	x
Attendant Destination (DEST) Key	x
Attendant Directed Call Pickup	x
Attendant Direct Trunk Select	x
Attendant DISA Code Setup	x
Attendant Do Not Disturb Setup, Cancel or Override	x
Attendant Emergency Call (911) Detection	x
Attendant Extension Busy-Out	x
Attendant Flash Over Trunk	x
Attendant Function Access	x
Attendant Hold Positions	x
Attendant Implicit New Call	x
Attendant Individual Directory Number	x
Attendant Interposition Calling and Transfer	x
Attendant Lockout Alarm	x
Attendant Message Waiting Setup and Cancel	x
Attendant Multi-New Call Tone	x
Attendant New Call Ring	x
Attendant Night/Day Switching	x
Attendant Paging Access	x
Attendant Paged Hold Access	x
Attendant Serial Call	x
Attendant Source Key	x
Attendant Timed Recall	x
Attendant Tone Signaling	x
Attendant Training Jacks	x
Attendant Transfer To Campon	x
Attendant Transparent Multi-Console Operation	x
	Page 2 of 12

Table 4: SX-200 ICP Features Supported (continued)

Feature	CX/CXi, MX controller and AX Controllers
Attendant Trunk Busy-Out	x
Attendant Trunk Group Status Display	x
Auto-Answer	x
Auto-Hold	x
Automated Attendant	x
Automated Attendant - Auto-Attendant Group	x
Automated Attendant - Default Destination	x
Automated Attendant - Front End Recording	x
Automated Attendant - Illegal Number Handling	x
Automated Attendant - Prefix Digits	x
Automated Attendant - RAD Operation	x
Automated Attendant - Resource Allocation	x
Automated Attendant - Vacant Number Routing	x
Automatic Call Distribution (ACD)	x
ACD - Path	x
ACD - Positions	x
ACD - Displays	x
ACD - Longest Idle Agent	x
ACD - Mobility	x
ACD - Predictive Overflow	x
ACD - Printed Reports	x
ACD- Real Time Event	x
ACD - Recorded Announcements	x
ACD - Sets	x
ACD Agent Dss/BLF	x
Automatic Number Identification (ANI) on Outgoing Trunks	x
ANI/Dialed Number Identification Service (DNIS) on Incoming Trunks	x
Automatic Route Selection (ARS)	x
Background Music	x
Broker's Call (Station Swap)	x
Broker's Call With Transfer (Transfer With Privacy)	x
Busy Lamp Field	x
Call Forwarding	x
	Page 3 of 12

Table 4: SX-200 ICP Features Supported (continued)

Feature	CX/CXi, MX controller and AX Controllers
Call Forwarding - Busy	x
Call Forwarding - Busy/No Answer	x
Call Forwarding - Display Prime as Forwarded	x
Call Forwarding - No Answer	x
Call Forwarding - External	x
Call Forwarding - Always	x
Call Forward - Follow Me	x
Call Forwarding - Forced Call Forward	x
Call Forwarding - Forward Calls	x
Call Forwarding - I'm Here	x
Call Forwarding - Internal / External Split	x
Call Forwarding - Toggle Keys	x
Call Logging	x
Call Park from Single-line Sets	x
Call Park from Multi-line Sets	x
Call Park System Orbit	x
Call Park and Page	x
Call Park - Personal	x
Call Park System Orbit	x
Call Park - User Selectable Park Orbit	x
Call Rerouting	x
Callback	x
Callback - Busy	x
Callback - No Answer	x
Calling Party Number Substitution	x
Campon	x
Campon Priority Over Call Forward Busy	x
Campon Warning Tone	x
Centralized Attendant	x
Centralized Voice mail	x
CENTREX Compatibility (Double Flash Over Trunk)	x
CENTREX Compatibility (Single Flash Over Trunk)	x
CLASS (station side) for Analog Telephones	x
CLASS for Digital Sets	x
	Page 4 of 12

Table 4: SX-200 ICP Features Supported (continued)

Feature	CX/CXi, MX controller and AX Controllers
Class of Restriction (COR)	x
Class of Service (COS)	x
Clear All Features	x
CO Line Group Key	x
CO Line Key	x
CO Lines - Retain Conference Parties After Trunk Hangs Up	x
CO Line - Select Direct	x
CO Line Type - Direct Access - Bypass Key System Toll Control	x
Conference	x
Conflict Dialing	x
Consoleless Operation	x
Contact Monitor	x
Customer Data Entry	x
Customer Data Entry - Default Data	x
Customer Data Entry - Range Programming	x
Customer Data Print	x
Data: Security	x
Data Transceiver (DTRX)	x
Data: DTRX Call By Name	x
Data: DTRX Call Originate/Disconnect	x
Data: DTRX Help	x
Data: DTRX Hotline	x
Data: DTRX Messages	x
Daylight Savings Time Adjustment	x
DCO - Supervisors	x
Default Database	x
Device Interconnection Control	x
Dial Tone Disable	x
Dial Tone - Discriminating	x
Dialed Intercom	x
Dictation Trunks	x
DID/Dial-In /Tie Intercepts	x
Digit Translation	x
	Page 5 of 12

Table 4: SX-200 ICP Features Supported (continued)

Feature	CX/CXi, MX controller and AX Controllers
Direct-In Lines (DIL)	x
Direct Station Page/Busy Lamp Field	x
Direct Station Select (DSS) Key	x
Direct Station Select / Busy Lamp Field (DSS/BLF) Call Pickup	x
DSS/BLF Interface Unit	x
Direct to ARS	x
Direct to ARS - Voicemail support	x
Direct Trunk Select	x
Disable Key-line Conference Beep	x
Disconnect Alarm	x
Display Caller ID on Non-Prime Lines	x
Display Keys	x
Distinctive Ring Tones	x
Do Not Disturb	x
Emergency Call Handling	x
Emergency Calls (911) - Detection and Reporting to Attendant Consoles	x
Emergency Calls (911) Reporting and Detection to Display Sets	x
Emergency Calls (911) Detection to ONS (CLASS) Sets	x
Emergency Calls (911) - Reporting to PSAP	x
Expensive Route Warning	x
FAX Tone Detection	x
Feature Keys	x
Flash - Calibrated	x
Flash Control	x
Flash Disable	x
Flash For Dial 0 (Attendant)	x
Flash For Waiting Call	x
Flash Timing	x
Forward Campon	x
Group Listening	x
Guest Room	x
Handset Mute (Microphone Mute)	x
	Page 6 of 12

Table 4: SX-200 ICP Features Supported (continued)

Feature	CX/CXi, MX controller and AX Controllers
Handset Receiver Volume Control	x
Handsfree Announce	x
Handsfree Answerback to a Directed Page	x
Handsfree Operation	x
Headset Mode Feature Key	x
Headset Mode - Automatic	x
Headset Operation	x
Headset Operation (Amplified Headset)	x
Headset Operation (Cordless)	x
Headset With In-line Switch Operation	x
Hold	x
Hold Reminder	x
Holiday Messages	x
Hot LIne	x
Hot Swap	x
Hotel / Motel (Lodging)	x
Hotel / Motel - Attendant Console Guest Room Softkey	x
Hotel / Motel -Attendant Message Register	x
Hotel / Motel - Attendant Message Waiting Setup and Cancel	x
Hotel / Motel -Audits	x
Hotel / Motel -Audit Screen	x
Hotel / Motel - Wakeups	x
Hotel/Motel - Personal and Multiple Wakeups	x
Hotel / Motel -Call Blocking	x
Hotel/Motel - Sub Attendant Call Blocking	x
Hotel / Motel -Call Restriction	x
Hotel / Motel -Check Out	x
Hotel/Motel - CLASS (station side) for Analog Telephones	x
Hotel / Motel -Do Not Disturb (DND	x
Hotel / Motel -Front Desk Features	x
Hotel / Motel -Guest Names	x
Hotel / Motel -Guest Room Message Retrieval	x
Hotel / Motel -Guest Room Superset Key Programming	x
Hotel / Motel -Guest Room Update Screen	x
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Table 4: SX-200 ICP Features Supported (continued)

Feature	CX/CXi, MX controller and AX Controllers
Hotel / Motel -Guest Search Screen	x
Hotel / Motel -House Statistics Screen	x
Hotel / Motel -Maid in Room Status Display - Superset Display Telephones	x
Hotel / Motel -Message Lamp Test	x
Hotel / Motel -Message Register	x
Hotel / Motel - Multi-user	x
Hotel / Motel - Passwords	x
Hotel / Motel - Property Management System (PMS)	x
Hotel / Motel - Room Condition	x
Hotel / Motel - Room Occupancy	x
Hotel / Motel -Room Search Screen	x
Hotel / Motel - Room Status Display	x
Hotel / Motel - Room Types and Room Codes	x
Hotel / Motel - Single Line Reports	x
Hotel / Motel - Suite Services	x
Hunt Groups	x
I Hold You Hold	x
Illegal Access Intercept	x
Inhibit Trunk Ring-Me-Back During Dialing	x
Intercept to Recorded Announcement	x
Intercom Calls	x
Internal Number Block	x
Inward Restriction (DID)	x
Language Change	x
Last Number Redial	x
Last Party Receives Dial Tone	x
Line Lockout	x
Line Preference	x
Line Privacy	x
Line Selection	x
Line Types and Appearances	x
Lockout Alarm	x
Logical Lines	x
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Table 4: SX-200 ICP Features Supported (continued)

Feature	CX/CXi, MX controller and AX Controllers
Mailbox Lockout	X
Maintenance	x
Manual Line (Dial 0 Hotline)	x
Messaging - Advisory	x
Messaging - Call Me Back	x
MITEL Application Interface (MAI)	x
Moving Stations and Superset Telephones	x
Multi-Attendant Positions	x
Music - on - Hold (MOH)	x
Names	x
Never a Consultee	x
Never a Forwardee	x
New Call Ring	x
NI3 Calling Name Delivery	x
Night Bells	x
Night/Day Switching	x
Night Services	x
Night Services Flexibility	x
Node Identification	x
Non-Busy Extension	x
Numbering Plan Flexibility (Conflict Dialing)	x
Off-Hook Alarm to Display Sets	x
Off-Hook Voice Announce	x
Off-Premises Extension	x
ONS Positive Disconnect	x
Originate Only Extensions	x
Overlap Outpulsing	x
Override (Intrude)	x
Override Security	x
Paged Party Page Tone	x
Paged Party Ring Tone	x
Paging - PA	x
Paging - Telephones	x
Paging - PA and Telephones	x
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Table 4: SX-200 ICP Features Supported (continued)

Feature	CX/CXi, MX controller and AX Controllers
Paging - All Set Page	x
Paging - Group Page	x
Parallel Connection of Industry-standard Telephones	x
Personal Speed Call	x
Phonebook Softkey	x
Pickup - Local and Directed	x
Pickup Groups - Display Ringing Extension	x
PRI Support	x
Printer / Terminal Support	x
Priority Dial 0	x
Privacy Enable / Privacy Release	x
Programmable Key Module (PKM)	x
Q.SIG	x
RAD Support	x
Recall	x
Receive Only Extensions	x
Record a Call (Incoming and Outgoing)	x
Remote LAN Access	x
Reminder	x
Reminders - Multiple	x
Resale Package	x
Ring Groups	x
Ringer Control	x
Ringing - Discriminating	x
Ringing - Plan	x
Ringing Time-Out (Final Ringback)	x
Satellite PBX	x
Secretarial Line	x
SMTP Authentication	x
SMTP Client Forward Voice Mail to Email	x
SMTP Client E911 Notification via Email	x
SMTP Client Maintenance Alarms via Email	x
Speaker Volume Control	x
Speed Call Key	x
	Page 10 of 12

Table 4: SX-200 ICP Features Supported (continued)

Feature	CX/CXi, MX controller and AX Controllers
Split	x
Station Message Detail Recording (SMDR)	x
Subattendant - Basic Function	x
Subattendant - Enhanced Functions	x
Subattendant - Abbreviated Dial Programming	x
Subattendant - Advisory Message Setup	x
Subattendant - Automatic Call Wakeup	x
Subattendant - Call Blocking	x
Subattendant - Call Forward Setup and Cancel	x
Subattendant - Calls Waiting Indication	x
Subattendant - Date and Time Setup	x
Subattendant - Hold Positions	x
Subattendant - LDN Keys	x
Subattendant - Paged Hold Access	x
Subattendant - Recall	x
Subattendant - Station DND Setup	x
SUPERSET 4000 Series Sets	x
Swap (Trade Calls)	x
Swap Campon	x
System Fail Transfer (SFT)	x
System Identifier	x
System ID Module	x
Tandem Operation	x
Tenanting	x
Toll Control	x
Tone Demonstration	x
Tone Plans	x
Traffic Measurement	x
Traffic Shaping/IP Trunking	x
Transfer	x
Transfer Dial Tone	x
Transfer Security (Recall)	x
Trunk Answer From Any Station (TAFAS)	x
Trunk Dial Tone Detection	x
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Table 4: SX-200 ICP Features Supported (continued)

Feature	CX/CXi, MX controller and AX Controllers
Trunk Groups	x
Trunk Operation - Direct Inward Dial (DID)	x
Trunk Operation - Direct Inward System Access (DISA)	x
Trunk Operation - Non-Dial-in CO	x
Trunk Operation - Tie	x
Trunk Recall	x
Trunk Support - CO (LS/GS)	x
Trunk Support - Direct Inward Dial (DID)	x
Trunk Support - E&M	x
Trunk Support - IP	x
Trunk Support - T1	x
Uniform Call Distribution	x
Vacant Number Intercept	x
Voice mail Support	x
Voice mail Support - Centralized	x
Voice mail Support - softkeys	x
Voice mail Support - Single Button Transfer	x
Whisper Announce	x
Wireless - 802.11b Support	x
	Page 12 of 12

SX-200 Bay Services Supported on the MX Controller

Table 5: Bay Services Supported on the MX Controller only

Feature	MX Controller
BRI Card Support	x
Calculator	x
Data: Abbreviated Dial for ADL Calls	x
Data: Account Codes	x
Data: Associated Data Line (ADL)	x
Data: ADL Hotline	x
Data: ADL Speed Call Originate	x
Data: Associated Modem Line	x
Data: Auto-Answer	x
Data: Automatic Data Route Selection (ADRS)	x
Data: Hunt Groups	x
Data: Modem Pooling	x
Data: Modem Pooling Queuing	x
Data: Peripherals	x
Data Station Message Detail Recording (Data SMDR)	x
Data Station Queuing	x
DTMF-To-Rotary Dial Conversion	x
Meter Pulse Collection	x
Speak@Ease™ Support	x

SX-200 ICP Feature Descriptions

Table 6: Features supported by SX-200 ICP

Feature	Description
Abbreviated Dial	Allows trunks and extensions to be accessed by dialing a two-to eight-digit number that the system translates into the actual, longer number. The actual number can contain up to 26 digits. Can 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 ACD interflow and automated attendant, and as call-forwarding points.
Access Codes - Global Find	Allows authorized users to display 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.
Account Codes	Used to charge the cost of outgoing trunk calls to departmental cost centers or project accounts. The Account Code can be optional or required, and appears on all Station Message Detail Recording (SMDR) records.
Account Codes - Do Not Display	Determines whether or not account code digits are displayed on phones. When the option is disabled, an asterisk will appear for every account code digit entered.
Account Codes - Verified	Controls access to trunks and external (DISA) access to the system by checking the dialed account code against a list of preprogrammed codes.
Account Codes - Verified (Special DISA)	Can be used to replace the DISA Access 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.
Add Held	Allows a user engaged in an active call on a Mitel display telephone to add a call that is on hold on another line, to the current line.
Analog Networking	Allows the system to send and receive caller information over a private network. Analog Networking uses the ARS Modified Digit feature to insert feature access codes and other codes (called information elements) into the outgoing digit string.
Attendant Abbreviated Dial Number Entry	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 authorized attendants.
Attendant Access (Dial 0)	A feature access code (usually 0) is provided for reaching the attendant. The destination can change based on night/day service.
Attendant Advisory Message Setup	There are eight default messages and seven programmable messages for use on Mitel display telephones. The Attendant can read a set's currently displayed message, or read through the available messages and choose one for display on the set.
Attendant Alarm Readout	The Attendant Console can display the alarm logs active in the system. Using the softkeys, the Attendant can read the alarm messages one by one. The message indicates the fault and its location.
Attendant Automatic Overflow	Attendant Automatic Overflow provides a recorded announcement to incoming calls that are not answered by the attendant within a predefined time. This feature operates primarily during peak periods of incoming traffic.
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Table 6: Features supported by SX-200 ICP (continued)

Feature	Description
Automatic Call Distribution (ACD)	Automatic Call Distribution (ACD) is a purchasable option that distributes calls evenly among trained operators (agents). The ACD TELEMARKETER is an advanced Automatic Call Distribution (ACD) system that is fully integrated with the SX-200 ICP, and designed with the power and performance needed to ensure satisfaction in the most demanding telemarketing environments.
ACD DSS/BLF	Allows ACD agents to display other Busy agents on a Busy Lamp field. Direct Station Selection (DSS) allows agents to transfer calls by pressing the DSS key.
ACD - Path	An innovative call routing design that guides incoming calls through the system. The path defines all information required for each type of call including how the system is to handle callers placed in a queue to wait for an agent. With 99 ACD paths in the system, customized routing is available to every conceivable type of incoming call. Priority designations of 1 to 99 may be assigned to each path, allowing calls arriving on high priority paths to move directly to the front of the call queues.
ACD - Positions	The ACD TELEMARKETER feature package structures the personnel handling ACD calls into a hierarchy of ACD positions. The ACD package supports three types of positions: senior supervisors, supervisors, and agents. ACD calls entering the system usually terminate on an agent position. Agents handling similar types of calls are arranged in agent groups. Supervisors and Senior Supervisors monitor agent and system performance, but do not handle ACD calls.
ACD - Displays	The ACD TELEMARKETER feature includes real-time displays via sockets and ASCII terminals.
ACD - Longest Idle Agent	If multiple agents are free when an ACD call is presented to a group, the system sends the call to the longest idle agent.
ACD - Mobility	ACD Agents and Supervisors are completely mobile. All ACD positions are linked to software rather than hardware. The system recognizes a login from any telephone programmed as an ACD position within the system and immediately transforms the set to the user’s preprogrammed specifications.
ACD - Predictive Overflow	Used by the system to keep call queueing time to a minimum. The system performs a load calculation when each new call arrives at an agent group, or when the status of an agent changes. If the system predicts that a call will not be answered before the normal overflow time, it forces an immediate overflow.
ACD - Printed Reports	Provides summary reports of paths, groups, and agents. Printed Daily Reports include • ACD Agent Daily Activity Report listing hourly totals by agent ID • Agent Group Daily Activity Report with hourly totals handled by each agent group • Path Activity Report with detailed statistics for all ACD calls.
ACD - Real Time Event	Allows a PC to report real time events of ACD activities. The system transmits call status messages to the host computer reflecting the changes of state on the line or on the device.
ACD - Recorded Announcements	Recorded announcements are used to tell callers about the progress of their call while waiting in the queue for the first available agent.
ACD - Sets	Mitel IP Phones and Superset 4000 series that are equipped with displays can provide agent groups and individual agents with ACD information. Mitel 5212, 5224, 5304, 5312, 5324, 5330, 5340 IP Phones and Superset 4015 and 4025 telephones may be used in the senior supervisor, supervisor, or agent positions with the ACD TELEMARKETER feature package.
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Table 6: Features supported by SX-200 ICP (continued)

Feature	Description
Attendant Bell Off	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.
Attendant Busy Override	Allows an Attendant who encounters a busy connection to override the connection and enter the call.
Attendant Callback - Busy/No Answer	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.
Attendant Call Forward Setup and Cancel	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 the same time.
Attendant Call Selection	Allows you to choose which group of incoming calls to answer first. Each group is selected by pressing a softkey on the attendant console.
Attendant Call Splitting and Swapping	While setting up a call between two parties, the Attendant can speak to both parties at the same time, or to speak privately with either party. The Attendant does this using the CONF, SOURCE, and DEST softkeys
Attendant Calls Forwarded On No Answer	Calls directed to the console LDN that are not answered within a predetermined time-out period are rerouted to a NIGHT1 destination.
Attendant Conference	Allows the attendant to set up one or more conference connections between central office trunks and internal stations.
Attendant Console Display Language	Enables attendant the to choose the language of operation for the attendant console (English, French, Spanish)
Attendant Console Handset and Headset Receiver Volume Control	The attendant on a SUPERCONSOLE 1000 (Part Numbers 9189-000-300 and 9189-000-301) can use the volume keys to adjust the console ringer and the volume of the handset and the headset receiver.
Attendant Console Last Call Retrieve	Allows the attendant to retrieve a ringing call after accidently releasing a call to the wrong extension number.
Attendant Console LCD Display	The time of day, call information and total number of calls in the queue is displayed on the status line of the Attendant Console LCD display. When the console is idle, the date (month, day, year) is also displayed.
Attendant Console LDN Keys	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.
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Table 6: Features supported by SX-200 ICP (continued)

Feature	Description
Attendant Console Lockout	Allows the Attendant to enter an access code to restrict the capabilities of the Attendant Console. This can prevent system tampering via the console when the console is unattended. When the console is locked out, no outgoing trunk calls can be made and there is no Attendant function access. The Attendant Console can still be used to initiate internal calls, and to answer incoming trunk calls.
Attendant Console Macro Keys	The attendant on the SUPERCONSOLE 1000 (Part Numbers 9189-000-300 and 9189-000-301) can program macro keys using the blank key next to the Trunk Group key. The Trunk Group key and the Set Page Key can also be reprogrammed to be a macro key
Attendant Console Set Paging - Directed, Group, or All Set	The attendant can press the Set Page hardkey on the console to make a directed page or group page. The all set page is activated with a feature access code and then the softkey prompts or with pressing the Page hardkey.
Attendant Date and Time Setup	The time of day is continually displayed 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 operations and Mitel display telephones. The time may be displayed in 12- or 24-hour format. The console can change the date and/or time.
Attendant Default Call Positions	Three incoming call indicators identify calls to the console directory number. These three default positions are • F0 (NIGHT BELL): Calls ringing any night bell in the console's tenant group • F1 (RECALL): Recalls of calls handled by the console, or for multiple console operation, by any console in the system • F2 (INTERNAL): Calls directed to the console's internal directory number.
Attendant Destination (DEST) Key	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.
Attendant Directed Call Pickup	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.
Attendant Direct Trunk Select	The console may access (seize) a trunk directly to place a call or to test the trunk.
Attendant DISA Code Setup	Allows the Attendant to change the Direct Inward System Access (DISA) security code that a DISA caller must dial to access the system.
Attendant Do Not Disturb (DND) Setup, Cancel or Override	The Attendant may set up or cancel Do Not Disturb (DND) for an extension. When calling an extension with DND enabled, the Attendant may override the Do Not Disturb. See Do Not Disturb.
Attendant Emergency Call (911) Detection	See Emergency Call (911) - Detection and Reporting.
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Table 6: Features supported by SX-200 ICP (continued)

Feature	Description
Attendant Extension Busy-Out	Allows the Attendant to busy out any extension, and to remove the busy-out condition. A busied-out extension is removed from service and cannot originate or receive any calls. If the attendant dials the number of a busied-out extension, the console displays the extension number and “BUSY OUT” in the destination display, and the attendant receives reorder tone.
Attendant Flash Over Trunk	The attendant can flash on any type of trunk by pressing the FLASH softkey. A flash is sent out on the trunk, and dialing is restarted on the trunk.
Attendant Function Access	By pressing the console FUNCTION key and the ATT FUNCTION softkey, the Attendant can access Attendant features including, Alarm (read alarms), Application (To access CDE or maintenance), Busy Out, DAY/NIGHT1/NIGHT2 Switching, Francais (French language prompts and messages on console), Guest Room functions (if enabled), Message Waiting Setup and Cancel, Set System Date and Time, Trunk Status/Access/Busy Out.
Attendant Hold Positions	The Attendant can place an extension or trunk on hold in one of eight HOLD positions. If the Attendant is visually impaired and unable to see the HOLD key LED, the Attendant Hold Position Security feature can be enabled. This allows for an error beep to sound if the Attendant attempts to put a call on hold by pressing a HOLD key that already has a party on hold.
Attendant Implicit New Call	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. This feature is temporarily disabled by pressing the TONES ON softkey. See Attendant Tone Signaling.
Attendant Individual Directory Number	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) user dial to call 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 that is dedicated to a particular attendant position (useful when there is more than one Attendant position).
Attendant Interposition Calling and Transfer	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 manner as a call to an extension.
Attendant Lockout Alarm	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.
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Table 6: Features supported by SX-200 ICP (continued)

Feature	Description
Attendant Message Waiting Setup and Cancel	This 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 display telephone. • A continuously flashing lamp on the extension (if equipped). • A distinctive ringing pattern repeated every 20 minutes. The pattern is three 350 ms bursts of ringing.
Attendant Multi-New Call Tone	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. This feature is disabled if the attendant bell is turned off from the console. Also see Attendant Bell Off.
Attendant New Call Ring	If an Attendant is active on a 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 Console LCD Display.
Attendant Night/Day Switching	The Attendant can select NIGHT1, NIGHT2, or DAY service using softkeys. Also see Night Services.
Attendant Paging Access	The Attendant may access a paging zone or zones 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).
Attendant Paged Hold Access	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 access code (for feature access code 16) plus the number of the call hold slot position that must be dialed to pick up the call.
Attendant Serial Call	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 system without the need for transfers by the called extensions.
Attendant Source Key	Pressing the SOURCE softkey allows the Attendant to speak with 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 on Consultation Hold at the Console does not hear music.
Attendant Timed Recall	This feature automatically alerts the Attendant when a call extended through the Console or a call on hold at the Console has not been answered within a programmed time-out period.
Attendant Tone Signaling	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.
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Table 6: Features supported by SX-200 ICP (continued)

Feature	Description
Attendant Training Jacks	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.
Attendant Transfer To Campon	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 cannot camp on but can transfer calls into Campon. See Campon. For details of recall from Campon, see Recall.
Attendant Transparent Multi-Console Operation	Allows some features to apply to a group of consoles within a tenant. For example, when Transparent Multi-Console is used, a console may review or cancel a Message Waiting indication for any station. Without this feature, only the console that set the Message Waiting for a specific station, can review or cancel it. Recalls to the RECALL softkey for any console in the group can be answered by any console in the group.
Attendant Trunk Busy-Out	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.
Attendant Trunk Group Status Display	This feature allows the Attendant to display the status of trunk groups in the system. If this feature is activated while the Console is idle, the display is refreshed approximately every 5 seconds to allow a constant up-to-date monitoring of the trunk groups
Auto - Answer	When the Auto-Answer feature is active, incoming calls give a burst of ringing and 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 or IP telephone’s speaker and the set goes idle. Call origination is not affected.
Auto - Hold	A Superset or Mitel IP telephone user automatically puts a call on hold when a Line Select key on the set 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.
Auto Latch Microphone	Auto Latch Microphone allows the handsfree microphone to automatically turn on or off when receiving a page. If this feature is enabled, a paged phone’s microphone LED will flash.
Automated Attendant	Automated attendant features connect incoming calls from a DTMF telephone to a recording. The recording instructs 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 is 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.
Automated Attendant - Auto-Attendant Group	Introduces an additional hunt group type called an Auto-Attendant group. This group is similar to the recording groups used in the ACD TELEMARKETER application but will not accept caller’s input (DTMF). The Automated Attendant Feature is accessed by either rerouting or dialing into an Auto-Attendant group.
Automated Attendant - Default Destination	When a recording ends, callers who have not dialed at least one digit during the recording are routed to the default destination.
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Table 6: Features supported by SX-200 ICP (continued)

Feature	Description
Automated Attendant - Front End Recording	Front End Recordings present a message to the caller as soon as the call enters the system. For each recording group, dialing can be disabled during the recording. This provides a simple front-end recording without assigning a DTMF receiver. Digits dialed by the caller are ignored, and the prefix digits have no affect. Calls are routed to the default destination as usual.
Automated Attendant - Illegal Number Handling	If the dialed number is illegal, the system checks for illegal number routing. If the tenant group of the first member of the automated attendant group has illegal number routing programmed, the system redirects the caller to the routing point, which could be another group. If there is no illegal number routing programmed the caller is given reorder tone.
Automated Attendant - Prefix Digits	Each automated attendant group can be programmed in CDE with a string of prefix digits. The prefix can contain from 0 to 4 digits, and is inserted in front of the digits dialed by the caller. This allows the caller to dial a single digit and be routed to devices that have normal multi-digit extension numbers.
Automated Attendant - RAD Operation	RAD operation is similar to the RADs in the ACD TELEMARKETER application. The Automated Attendant feature uses the Auto-Attendant group as an enhanced recording group so the basic recording group features apply.
Automated Attendant - Resource Allocation	Each call entering the Automated Attendant feature uses two primary resources: a RAD and a DTMF Receiver. Usage differs between the two resources as follows. In the case of RADs, every time a RAD becomes free an unlimited amount of that resource becomes available because of the unlimited number of listen-only conferences that can be serviced by that one RAD. When a receiver becomes free, however, only one piece of that resource becomes available because only one caller can use the receiver at a time. Receiver availability therefore becomes the primary resource limitation for the Automated Attendant feature.
Automated Attendant - Vacant Number Routing	Handling of callers dialing a vacant number, such as an unassigned access code, is similar to the illegal number handling described above. In the case of a vacant number, vacant number routing is checked instead of illegal number routing.
Automatic Number Identification (ANI) / Dialed Number Identification Service (DNIS) on Incoming Trunks	Allows the SX-200 ICP to identify Automatic Number Identification (ANI) numbers and Dialed Number Identification Service (DNIS) numbers that are transmitted to the system on an incoming trunk. ANI provides the telephone number of the calling party, while DNIS provides the telephone number dialed by the calling party.
Automatic Number Identification (ANI) on Outgoing Trunks	Allows the system 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 system has successfully dialed an external number on that trunk.
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Table 6: Features supported by SX-200 ICP (continued)

Feature	Description
Automatic Route Selection (ARS)	Automatically selects one of a preprogrammed (programmed during CDE) list of trunk routes every time an outgoing call is made. The routes are selected based upon the digits dialed, in order of cost (i.e., least expensive route first), and in accordance with the caller's toll restriction. The use of digit analysis and digit modification within the ARS package allows the system to recognize and modify any digit string which is dialed by the user, alleviating the need for the user to dial special trunk access codes, or to dial a different digit string for each of the various routes to the same destination. The complete ARS package provides the following: • **Alternative Routing** - automatically selects an alternate trunk route when the first choice is busy. • **Least Cost Routing** - enables the customer to capitalize on the cost benefits offered by each type of trunk by allowing the installation company to define, via the Route Plans and Route Lists Tables, the order in which the trunk groups are to be selected. • **Toll Control** - allows the customer to restrict user access to specific trunk routes and/or specific directory numbers. • **Overlap Outpulsing** - seizes a trunk and commences outpulsing as soon as sufficient digits have been received to identify the route. • **Expensive Route Warning** - presents a tone to the user during call setup, and the message EXPENSIVE ROUTE appears on the LCD when the route selected by ARS is programmed as an expensive route. • **Callback Queueing** - allows the user who encounters busy tone after dialing an ARS digit string (i.e., all trunks busy) to dial a callback access code, or to select CALLBACK, and be placed in a queue for the first available trunk. • **Camp-on Queueing** - allows the user who encounters busy tone after dialing an ARS digit string (i.e., all trunks busy) to wait off-hook, or, to select CAMP ON and remain off-hook until a trunk becomes free. • **Return Dial Tone** - allows the system to simulate CO dial tone for customers who consider that its absence would confuse the users of their system.
Background Music	Allows the user of a Mitel multi-line telephone to have background music played through the set's speaker while the set is idle. The Music-on-Hold source provides the music. See Music-on-Hold.
Broker's Call (Station Swap)	Broker's Call is similar to the Transfer With Privacy feature. It allows the user to speak privately with two separate parties. When the user hangs up, however, the two parties are disconnected.
Broker's Call With Transfer (Transfer With Privacy)	The Transfer With Privacy feature is similar to the Broker's Call feature by allowing the user to speak privately with two separate parties. When the user hangs up, however, the two parties are connected. Transfer with privacy interprets a flash as a swap. A conference cannot be formed by an extension with this feature enabled.
Busy Lamp Field	A Busy Lamp Field (BLF) on the Programmable Key Module indicates to the user the status (Idle, Busy, DND) of a line appearance for a device such as, normal stations, Superset prime lines, logical lines, and trunks. Any Mitel IP Phone or Superset telephone with line keys may be programmed to use BLF indicators.
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Table 6: Features supported by SX-200 ICP (continued)

Feature	Description
Calculator	Superset 4150, Superset 430, and Superset 4DN telephones provide a basic, four-function calculator using the telephone keypad, display, and softkeys. The calculator has 2 modes of operation - General and Programmer. In the General mode the calculator allows for arithmetic operations and the telephone keypad is used as the numeric keypad. In the Programmer mode the calculator allows for integer arithmetic/logical operations in either decimal or hexadecimal.
Call Forwarding	Allows you to redirect incoming calls to an alternate number.
Call Forwarding - Busy	Forwards all calls when the extension is busy. While the extension is idle, calls can be made and received as usual.
Call Forwarding - Busy/ No Answer	Forwards all calls that are received when the extension is busy or are not answered within a selected time-out period. While the extension is idle, calls can be made and received as usual.
Call Forwarding - Display Prime as Forwarder	Displays either the forwarder's prime extension number or the line on the forwardee's set display. If the feature is enabled, the prime extension number of the set that forwarded the call is displayed. When the feature is disabled, the logical line appears as the forwarder for all types of forwarding.
Call Forwarding - No Answer	Forwards all calls that are not answered within a selected time-out period. Calls may be made and received normally.
Call Forwarding - External	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 personal speedcall.
Call Forwarding - Always	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 extension that can call the forwarding extension while Forwarding - Always is active. The extension can originate calls in the usual manner.
Call Forwarding - Forced Call Forward	Allows a Mitel telephone user to force a dialled call to forward immediately rather than waiting for the ringing timeout. One application of this feature is to leave a quick voice mail message for someone who you know is not at their desk.
Call Forwarding - Forward Call	Allows the user of a multi-line Superset or Mitel IP telephone which is ringing, to force the call to be forwarded to a pre-programmed forward destination. Users can forward both ringing calls and camped on calls. With a Superset 4150 or Superset 430 telephone, the user may 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.
Call Forwarding - I'm Here	This type of forwarding operates the same as Forwarding -Always, but it is activated from another extension. All calls are forwarded to the new location. The forwarded extension can originate calls in the usual manner.
Call Forwarding - Internal / External Split	Allows internal and external calls to be forwarded to different destinations. For example, internal callers can be forwarded to Voice mail, while external callers can be routed to an attendant.
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Table 6: Features supported by SX-200 ICP (continued)

Feature	Description
Call Forwarding - Toggle Keys	Feature keys may be programmed as toggle keys for Call Forwarding - Always, Call Forwarding - Busy, Call Forwarding - No Answer, or Call Forwarding - Busy/No Answer. The toggle keys allows you to direct your call forwarding temporarily to another destination with a touch of a button. You may toggle back and forth between one forward destination and another without having to reprogram the call forwarding. The SX-200 ICP also supports tenant-based call forwarding. This allows the installer to assign common call forwarding destinations for Call Forwarding - Busy and Call Forwarding - No Answer to tenant groups in CDE Form 19.
Call Logging	Provides the telephone user with a log showing a list of all the calls to their telephone set. The log includes all calls answered and not answered, as well as, calls forwarded to another destination. The user can press the Call softkey to return the call. Call Logging is available for multiple appearances of DTS and CO line keys.
Call Monitor	Call Monitoring allows a user to listen in on calls. During a call monitor, the system gives the monitoring set a one-way audio path preventing the monitored user and the caller from hearing the monitoring user.
Call Park from Single-line Sets	allows users of single-line 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.
Call Park from Multi-line Sets	Allows you to park a call from your prime line. After you park a call you can make or accept other calls from your prime line.
Call Park - Destination Phone	Allows the called party to park an answered call on another phone. If a parked call is not retrieved after a specified length of time, a reminder occurs, if programmed.
Call Park - Fixed Keys	<<need description>>>
Call Park System Orbit	Allows you to park up to 25 calls in a parking orbit (auto-selected by the system) from any line on your display set or console. Your display shows the park access code and the orbit number. You can then page and inform the paged party where they can retrieve the call. After you park the call, your line is free to make or accept other calls.
Call Park System - Specific Orbit	Allows the called party to park a call in a specific orbit by entering a two-digit (01-25) orbit number or by pressing a feature key assigned to a specific orbit number. You can retrieve a call from orbit from any phone or attendant console.
Call Rerouting	Different types of calls can be routed to different answering points in DAY, NIGHT1, and NIGHT2 service for each tenant. Rerouted calls are processed differently than normal calls. The system considers rerouted calls to be important calls that must get through.
Callback	Allows the system to notify a caller 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 destination (set, hunt group, or trunk group) to have the call completed when the destination becomes idle. The system continuously monitors the originating set or console and the destination. When the originating set is idle and the call can be completed, the system calls the originating set. When that set or console answers, it calls the destination.
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Table 6: Features supported by SX-200 ICP (continued)

Feature	Description
Callback - No Answer	Allows a user, after dialing an extension which does not answer, to have the call completed 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.
Campon	Allows you to notify a busy party that you are waiting. An attendant may also put a call through to a busy station to indicate that a call is waiting. Upon hearing the Call Waiting tone, the busy party can either respond or finish the current call.
Campon Priority Over Call Forward Busy	If an internal call to a set that is busy and has Call Forward - Busy (or Call Forward - Busy/No Answer) activated, the call is immediately forwarded. With this feature, the caller has the option of camping on to the busy set or allowing the forward to take place. This feature can be used for both calls on internal lines as well as trunk calls.
Campon Warning Tone	When a device sets Campon 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.
Centralized Attendant	Allows an attendant or subattendant on one system to answer calls that arrive at another interconnected system. The call arrives at the attendant via a dedicated Release Link Trunk (RLT), which can be T1 E&M, T1 E&M DISA, E&M, or E&M DISA. When the attendant releases a call to its destination, the RLT is released. This is a purchasable option.
CENTREX Compatibility (Double Flash Over Trunk)	Provides the ability to send a double switchhook flash out over a trunk. Flashing over a trunk allows for the use of CENTREX features by telephones within the system.
CENTREX Compatibility (Single Flash Over Trunk)	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 system. 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 make sense to the local central office may be programmed into a Speedcall key, or a System Abbreviated Dial Number. The trunk must be in a trunk group in order to flash over the trunk.
CLASS (Custom Local Area Signaling Services) for Analog Telephones	CLASS (station side) for Analog Telephones allows the SX-200 ICP to pass Calling Line ID digits and CLASS name information through to analog stations, such as display sets, that support Caller ID functionality.This feature supports a calling name and number display (if available) to a ringing set and to a set in talk state (a visual call waiting or campon). The CLASS message is also able to activate and de-activate the message lamp. To maintain privacy, the CLID information is cleared on checkout.
CLASS (Custom Local Area Signaling Services) for Digital Sets	Available on the LS/CLASS Trunk. The system receives Calling Line ID digits or CLASS name on incoming Loop Start (LS) CLASS trunks. The calling directory number or name is presented to Superset and Mitel IP display telephones, to SUPERCONSOLE 1000 consoles, to SMDR printers, and to the MITEL Application Interface (MAI) platform package.
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Table 6: Features supported by SX-200 ICP (continued)

Feature	Description
Class of Restriction (COR)	Fifty Class of Restriction (COR) groups are available in the system to give 50 different levels of outgoing call capabilities. Each extension, Superset telephone, dataset, console or dial-in trunk is assigned a COR that defines the outgoing call privileges for that device. All devices with the same COR have the same outgoing call privileges. The Class of Restriction allows the system to restrict which trunks cannot be accessed by a user.
Class of Service (COS)	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 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.
Clear All Features	An extension user may cancel all Call Forwarding, Do Not Disturb, and Callbacks Active at that extension.
CO Line Group Key	The CO Line Group key allows the 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 or LED indicator corresponding to the key has no function.
CO Line Key	The CO Line key originates and answers calls to or from parties outside the system. The key accesses a specific trunk directly. A CO line key may be shared by up to 64 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.
CO Line - Select Direct	Allows a direct access to 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.
CO Line Type - Direct Access - Bypass Key System Toll Control	Allows an extension seizing a CO trunk with a line key to bypass the system 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 or they may dial an external destination number.
Conference	Allows a set user to establish a conference of up to five parties (including the originating extension), without the assistance of the Attendant.
Conflict Dialing	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). Users could experience slower performance with conflicts.
Consoleless Operation	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 3DN, Superset 4DN, and Superset 4150 telephones may be used as Subattendant positions. These may switch the system to night service. See Night/Day Switching. Subattendant positions can also be given enhanced call handling and recall capabilities.
Contact Monitor	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 system handles the call as a call reroute.
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Table 6: Features supported by SX-200 ICP (continued)

Feature	Description
Customer Data Entry (CDE)	Customer Data Entry is a full screen application, using softkey prompts and simple graphics for entering and changing customer programming. CDE can also be performed from a remote location, using a terminal connected to the system via a direct connection, remotely through a modem, IP secure telnet, or through a web-based graphical user interface.
Customer Data Entry - Default Data	The system is preprogrammed with a complete default database for rapid deployment; if no alternates are programmed, the system defaults to the preprogrammed data.
Customer Data Entry - Range Programming	Allows range programming for blocks of extensions and trunks. 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.
Customer Data Print	Displays the current programming of the SX-200 ICP. Each or all of the CDE forms may be printed, one at a time, in a presentable format.
Daylight Savings Time Adjustment	Allows you to program the system to automatically set the system clock ahead for daylight savings time or back for standard time.
Device Interconnection Control	Provides a means of disallowing connection between devices of different types. The feature is primarily intended to control trunk interconnections but applies to other devices as well. This is intended to provide a method of meeting interconnection restrictions 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.
Dial Tone Disable	Assigning 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 when dialing is initiated.
Dial Tone - Discriminating	An extension that has 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, followed by continuous tone) when going off-hook to make a call. These features include Do Not Disturb, Call Forwarding - Always, Call Forwarding - I Am Here, and Message Waiting.
DID/Dial-in/Tie Intercepts	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 redirected 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, to see how this fits in with general recall operation.
Digit Translation	The SX-200 ICP may be programmed to provide one of four digit translation plans for rotary telephone sets. The translation plans specify the number of pulses to be outpulsed for each digit dialed.
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Table 6: Features supported by SX-200 ICP (continued)

Feature	Description
Direct-in Lines (DIL)	Allows non-dial-in trunks to ring specific answering points, rather than the Attendant Console. The answering point may vary with Night Service changes. An answering point may be a: • ACD path • extension number (industry-standard telephone, Superset telephone, Mitel IP telephone, logical line) • hunt group access code • night bell access code • system speed dial number to route to a Central Attendant at another system.
Direct Station Page/Busy Lamp Field	Allows you to direct a page with a DSS (Direct Station Select) softkey from a telephone to a selected telephone station.
Direct Station Select (DSS) Key	The Direct Station Select (DSS) Keys on a multi-line Superset or Mitel IP phone, or on a Programmable Key Module functions together with the adjacent Busy Lamp Field (BLF) indicator. While the BLF appearance indicator monitors the status of another device, the corresponding DSS key can be used to call, and transfer calls to that device. The DSS key operates when the appearance device is • **Idle** - pressing the DSS key will initiate a call to the appearance 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 connect the DSS key user to the appearance device • **Listening to Dial Tone** - pressing the DSS key will connect the 2 parties • **Dialing** - pressing the DSS key will connect the 2 parties (dialing will be suspended). The DSS key is inoperable in all other states. For example, if the appearance device is talking to one party with another party on soft hold, the DSS key will have no effect.
Direct Station Select/Busy Lamp Field (DSS/BLF)	Allows a Superset or Mitel IP telephone to have DSS keys and BLF LCDs. The LCDs indicate the status of each associated telephone. 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.
Direct Station Select/Busy Lamp Field (DSS/BLF) Call Pickup	Allows you to pickup a held or ringing call from a selected directory number (DN) with a DSS (Direct Station Select) key from a multiline Superset or Mitel IP telephone or from a Programmable Key Module. Enabling the system option DSS/BLF Call Pickup enables the BLF lamp to indicate the state of a DN instead of the state of the telephone set. The BLF lamp flashes differently to indicate a held or ringing call.
Direct Station Select/Busy Lamp Field (DSS/BLF) Interface Unit	Allows you to associate up to two PKM 48 devices with an attendant console. The DSS/BLF Interface Unit uses a separate line connection to a DNIC port. You attach the PKM 48 devices to the DSS/BLF Interface Unit and associate the PKM 48 devices with the attendant console through Customer Data Entry (CDE). You can attach up to two PKM 48 devices to the DSS/BLF Interface Unit, for a maximum of 96 DSS/BLF keys.
Direct to ARS	Allows a standard telephone to be routed directly to ARS without dialing the ARS access code, and for other devices to be routed after dialing a valid account code. The system automatically dials up to five digits for the extension.
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Table 6: Features supported by SX-200 ICP (continued)

Feature	Description
Direct Trunk Select	Gives the user direct access to an outside trunk for both incoming and outgoing calls without using trunk access codes. The trunk is assigned to a line appearance of the telephone through CDE programing. 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). The user can enter an account code while in an established call.
Display Caller ID on Non-Prime Lines	Allows set users to automatically see the caller ID on non-prime lines.
Display Keys	Allows users of Mitel display telephones to display the function of certain keys on their sets.
Distinctive Ring Tones	This feature offers 16 distinctive ring tones for DTS, CO line, and Keyline keys programmed on phones and PKMs. The DTS, CO line, and Keyline keys must have ring variants of "immediate ring" or "delay ring".
Do Not Disturb (DND)	Allows a telephone user to block incoming calls from ringing the telephone. It also prevents incoming Directed and Group Paging announcements from occurring over the set speaker. Outgoing calls are not affected.
DTMF-To-Rotary Dial Conversion	This feature automatically converts DTMF tones from DTMF equipment to rotary dial outpulsing on outgoing trunks that have been programmed as rotary dial trunks.
Emergency Call Handling	Allows you to remove call blocking for emergency calls by designating digit strings (for example; 911 or 8888) as emergency calls in ARS programming. Because you define the emergency call digits defined in ARS and Toll Control, emergency call handling is not restricted to 911 calls. This permits emergency call handling in areas where 911 service is not available or where the emergency code is another set of digits.
Emergency Calls (911) - Detection and Reporting to Attendant Consoles	This feature alerts the attendant if an extension user places a 911 call and the feature identifies the extension that placed the 911 call. With this information, the attendant is in a position to provide assistance or to direct emergency services (for example, police or ambulance personnel) to the extension where the call originated. The system also generates maintenance logs for 911 calls. This feature is not supported with a Centralized Attendant. If a 911 call is detected, ready to be delivered to a trunk, ONLY attendants ON THE SAME system as that trunk will receive the 911 reports/alarms.
Emergency Calls (911) - Detection and Reporting to Display Sets	This feature presents an audible alarm and a visual signal to the display set when an extension user places a 911 call or causes the handset to go offhook.
Emergency Calls (911) - Detection and Reporting to ONS (CLASS) Sets	This feature presents a distinctive-ringing alarm on ONS (CLASS) sets and displays a caller's location when an extension user places a 911 call, or an extension goes off-hook for an extended period.
Emergency Calls (911) - Reporting by Email	This feature sends an email to as many as three addresses whenever a 911 is dialed from a station in the system.
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Table 6: Features supported by SX-200 ICP (continued)

Feature	Description
Emergency Calls (911) - Reporting to PSAP	Allows you to assign a Customer Emergency Services Identification (CESID) number to an extension. For a call made to 911, the SX-200 ICP transmits the extension's CESID number to the public network. Where local networks are equipped with a 911 Selection Router, the CESID will be used to route the call to the Public Access Answering Point (PSAP). When the call reaches the PSAP, the CESID will be transmitted to the PSAP ALI database which will allow the dispatching system to show the location information associated with the CESID, for example street address, station number and room number. The CESID functionality requires the PRI card or the ISDN PRI Application Software Release 6.0 or greater software.
Expensive Route Warning	A trunk route can be programmed to give an Expensive Route Warning of three short tones. On Mitel display telephones and the Superset 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. Mitel Display telephones have the additional option of camping on to wait for a less expensive route or placing a callback on a less expensive route.
FAX Tone Detection	Allows an incoming FAX call on an automated attendant trunk to be detected automatically and routed directly to a FAX destination. You must purchase the FAX Tone Detection and the Automated Attendant feature.
Feature Keys	The programmable line keys on multi-line Mitel 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. For a detailed list of the feature keys, refer to "Feature Keys to activate features" on page 146.
Flash - Calibrated	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 hookswitch flash to the system. This feature is not used in North America.
Flash Control	This set of options limits the use of Consultation Hold (hookswitch 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** allows extension users to put 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** is similar to the previous option but it applies to outgoing trunks. • **Cannot Dial A Trunk After Flashing** 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 Broker's Call or 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** 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 Broker's Call or Broker's Call With Transfer in their COS.
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Table 6: Features supported by SX-200 ICP (continued)

Feature	Description
Flash Disable	An extension can be inhibited from using all services requiring the use of the hookswitch flash. For Superset and Mitel IP telephones, this prevents the extension from putting a call on Consultation Hold.
Flash For Dial 0 (Attendant)	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.
Flash For Waiting Call	Allows a user to place a call (2-party or multi-party) on consultation hold and connect to a waiting call. This is accomplished via a hookswitch flash. See Campon Warning Tone of this guide for information on allowing the telephone to receive an audible notification of waiting calls.
Flash Timing	The Flash Timer is a system wide programmable item. Its value applies to all industry-standard telephones in the system. The minimum flash timer is 200 ms. The flash timer can be programmed from 200 to 1500 ms.
Forward Campon	Calls that camp on to a multi-line Superset or Mitel IP 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.
Group Listening	Allows a user of a Mitel telephone to carry on a conversation using the handset while allowing others nearby to listen to the far end voice over the telephone set's speaker. The microphone is disabled in group listening mode.
Guest Room	See Subattendant - Guest Room Enhanced.
Handset Mute	Allows you to mute the handset microphone during an off-hook conversation. You disable or enable the handset microphone during a conversation by pressing the HANDSET MUTE key (toggle action).
Handset Receiver Volume Control	Enables the user of Mitel 5000 series IP Phones and Superset telephones to adjust the volume of the set's handset receiver. The handset receiver volume is independent of the set speaker.
Handsfree Announce	When paging to a set that is in handsfree conversation, the paging set receives a burst of busy tone. The paged set can press a "RESPOND" key to answer the page (the paging party has no indication until the paged party speaks).
Handsfree Answerback to a Directed Page	Allows users of Mitel 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 is broadcast over the users set, the set microphone is activated allowing the user to speak handsfree to the calling party.
Handsfree Operation	Enables users of Mitel telephones to have a telephone conversation without lifting the handset.
Headset Mode Feature Key	You can equip Superset and Mitel IP telephones with headsets. This feature allows you to turn headset operation on and off with a feature key.
Headset Operation	Allows you to use a Headset to make and receive telephone calls. Headset operation is not supported on the Superset 4001 and Mitel 5201 IP and 5207 IP telephones.
Headset Operation (Amplified Headset)	Allows you to equip Superset and Mitel IP telephones (excluding the 4001 and 5201 IP telephones) with amplified (externally powered) headsets instead of handsets.
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Table 6: Features supported by SX-200 ICP (continued)

Feature	Description
Headset User Control	The Superset 4000 series telephones and selected Mitel IP telephones have a separate headset jack, so the handset and headset can both be plugged in. When a headset is plugged in, the set directs the speech path to the headset and disables the handset microphone. This feature is controlled through either a feature access code or feature key. To restore to handset operation, you separate the quick-disconnect connector or unplug the headset from the dedicated headset jack.
Headset with In-Line Switch Operation	Superset 4000 and Mitel IP telephones that support headset operation can be equipped with headsets that have an in-line switch. You can use this in-line switch to answer an incoming call, terminate a call or mute the headset microphone. You can combine headset operation with the auto-answer feature for complete handsfree operation.
Hold	Allows you to temporarily suspend a telephone call. While the call is on hold, you can use the other telephone features. The call can be retrieved either at the original answer point or at another extension. Hold can be used during conferences with multiple DTS and CO lines. This hold feature differs in operation from the Temporary Consultation Hold that occurs during a Call Transfer.
Hold and Page	The Hold and Page feature provides users with the option of placing a Page after putting a call on hold. For Subattendants using Hold Position keys, the user must press the same Hold Position key within the time period to activate the page.
Hold Reminder	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.
Holiday Messages	Superset and Mitel IP telephones can display a holiday message at Christmas and New Year's. Each minute, the holiday message alternates with the usual time and date message that appears on the telephone display.
Hot Line	Individual sets can be programmed through CDE as hot lines. When the caller goes off hook, the system automatically dials a preprogrammed internal or external number. This feature is typically used for accessing taxi dispatchers or help desk operations.
Hot Swap	Allows individuals to enter a feature access code, extension, and PIN, and swap with their extension.
Hotel/Motel - Attendant Console Guest Room Softkey	Gives access to the Hotel/Motel features available at the Attendant Console.
Hotel/Motel - Attendant Message Register Audit	Allows the Attendant to print the message register for all rooms with a message register count greater than zero. These rooms are listed in the audit report in order of room number (lowest to highest). See Message Register.
Hotel/Motel - Attendant Message Waiting Setup and Cancel	A message waiting indication can be left for a guest room. The setting and clearing of a message waiting for a room can be recorded on a system printer. A single line printout is generated, giving the room number, date, time and status change. See Single Line Reports.

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Table 6: Features supported by SX-200 ICP (continued)

Feature	Description
Hotel/Motel - Audits	Provides audits (printed report that gives a record of guest room activities or conditions). Most audits are requested by the Attendant, but one (the Auto Room Status Conversion/Auto Wakeup Print) is programmed to be printed automatically each day. • **Message Register Audit:** Allows the Attendant to print the message register for all rooms with a message register count greater than zero. These rooms are listed in the audit report in order of room number (lowest to highest). See Message Register. • **Room Status Audit:** Allows the Attendant to request a report of the current status of all guest rooms. The Attendant Console produces one audit containing information on all rooms. The Front Desk Terminal can produce two types of Room Status Audits: Room Occupancy Audits and Room Condition Audits. These audits can be requested by room type, or for all rooms. See Room Occupancy and Room Condition. Each audit (Console or Front Desk Terminal) also shows the call restriction status of the room and whether there is a maid in the room. See Call Restriction on and Maid in Room Status. • **Room Type Audit:** Allows the Attendant to generate a report of all guest rooms of a particular type, giving their room number, occupancy and whether there is a maid in the room. This report can only be generated from a Front Desk Terminal. • **Wakeup Audit:** Indicates all guest rooms having wakeup calls enabled. The Attendant Console and the Front Desk Terminal generate identical reports. • **Wakeup/Room Condition Audit:** The system is programmed through CDE to print a report of all guest rooms having wakeup calls enabled, automatically, at a set time each day. At the same time it is programmed to change the occupancy and condition of all Occupied/Clean rooms to Occupied/Dirty.
Hotel/Motel - Audit Screen	The Audit Screen in entered from the House Statistics screen by pressing the Audit softkey. This screen is used to generate various kinds of audits (hard copy reports). Each kind of audit appears as a corresponding softkey.
Hotel/Motel - Wakeups	Provides automatic and personal wakeup calls to the guest room. The attendant, sub-attendant, or guest can set up multiple automatic wakeup calls that will ring the guest room at a prearranged time. The attendant or sub-attendant can set up multiple personal wakeup calls that issue a callback to the attendant or the sub-attendant so they can personally provide the wakeup call to the guest. The wakeup calls can also be set using Embedded Voicemail.
Hotel/Motel - Call Blocking	Allows the attendant or subattendant to inhibit room-to-room calls. Users of an attendant console or subattendant telephone can activate or deactivate this feature. Calls to the attendant, subattendant, or to extensions without the Call Blocking COS option selected may be made as usual. The system treats attempted calls between restricted extensions as illegal numbers and gives the calling party reorder tone. Alternatively, Call Rerouting can be used to intercept blocked calls to an appropriate destination such as the Attendant Console. See Call Rerouting.
Hotel/Motel - Call Restriction	Gives each guest room a calling privilege level (internal, local, long distance). The system automatically sets the call restriction for a room to a programmable value when an occupancy change to either vacant or occupied occurs.
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Table 6: Features supported by SX-200 ICP (continued)

Feature	Description
Hotel/Motel - Check Out	Provides a simple, fast checkout procedure. one keystroke will • Change the room occupancy field to Vacant • Set the room condition to Dirty • Erase the Guest Name • Turn off Do Not Disturb and Wakeup (if set) • Clear message registers • Apply predefined Call Restrictions to the room telephone. • Delete Last Number Redial for the room.
Hotel/Motel - CLASS (station side) for Analog Telephones	Refer to CLASS for Analog Telephones.
Hotel/Motel - Do Not Disturb (DND)	Blocks calls from ringing at a guest's telephone. Outgoing calls are not affected.
Hotel/Motel - Front Desk Features	The Front Desk features are all accessed through the House Statistics Screen on the Front Desk Terminal (VT100). This screen appears on start up and contains the current status of the guest rooms as well as the time of the last status update. Four additional screens can be accessed via the House Statistics screen.
Hotel/Motel - Guest Names	The Front Desk Terminal can enter and store Guest Names in two data fields. Guest names can be displayed on an Attendants Console if entered from the Front desk Terminal or a Superset 4150 or Superset 430 telephone, but they cannot be entered or searched for.
Hotel/Motel - Guest Room Message Retrieval	A message waiting indication set by the Attendant can be • A flashing lamp on the telephone. • A distinctive ringing pattern every 20 minutes. The telephone rings with this distinctive ringing pattern if the extension has been busy, or has Do Not Disturb set, or until message waiting is canceled. • A message on the display of a Superset 4150 or Superset 430 telephone (if used as a guest room telephone).
Hotel/Motel - Guest Room Template Programming	A block of guest room Superset or Mitel IP telephones can be programmed with Speed Dial and Feature Access keys. This can only be done through CDE. It is possible to program up to three separate blocks of telephones with unique speed dial and feature access keys.
Hotel/Motel - Guest Room Update Screen	This screen is accessed through the Room Update softkey and it greatly simplifies the guest check in and check out procedure. From this screen Guest Name, Room Occupancy Status, Room Condition, and Call Privilege can be entered or changed; Wakeup time, Message Waiting and Do Not Disturb can be set or cleared; and the Message Register can be cleared. A simple method of checking out a guest is provided via a softkey that is accessed through the Occupancy field. See Check Out.
Hotel/Motel - Guest Search Screen	The screen is accessed via the Guest Search softkey on the House Statistics Screen. This search facility allows searching by last name. A partial or complete text string can also be entered. All names matching the input string are displayed. See Guest Names.

Table 6: Features supported by SX-200 ICP (continued)

Feature	Description
Hotel/Motel - House Statistics Screen	The House Statistics Screen displays the following summaries. • **Guest Room Occupancy Summary:** Vacant, Occupied, Reserved, and Guaranteed. • **Room Conditions Summary:** Clean, To be cleaned, To be inspected, Not in service, and Maid present. • **Feature Usage Summary:** Do Not Disturb on, Wakeup set, Message Waiting on, Non Zero Message Registers, and Call Blocking.
Hotel/Motel - Internal Number Block	Provides room number confidentiality in Hotels. The hotel operator has the choice of displaying the room number on the telephone display for Guest to Guest calls.
Hotel/Motel - Maid in Room Status Display - Superset and Mitel IP Display Telephones	This feature allows an authorized Superset and Mitel Networks IP display telephone to determine which guest rooms have maids in them.
Hotel/Motel - Message Lamp Test	The Message Waiting Lamp on a guest room telephone is tested automatically whenever the room status changes from occupied to vacant and there is no message waiting. It runs whether the change was made from a Front Desk Terminal of from an Attendants Console. However there is no indication at the Front Desk Terminal that it has run. If there is a failure, notification is through the alarm icon at the Console. The test verifies lamp operation and confirms that the telephone is still connected in the room. The test does not verify bell operation.
Hotel/Motel - Message Register	Tracks the number of completed external calls or call units for each extension. There are two modes of operation. The first counts the number of external calls made by each room. The second keeps track of meter pulses being sent from the far end to the associated trunk circuit. These pulses can be used to determine the amount charged against the guestroom making the call. The Attendant Console displays the current value of the message register for a room each time a room number is entered. The message register can be cleared by the Attendant from the Console or automatically upon requesting an audit. Clearing the message register can be recorded on a system printout. Meter pulses are recorded in SMDR.
Hotel/Motel - Multi-user	Four front desk terminals can run the Hotel/Motel application at the same time. However, two terminals cannot edit information for the same room at the same time. The Front Desk Terminal also checks that the room is not being accessed by an Attendant Console. If it is, the message “Room being accessed by another user. Try again later.” appears on the screen. Up to11 consoles can be configured on one system. Two (or more) consoles can access a guest room at the same time.
Hotel/Motel - Passwords	Entering or changing guest room information from the Front Desk can be controlled by passwords. The user of an attendant password can read information about rooms, request audits, and conduct searches. The user of a supervisor password can, in addition, enter and change information about a guest room, since this password presents the Guest Room softkey.
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Table 6: Features supported by SX-200 ICP (continued)

Feature	Description
Hotel/Motel - Property Management System (PMS)	The SX-200 ICP can interface to a Property Management System (PMS) to provide integration of system and PMS functions. A PMS provides a computerized method of controlling and monitoring hotel/motel functions. The system interfaces to personal computers with the Lodgistix PMS software package (or a package that follows the same protocol) through a serial-to-IP connection. Hotel/Motel information is stored in the PMS. The PMS Room Status feature and the Console or Front Desk Terminal Room status are mutually exclusive. See Room Status Display.The PMS interface maintains the following information between the system and the PMS: • **Automatic Wakeup:** This feature allows the Attendant to enable or disable an automatic wakeup on a room phone from a PMS terminal. • **Check in/out:** When a guest checks in (PMS Check In), the room telephone is enabled to allow outgoing trunk calls. The attendant may restrict the room phone to internal calls, local calls, or long distance calls using the Outgoing Call Restriction feature described in the Attendant Console Guide. Upon check out (PMS Check Out), the phone is disabled from making calls and the PMS clears Message Register, Message Waiting, Do Not Disturb, and Wakeups from the guest room database. • **Guest name:** This feature allows the name of a guest to be associated with a room in the PMS. It is sent to the system, when a guest checks in, and is stored against the room extension. • **Maid in room status:** This feature allows the maid to change the room status (clean/dirty) from the room telephone. The Maid in room status is also indicated on the PMS terminal. This feature is functionally identical to that of the attendant room status, however the displaying and monitoring of room status is completely controlled from the PMS terminal. • **Message registration of outgoing trunk calls:** This feature provides the PMS with the number of trunk calls made from a room (local and long distance). A call-accounting device connected to the system monitors SMDR reports for long distance calls. The charge for these calls is automatically added to the guest's bill at check out time. Call-costing equipment may be attached to the PMS to allow the PMS to handle call costing. • **Message waiting:** This feature allows the Attendant to enable or disable the Message Waiting Lamp on a room phone from a PMS terminal. • **Confirm Wakeup by Offhook:** This feature allows users of Superset and Mitel IP display telephones to acknowledge a wakeup by going offhook, instead of having to read the message and press a softkey.
Hotel/Motel - Room Condition	Indicates the current housekeeping condition of a guest room: Clean, Dirty, Out of service, or To be inspected. It can be set from the Attendant Console or the Front Desk Terminal. Some conditions can be set from the guest room telephone. For more information see Room Status Display and Maid in Room Status Display on. Room Condition can be displayed as part of the Room Status display on the Superset or Mitel IP display telephone.
Hotel/Motel - Room Occupancy	Indicates the current occupancy of a guest room. It can be set from the Attendant Console or the Front Desk Terminal. The four types of Room Occupancy are: Vacant, Occupied, Reserved, or Guaranteed. See Room Status Display.
Hotel/Motel - Room Search Screen	Allows search on nine criteria (Dirty, Guaranteed, Maid in Room, Reserved, Room Number, Room Type, Service, Vac, Vac/Clean).
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Table 6: Features supported by SX-200 ICP (continued)

Feature	Description
Hotel/Motel - Room Status Display	Allows the Attendant to display and change the status of one or more rooms. Room status is made up of: Room Occupancy, Room Condition, Telephone Privileges (call restrictions) and Maid in Room. Each (except Maid in Room) can be set independently of the other by the Attendant. The system can be programmed to change the status of all "occupied/clean" rooms to "occupied/dirty" at a predetermined time. At the same time the system generates a list of all Auto Wakeup requests. Room Status is displayed on an attendant console, a front desk terminal, or a Superset or Mitel IP display telephone. Displays information on the Attendant Console, Front Desk Terminal, Superset Display Telephone. Maid in Room is displayed separately as part of the Maid in Room feature. Call restriction is not displayed.
Hotel/Motel - Room Types and Room Codes	With this feature hotel guest rooms can be divided into 50 different types, such as single, double, queen, smoking, and nonsmoking. This is done through customer data entry (CDE) programming by putting each room type in a separate Class of Service (COS). Since each COS can have a different name associated with it, the room is identified by the COS name. By default, it can be identified by the code (COS number) associated with the COS name. Searches and audits can be requested by room type or code. Since the Front Desk has an alphanumeric keyboard, the COS name can be alphabetic. When an alphabetic name is entered on the Front Desk Terminal, it can be displayed on the Attendant Console.
Hotel/Motel - Single Line Reports	Audits used to record changes in status for individual rooms. These reports are generated automatically, and provide hard-copy evidence that a change has occurred. The printouts produced by Single Line Reports are limited to 40 characters in length and start with the room extension number, date and time. There are three categories of single line reports: • Wakeups • Message Registration • Message Waiting.
Hotel/Motel - Suite Services	Associates multiple telephones with one another in a hotel suite for basic call handling, call privileges, SMDR, room check-in and check-out, Caller ID, messaging, and call forwarding.
Hunt Groups	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, 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.
I Hold You Hold	Allows users who share lines to identify which phone put a call on hold. The line key LED on the phone where the call was placed on hold flashes green (I Hold). The LED flashes red for users with a line appearance of that held call (You Hold). I Hold You Hold is supported by Mitel 5212/5312 IP Phones only.
Illegal Access Intercept	Calls to restricted access codes or extension numbers can be routed to an answering point for completion. The illegal access intercept point can be an LDN position on the attendant console or any valid reroute point. Illegal number intercept points can be programmed to be different for DAY, NIGHT1, and NIGHT2 operation.

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Table 6: Features supported by SX-200 ICP (continued)

Feature	Description
Inhibit Trunk Ring-Me-Back During Dialing	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 recall the station and is instead dropped. This prevents a trunk from locking when the flash on the trunk was intended as a hangup and the station user did not expect a trunk to be on consultation hold.
Intercept To Recorded Announcement	Incoming trunk calls can be intercepted to groups of recording devices after dialing vacant numbers, reaching busy extensions, getting no answer, or as required.
Intercom Calls	Determines whether a phone receiving an intercom call is paged or rung.
Internal Number Block	Blocks the number of the station/set on the display of Superset 4000, Superset 400, and Mitel IP telephones and ONS/CLASS telephones on the same system.
Inward Restriction (DID)	An extension can be barred from receiving calls directly from DID trunks.
Language Change	Allows Mitel display telephones to display text and softkey prompts in a different language.
Last Number Redial	Allows the Attendant Console and any telephone user to redial the last manually dialed internal or external number with a single key operation.
Last Party Receives Dial Tone	Allows the last party left on a call, after the other parties) hang up, to receive dial tone and be able to dial. Normally, this party would receive silence and after 30 seconds be locked out.
LCD Display	Liquid crystal displays on Mitel telephones so equipped show 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 display telephone accesses a trunk and establishes a call, the duration of the call is displayed.
Line Lockout	The system 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.
Line Preference	Allows the system to automatically select which line is used when the set goes Off-hook 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. This feature has no effect on the answering of calls.
Line Privacy	This feature ensures that, if desired, conversations on Key, Direct Trunk Select, CO Line, and Private Trunk Lines are private. When such a line is in use at one set, other appearances of the line cannot join the conversation.
Line Selection	When the user starts dialing, the system selects a line (if programmed to do this). When the set is ringing and the user goes off hook, the system selects the line to answer.
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Table 6: Features supported by SX-200 ICP (continued)

Feature	Description
Line Types and Appearances for Superset and Mitel IP Telephones	Superset and Mitel IP telephones are equipped with keys that can be used as Line Select Keys. These lines can provide additional calling capability, direct access to calls appearing at other sets, and direct access to trunks. There are seven types of lines: • **Prime Line:** The SX-200 ICP identifies each Superset or Mitel IP telephone by an extension number known as the Prime Line. The Prime Line is always a Both-Way and Immediate-Ring Line. A Prime Line can be changed to operate as a Multicall or Key Line by giving the line an appearance at another set. • **Key Line:** A Key Line is an appearance of one extension number on two or more telephones. This extension number can be for another Superset or Mitel IP telephone or an industry-standard telephone. If the line is in use at one set, the other appearances of the line are busy and unavailable. • **Multicall Line:** A Multicall Line is an appearance of one extension number on two or more telephones. This extension number can be for another Superset or Mitel IP 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. • -**Direct Trunk Select (DTS) Line:** A DTS Line operates like a Key Line, but it directly accesses a specified dedicated CO trunk. It can be used for incoming and outgoing calls. For further information see Direct Trunk Select. The DTS trunk can appear on up to 64 Superset or Mitel IP telephones. • **Private Line:** A Private Line accesses a dedicated CO trunk directly. As operating TELCOs often provide a less expensive rate for trunks connected to private trunk lines, Superset or Mitel IP telephones accommodate this type of operation. The user can transfer established calls on this line only to other Superset or Mitel IP telephones that have an appearance of the line, using privacy release. • **BLF/DSS Line:** This line type is used by two distinct features; Busy Lamp Field (BLF), and optionally Direct Station Select (DSS). A BLF is an appearance of a station, Superset, logical line or trunk. The LED indicator indicates the state of the BLF appearance (Idle, Busy, DND). A DSS key is a BLF appearance (of a station, Superset telephone, Mitel IP telephone, or logical line), associated with the key. Some line types can appear on only one Superset or Mitel IP telephone. Other line types can appear on more than one telephone. Because the same line may appear on up to a maximum of 64 other sets, the telephone and the lines are not always busy at the same time. One or more lines may be in use but the telephone is idle and available for a call.
Line Appearance Variants	Through CDE, the seven line types can be programmed to control call direction and ringing. Multicall Lines can also be programmed as Secretarial Lines. • **Direction:** Direction can restrict calling for an appearance of a line to Both-Way, Incoming-Only, or Outgoing-Only. The outgoing direction for a line on a set is only available if the line is programmed for No Ring. If programmed for Delayed Ring or Immediate Ring the line must be either Both Way or Incoming Only. • **Ring:** The ring option determines whether new calls to a Superset or Mitel IP telephone line appearance ring immediately, ring after a delay or not ring at all. • **Secretarial:** The secretarial feature interacts with the Do Not Disturb feature for improved call handling. See Secretarial Line.
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Table 6: Features supported by SX-200 ICP (continued)

Feature	Description
Lockout Alarm	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.
Logical Lines	A Logical Line is a line on a Superset or Mitel IP telephone that is not an appearance of any station or other Superset telephone. Each Logical Line has its own extension number and can exist on up to 64 different Superset or Mitel IP telephones. Logical Line extension numbers can be used in many places where telephone lines can be programmed such as Call Rerouting.
LS Measurement Tests	This tool allows you to obtain optimum circuit descriptor settings for Loop Start (LS) trunks connected to the Analog Main Board and Analog Option Board (onboard ASUs) in the controller. The tests are based on the signals received from the CO. Correct programming reduces the possibility of echo between the trunks and IP phones.
Mailbox Key	A Mailbox Key is the line key on a telephone or programmable key module (PKM) that has been programmed with the extension number of a device with an associated mailbox. Subscribers can use Mailbox Keys to receive notification of new messages, and to access their voice mailboxes to listen to messages. Typically, the administrator will configure a number of Mailbox Keys on a single telephone or PKM. This enables multiple subscribers to manage their voice mail from one location. For example, although a number of teachers may share a single telephone in a school staff room, each teacher can have a personal Mailbox Key on the telephone. When the teachers receive new messages, the LEDs corresponding to their Mailbox Keys will flash.
Mailbox Lockout	If this feature is enabled, the system locks the mailbox after three failed attempts to log in with an invalid password. When lockout occurs, the system plays a message that instructs the user to contact the system administrator in order to reset the password. The feature is disabled by default.
Manual Line (Dial 0 Hotline	When a Manual Line extension goes off-hook it is routed directly to the extension's dial 0 routing point. The extension can still receive calls.
Message Flash Notification	When an IP phone receives a new call, or when it receives a call while busy with another call, the message lamp flashes.
Messaging - Advisory	Allows users of Superset or Mitel IP display telephones to provide a short message to be displayed on any Superset, Mitel IP display telephone, or attendant console that calls the set. The message replaces the time and date display on the sets where it is activated. The system provides 15 system-wide messages.
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Table 6: Features supported by SX-200 ICP (continued)

Feature	Description
Messaging - Call Me Back	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 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 when the set is idle or during a call. On Superset or Mitel IP display telephones, the display shows 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 when the message is sent from the Console, the Front Desk Terminal or PMS.
Meter Pulse Collection	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 system without additional hardware include • Tip-Ring reversals • XT lead signaling (Analog CO Trunk) • M&MM lead signaling (Digital LS/GS Trunk). Other types of Meter Pulses common in the telephone industry include 50 Hz, 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 Vdc 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 Property Management System for additional information.
Moving Stations and Superset Telephones	Allows extensions to be moved easily from one circuit to another. Previous programming for the extension, such as name, COR, and COS, is preserved and moved with it.
Multi-Attendant Positions	The system can handle multiple attendant consoles, giving unique hold slots to 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, 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. Also see Recall, and Attendant Console LDN Keys, Attendant Transparent Multi-Console Operation and Tenanting.

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Table 6: Features supported by SX-200 ICP (continued)

Feature	Description
Music-on-Hold (MOH)	A customer-provided music source can be connected to the SX-200 ICP via the Music-On-Hold connector on the back panel of the cabinet, the Music-on-Hold/Pager Module on the Universal card, or on the Music-on-Hold/Pager Module via a DNIC port, or via an ONS port (see below). Music-on-Hold can be used with Campon, Hold, Universal Call Distribution (UCD), ACD, and other features. Each tenant of the system can also have its own Music-on-Hold source through a DMP unit that is connected to a DNIC port. Each DMP unit is accessed by a directory number programmed in its tenant group. Up to 76 Music-On-Hold sources can be programmed (75, for the 25 tenants in Day, Night 1 and Night 2 service, and the system source). Use of the Music-on-Hold feature may require, under applicable copyright or other provincial, local, state and/or federal rules, regulations and/or statutes, that you obtain a license from the local performing rights society or copyright owner before you can provide music on hold to telephone users. Contact your music supplier for more information.
Music from an ONS Source	A customer-provided ONS music source can be connected to the system via an ONS port, which provides a cost-saving alternative to music from DMP and Music-on-Hold/Pager sources. Music from an ONS Source can be used with Campon, Hold, Universal Call Distribution (UCD), ACD, and other features. This feature also supports system tenants. Use of the Music from an ONS Source feature may require, under applicable copyright or other provincial, local, state and/or federal rules, regulations and/or statutes, that you obtain a license from the local performing rights society or copyright owner before you can provide music on hold to telephone users. Contact your music supplier for more information.
Names	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. A user of a Superset or Mitel IP display telephone can program their name from their telephone.
Never a Consultee	Protects an extension from being dialed or retrieved by extensions that have a Consultation Hold in progress.
Never a Forwardee	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.
New Call Ring	When a Superset or Mitel IP 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.
NI3 Calling Name Delivery	Allows the called party to see the name of the caller on the display screen of the telephone on incoming calls. The NI3 protocol also allows a link between calling name and calling number for outgoing calls. The caller can set the calling party number presentation indicator to "Allowed", and the calling name stored at the Central Office will be displayed with the calling number. The presentation of calling number and calling name can be allowed or prohibited through IMAT programming.
Night Bells	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 extension number assigned to the Night Bell can be used as an answer point or alternate answer point for most features in the system. The system provides a contact closure which operates the alerting device. Night Bells are activated by relays on analog main board in the SX-200 ICP controller, a Universal Card receiver/relay module or on the Music-on-Hold/Pager Module via a DNIC port.
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Table 6: Features supported by SX-200 ICP (continued)

Feature	Description
Night/Day Switching	A Superset or Mitel IP display telephone can put the system (or particular tenant group or groups) into DAY Service or one of two Night Service modes, NIGHT1 or NIGHT2. In Night Service the telephones display NIGHT1 Service or NIGHT2 Service as appropriate.
Night Services	The system has three different service modes: DAY, NIGHT1, or NIGHT2. When the system or tenant group is in Night-Service mode, incoming trunk calls and calls to the Attendant can be rerouted to specified extensions or activate common alerting devices (Night Bells).
Night Services Flexibility	Allows the Attendant to change the Night Service assignment of non-dial-in trunks. The system allows full flexibility of trunk assignment.
Node Identification	The Node Identification feature works with the Analog Networking feature to provide consistent dialing of extension numbers throughout a network of SX-200 ICP systems. 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.
Non-Busy Extension	An extension with the Non-Busy Extension feature enabled can have a maximum of 5 parties connected and never appears busy to the system. If a new call is directed to a non busy extension that is already in a call, the system automatically overrides the existing call. After a warning tone, the new caller joins the conversation.
Numbering Plan Flexibility (Conflict Dialing)	The numbering plan used within the system is completely flexible. The system can be programmed through CDE with any combination of 1-, 2-, 3-, 4-, and 5-digit numbers. Also see Conflict Dialing.
Off-Hook Alarm to Display Sets	Notifies a user of a display set that a set user has put the handset offhook.
Off-Hook Voice Announce	Allows a party to place a directed page to a busy Mitel display telephone that is not in handsfree mode. The announcement is heard through the speaker, only by the paged party.
Off Premises Extension (OPS)	Industry-standard telephones not in the immediate vicinity of the system can be directly connected to the system without the use of special trunks using a six-circuit OPS (Off Premises) Line Card.
Originate Only Extension	Allows an extension 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.
Overlap Outpulsing	Overlap Outpulsing occurs when the system 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.
Override (Intrude)	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. Superset and Mitel IP telephones display the name and/or extension number of the overriding party.
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Table 6: Features supported by SX-200 ICP (continued)

Feature	Description
ONS Positive Disconnect	ONS Positive Disconnect forces an ONS device to go on-hook when the far-end disconnects. A momentary electrical break on the ONS device forces the disconnection. Some FAX machines or answering machines require this electrical break to recognize that the other party has terminated the call.
ONS Ring Groups	The ONS Ring Groups feature provides the ability of multiple ONS telephones to ring when a master telephone is called. Each ONS telephone in the ring group still has its own extension number, therefore allowing the ONS telephone user to accept calls intended for that telephone plus answer calls directed to the master telephone. Each user can also answer his/her own calls and have his/her own voice mail.
Override Security	Provides an extension, DISA trunk, or dial-in tie trunk with security against Override. See Attendant Busy Override and Override (Intrude) on/
Paged Party Page Tone	Allows the paged party to differentiate between ringback tone and paging tone. Paging tone applies to the pager and the paged party for Set-to-Set Pages, Group Pages, and All Set Pages.
Paging - PA	Paging equipment can be connected to the SX-200 ICP via the Paging connector on the back panel of cabinet, a Paging/Music-on-Hold module on the Universal Card or on the DNIC Music-on-Hold/Pager Unit (DMP) via a DNIC port. Up to nine paging zones, with separate or simultaneous access, can be provided. An extension, tie trunk, or DISA trunk can 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. See Attendant Paging Access. Also see Paging - PA and Telephones.
Directed Page	Allows a party to page a specific 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 Prime key.
Group 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.
Meet Me Answer:	Allows a party to respond to a group page. It does not apply to Directed Page. 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 - after this, the system cancels the page.
All Set Page	Allows a party to page all telephones simultaneously via their telephone speakers. The connection(s) are one-way audio to each telephone, 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.
Paging - PA and Telephones	Enables overhead paging with telephone set paging. The attendant console user or the telephone set user can initiate a page that includes the PA with Group Pages or All Set Pages.
Park and Page	Combines the call parking in an auto-selected orbit and paging operations in one step.
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Table 6: Features supported by SX-200 ICP (continued)

Feature	Description
Parallel Connection of Industry-Standard Telephones	A maximum of three industry-standard telephones equipped with bells can be connected (hard-wired) together on one ONS line.
Personal Speed Call	Allows the user 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. Note that in addition to this feature, users also have access to the Abbreviated Dial feature - see Abbreviated Dial.
Pickup Groups	Extensions may be programmed as Pickup Groups, permitting users to answer calls to any other extension within their particular group. See Pickup - Local and Directed.
Phonebook	Provides access to the voice mail directory which allows callers to reach an extension by entering the user's first or last name rather than their extension or mailbox number. The voice mail system can be configured to search either on first or last names (but not both at the same time).
Phone Twinning	Provides concurrent ringing and message waiting indication on as many as five phones. With twinning, a user who has phones in different locations, or a wireline (desk) phone and a Wireless phone, can receive calls and message waiting indications on either device. Twinning also benefits teleworkers who have a phone at the corporate office and a remote IP phone at home. Users can turn off/on the ringer on the phone that they are not using as well as enable/disable the SUPERKEY to prevent tampering with the phone's programming.
Pickup - Local and Directed	**Local Pickup**: A telephone can be assigned to a pickup group, and can answer any ringing telephone within that group. T **Directed Call Pickup**: Allows an extension user to answer any ringing telephone within the system.
Ping Command	Use the ping command to send an echo request to a network host.
PRI Card Support	Allows a PRI card to be installed in the system. With the T1/E1 module, the PRI card provides one or two links (23 or 46 channels) of T1 ISDN connectivity.
Printer/Terminal Support	Allows the routing of printouts to the system printer port, to any data port, an SX-200 ICP IP socket, or to the printer port on the SUPERCONSOLE 1000 Attendant Console. All printer ports are RS-232C interface.
Priority Dial 0	The Priority Dial 0 feature can be used to provide an alternate dial 0 routing for extensions in the system. Priority Dial 0 and Dial 0 have separate DAY/NIGHT routing points.
Privacy Enable/Privacy Release	A multi-line Superset or Mitel IP telephone may have 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, other sets with an appearance of the same line are denied access. The user of the line can, however, use the Privacy Release feature to allow the other sets to join the conversation. See Line Privacy. If the customer wishes to have their calls public to begin with (privacy released at the beginning of a call), the COS option "Privacy Released at Start of Call" may be enabled. To obtain privacy, the user then presses a Make Private softkey or a Privacy Release feature key.
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Table 6: Features supported by SX-200 ICP (continued)

Feature	Description
Q.SIG	With the Q.SIG MOSS option, the SX-200 ICP can connect with any Q.SIG compatible SX-2000 LIGHT or 3300 ICP system or any other Q.SIG compatible PBX to form a private network. The SX-200 ICP provides end node functionality, which means that the system can only connect to one other system in the network. Q.SIG supports incoming calls, incoming Calling Name, Message Waiting Indication, Call Transfer, Call Diversion, Call Offer, and Path Replacement (Partial).
RAD Support	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 the ACD, UCD, Hotel/Motel Wakeup, Automatic Attendant Overflow, and Automated Attendant features. For ACD, Attendant Automatic Overflow and Automated Attendant, more than one caller at a time 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. See Attendant Automatic Overflow, Automated Attendant, and Uniform Call Distribution (UCD). Also see Wakeups.
Recall	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. Recall 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.
Receive Only Extensions	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.
Record a Call	Record a Call allows you to record both ends of a two-party conversation (internal or external call) in progress at your set. The recorded conversation is stored in your voice mail mailbox. This feature is a purchasable option.
Remote LAN Access	Provides LAN access to the wide area network (WAN) for both incoming and outgoing calls through LAN servers (for example, routers, bridges) using Mitel's PRI Gateway interface.
Reminder	Allows an extension user to program the telephone to ring at a particular time. This can be used, for example, as an appointment reminder. You may program up to three timers (in a 24-hour period) to occur once or to repeat daily. See Wakeups
Resale Package	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 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 a specialized application of the SX-200 Automatic Route Selection, Toll Control, and Verified Account Code features.
Ringer Control	Allows users of a Superset or Mitel IP telephone or an attendant console to adjust the volume of the ringing telephone or console.
Ringing - Discriminating	This feature provides two different ringing cadences to allow a user to distinguish between internal incoming calls (standard ringing) and external incoming or Attendant calls (discriminating ringing). The system can also be programmed to provide discriminating ringing for all calls.
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Table 6: Features supported by SX-200 ICP (continued)

Feature	Description
Ringing Plan	The SX-200 ICP are fully compatible with the North American public switched network using the North American ringing plan. The ringing plan is stored in the database.
Ringing Time-Out (Final Ringback)	A call to an extension can ring for 1 to 30 minutes before the call is dropped. The default ringing time is 1 minute.
Satellite PBX	The SX-200 ICP can be installed as a satellite PBX. In this configuration, the PBX has no direct connection to the serving central office for incoming traffic. Enabling the satellite PBX system option automatically adjusts any required settings for the loss and level plan.
Secretarial Line	A Superset or Mitel IP telephone programmed with a secretarial multicall line appearance of another extension can override the Do Not Disturb feature on the second set. In a typical operation the second telephone has Do Not Disturb active and the first telephone answers the calls. The secretary can override Do Not Disturb at any time by making the call on one of the multicall 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.
SMTP Client	The SX-200 ICP offers the following SMTP client features: • **Auto-Forward Voice mail to E-mail** - When programmed in a user's mailbox, this feature automatically forwards the user's voice mail messages to an e-mail address. In addition to voice mail forwarding, the SX-200 ICP can also send Record-a-call messages to e-mail which can be Played, archived or forwarded to another e-mail address. When using Record-A-Call it is recommended that the Voice mail system be upgraded to a hard drive. • **Notification of E911 calls** - When a user dials E911, a distribution list (maximum 3 users) will be sent an e-mail with the subject "E911". The body of the e-mail includes the caller's name, extension number and location (if programmed). Also, for accountability and potential liability purposes, a log with the same information shall be generated when the e-mail is sent. • **Notification of Alarms** - If the SX-200 ICP system detects an Alarm, the system will create an alarm log that can be sent in an e-mail message to three different addresses with the subject "Alarm Notification". The body of the e-mail can include - minor, major or critical alarms. • **On-demand Maintenance Logs** - The SX-200 ICP advanced Maintenance feature can be used to send higher level maintenance logs directly to Mitel Technical support saving on site trouble shooting time.
Speak@Ease Support (Mitel Speech Server)	Users of the Speak@Ease softkey can place a call by a spoken command. The Mitel Speech-Enabled Applications is the name of the software that enables the Speak@Ease button functionality.
Speaker Volume Control	Allows users of a Superset or Mitel IP telephone to adjust the volume of the telephone’s speaker
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Table 6: Features supported by SX-200 ICP (continued)

Feature	Description
Speed Call Key	Allows a 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. Users can also include a pause character in the speed dial numbers they program on their telephones. This allows them, for example, to dial through an auto attendant to an extension, or to dial an internal voice mail machine and password with a single keystroke. This feature also makes it easier to send out a FAX and to access long distance service providers.
Split	Allows a Superset or Mitel IP 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. Split can be used during conferences with multiple DTS or CO lines.
Station Message Detail Recording (SMDR)	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.
Subattendant - Basic Function	This feature provides a Superset or Mitel IP 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. Usually, a Superset or Mitel IP telephone is considered to be busy when the set and/or the prime line appearances are busy. For a Subattendant, the set is busy only if the prime and all of the appearances of the prime line are busy. The state the telephone itself is not checked. This allows as many callers as there are appearances to call the telephone under some circumstances. This special line appearance checking makes the set a better backup position. The Night/Day Switching feature can be used to allow the subattendant to select DAY, NIGHT1, or NIGHT2 service for the system. The Subattendant telephone can also be used as the Alternate Trunk Recall point. Note: This Subattendant - Basic Function feature is not related to, and does not interact with, the Subattendant - Enhanced Function features.
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Table 6: Features supported by SX-200 ICP (continued)

Feature	Description
Subattendant - Enhanced Functions	The Enhanced Subattendant functions allow Mitel 5224, 5330, and 5340 IP telephones to be used as Subattendant stations 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 these telephones remain intact; the Enhanced Subattendant functions are an addition. The Enhanced Subattendant position can perform functions for extension users; set up and cancel Call Forwarding, set up and cancel Advisory Messages, and toggle Do Not Disturb on and off. The Enhance Subattendant features include • System Abbreviated Dial Programming • Station Advisory Message Programming • Station Call Forward Setup and Cancel • Calls Waiting Indication • Hold Positions - 6 • Listed Directory Number (LDN) Keys - 6 • Enhanced Zone Paging • Station Do Not Disturb Setup and Cancel • System Date and Time Setup • Enhanced Recalling Capabilities.
Subattendant - Abbreviated Dial Programming	The Subattendant - Abbreviated Dial Programming feature allows the Subattendant to program system abbreviated dial numbers from the Subattendant set. The Subattendant has the option of making abbreviated dial numbers confidential.
Subattendant - Advisory Message Setup	There are eight default messages and seven programmable messages that the Subattendant may set up on behalf of another extension. The Subattendant can read a currently displayed message, or read through the available messages and choose one for display on the set, or program one for display.
Subattendant - Wakeups	Allows a Mitel 5224, 5330, or 5340 telephone programmed as a subattendant to set up wakeup alarm calls that ring the guest room telephone at a prearranged time. With MOSS Option 102 (Feature Level) enabled to level 1 or greater, users of the Superset 4150 and Superset 430 subattendant telephones can program wakeup calls (multiple and personal) that occur once or repeat daily. You may set a maximum of three wakeup calls in a 24-hour period. Personal wakeup calls issue a callback to the subattendant telephone so you may personally provide a wakeup to the VIP guest.The programmed wakeup times can be changed or cancelled.
Subattendant - Call Forward Setup and Cancel	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.
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Table 6: Features supported by SX-200 ICP (continued)

Feature	Description
Subattendant - Calls Waiting Indication	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 appears on the Subattendant set and any calls ringing the night bell. Each new call ringing the Subattendant position increments the indicator; similarly, the indicator is decremented each time a caller hangs up.
Subattendant - Date and Time Setup	When the Subattendant position is idle, it continually displays the time and date (day, month year) on the LCD display. The time may be displayed in 12- or 24-hour format depending on the system feature settings. The Subattendant can change the time and/or date by using the Subattendant SUPERKEY and softkeys.
Subattendant - Guest Room Enhanced	The Guest Room feature converts the Mitel 5340 IP Phone into an enhanced subattendant with capabilities for room occupancy/status updates, multiple wakeup programming, outgoing call restriction programming, and increased reporting abilities. The Guest Room feature provides the subattendant feature support of the Superset 4150 and more.
Subattendant - Hold Positions	Provides the Subattendant with up to six hold position keys. When enabled, the hold position keys permit the Attendant to answer other LDN or Prime lines without having to release current calls on the Subattendants telephone first. 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. If 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 cannot be answered until the call on hard hold is released.
Subattendant - Listed Directory Number (LDN) Keys	Each Subattendant can have up to six keys programmed as Listed Directory Number (LDN) keys. The LDN keys appear on the Subattendants 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 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 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. Once answered by the subattendant, an LDN call is treated as though it were a regular call received on a Superset or Mitel IP telephone, with the exception of Serial Calls.
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Table 6: Features supported by SX-200 ICP (continued)

Feature	Description
Subattendant - Paged Hold Access	The Subattendant can place an incoming call on hold, page the called party and inform them 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, 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. See Paging - PA.
Subattendant - Recall	Ensures that calls do not remain unanswered or on hold for an unlimited period of time. Any calls that have been extended by a Subattendant, recalls the Subattendant position if the call is not answered or remains on hold at the end of the timeout period. The 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 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 Recalls tying up the prime line of the Subattendant it is important to program the Recall key. Recalls to the Subattendant will then be queued on the Recall key.
Subattendant - Station DND Setup	The Subattendant may set up or cancel Do Not Disturb (DND) for an extension by selecting the Do Not Disturb softkey. Selection of the Do Not Disturb softkey turns the feature on. Selecting the softkey when the Do Not Disturb feature is activated, turns the feature off. The status change will be indicated on the Subattendants display and on the corresponding extension. See Do Not Disturb.
Superset 3DN and Superset 4DN Auto-Answer For Directed Page Calls	Enables users to respond handsfree to a directed page. If a directed page from a telephone is broadcast over the user's set, the set microphone is activated automatically allowing the user to speak handsfree to the calling party. The Handsfree Answerback feature provides similar functionality on other Mitel telephones.
SUPPORT Superset 3DN, 4DN, and 400 Series Set Types	Support for Superset 3DN, Superset 4DN and Superset 400 series telephones is a purchasable option. To use these sets, you must enable Option 98 "Support 3DN and 4DN Set Types" in Form 4, System Options/System Timers.
Swap (Trade Calls)	Permits a Superset or Mitel IP telephone user to switch the 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.
Swap Campon	Allows the user of a Superset or Mitel IP telephone to put the current call on hold and speak with a camped on party. The telephone user can alternate between the two calls as required, form a three-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.
System Identifier	A unique one- to three-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, ACD SMDR, and Analog Networking.
System ID Module	The system ID module enables operation of the features that were purchased. Removal of the system ID module generates a MAJOR alarm.
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Table 6: Features supported by SX-200 ICP (continued)

Feature	Description
Tandem Operation	The SX-200 ICP supports two PBXs connected in tandem using tie trunks to connect the two systems together. See the Analog Networking, Satellite PBX, and Resale Package features.
Tenanting	Using the tenanting feature, up to 25 small businesses, or departments of a larger business, can share the services of an SX-200 ICP. Each tenant can be provided with customized features and services.
Toll Control	The Toll Control feature forms part of the Automatic Route Selection (ARS) feature. It allows the system 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).
Tone Demonstration	Familiarizes users with the tones the system generates. This feature also allows Superset or Mitel IP telephone users to adjust ringer volume and pitch.
Tone Plans	The SX-200 ICP is compatible with countries that use the North American tone plan only. See Ringing Plan.
Traffic Measurement	Traffic measurements are printed through a printer port (see Printer / Terminal Support) for the following: • Console activity • System activity • DTMF, pseudo DTMF, and CLASS Receiver activity • DTRX Calls • Feature activity • Hunt Group activity • Line and Trunk activity • PCM Channel activity • Trunk activity • Trunk group activity. • IP Trunk Performance 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.
Traffic Shaping	IP trunking between SX-200 ICP controllers can be established over the internet using DSL or cable modem. Using the 6042 VPN software application, the Traffic Shaping scheme can provide the best quality voice possible over the VPN tunnel.
Transfer	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 Dial Tone	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.
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Table 6: Features supported by SX-200 ICP (continued)

Feature	Description
Transfer Security (Recall)	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.
Trunk Answer From Any Station (TAFAS)	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.
Trunk Circuit Descriptor Options	Trunk circuit descriptors specify the programmable hardware parameters of each trunk circuit in the system. Each trunk in the system must have a trunk circuit descriptor number with an associated set of selected options.
Trunk Dial Tone Detection	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.
Trunk Groups	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.
Trunk Operation - Direct Inward Dial (DID)	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 system 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.
Trunk Operation - Direct Inward System Access (DISA)	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 any features in the DISA trunk's COS which do not require a Switch Hook Flash. Optionally, the external caller can be required to enter a special account code rather than the standard DISA Access Code. See Account Codes - Verified (Special DISA), DISA trunks can be supported on many different hardware types. See Trunk Support - T1, Trunk Support - DID, Trunk Support - E & M. A trunk can be programmed as DISA at all times, or during night service only.
Trunk Operation - Non-Dial-in CO	CO trunks usually carry calls between the local central office and the PBX. 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, NIGHT1, and NIGHT2 service. They can optionally be assigned as a dedicated line on a Superset or Mitel IP telephone. The NIGHT1 or NIGHT2 service for CO trunks can be changed directly from the Attendant Console.
Trunk Operation - Tie	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 system 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.
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Table 6: Features supported by SX-200 ICP (continued)

Feature	Description
Trunk Recall	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 alternate call point: • For all trunk types, when an extension with a trunk on Consultation Hold is listening to reorder tones 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.
Trunk Support - CO (LS/GS)	The SX-200 ICP supports CO (LS/GS) trunks with the LS/GS Trunk card in digital bays.
Trunk Support - Direct Inward Dial (DID)	The DID trunk types supported are Wink Start, Delay Dial and Immediate Dial. DID Trunks support Tie, CO, DID, and DISA operation.
Trunk Support - E&M	E&M trunks are supported with the E&M Trunk module on the Universal Card in digital peripheral bays. 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.
Trunk Support - IP	IP trunks carry voice and signaling messages between networked systems. All IP trunk calls are routed through the Ethernet Switch to the WAN. The SX-200 ICP CX/CXi supports up to 16 and the MX supports up to 30 IP trunks.
Trunk Support - T1	T1 trunks are supported using T1/D4 Channel Associated Signaling (CAS), also referred to as DS1. 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 system provides a Stratum 3 or 4 clock source as an integral part of the SX-200 ICP controller. The system 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.
Uniform Call Distribution (UCD)	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.
Vacant Number Intercept	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 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.
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Table 6: Features supported by SX-200 ICP (continued)

Feature	Description
Voice mail Support	The SX-200 ICP provides embedded voice mail plus support for the following voice mail functionality: • **Centralized Voice mail:** Centralized voice mail allows one voice mail device to service several interconnected PBXs. • **Voice mail on DNIC Ports:** Voice mail devices may use the Mitel DNIC interface. • **Voice mail on ONS Ports:** This feature integrates an SX-200 ICP 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. • **Mitel Express Messenger:** Mitel Express Messenger is a voice mail card that sits in an SX-200 peripheral cabinet. Multiple cards with up to eight ports per card can be installed with each operating independently of the others. Softkey Support allows users of Mitel phones equipped with softkeys to press them instead of dialing codes to select features. • **NuPoint Messenger - Softkey Support:** NuPoint Messenger is a PC-based, voice mail and messaging system. Softkey Support allows users to press a softkey instead of dialing single-digit codes to select features. Some other softkey examples are Keep, Discard, Rewind and Fast Forward. Softkey support is only available when NuPoint Messenger has a DNIC connection. • **Single Button Transfer to Voice mail**: The Single Button Transfer to Voice mail feature provides a voice mail key that transfers a caller to a user's voice mail. The voice mail key can be programmed as a feature key, and in some cases can appear as a softkey. The voice mail key functions under two different modes: in a transfer-recall mode, or in a direct mode (no recall).
Whisper Announce	Allows a party to place a directed page to a busy Mitel telephones. 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 a Whisper Announce, if the paged party has COS Option 501, Override Announce, enabled, the paging party hears a short burst of ring-back tone and can talk immediately after the burst of ringing. After initiating a Whisper Announce, if the paged party has COS Option 501, Override Announce, disabled, the paging party hears a short burst of busy tone and must wait for the paged party to respond.
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Feature Keys to activate features

The programmable line keys on multi-line Mitel 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 are as follows:

Account Code	Campon (I Will Wait)	Forward Busy/ No Ans*	Open Door	Release
Alarm	Data Disc	Forward No Ans*	Override (Intrude)	Single Flash**
Auto Answer*	Direct Page	Group Listen	PA Paging	Speedcall
Call / Attn	Do Not Disturb	Handset Mute	Park & PA Page	Swap (Trade Calls)
Call Block	Do Not Disturb*	Headset Mode	Park & Page Group	System Park
Call Park / Remote Retrieve	Double Flash**	Intrude (Override)	Park & Page Set	VM Prompts
Call Pickup	Forward All*	Line Privacy	Phonebook	Voice mail*
Callback	Forward Always*	Music*	Privacy Release	
Callers	Forward Busy *	Night Answer	Record a Call*	

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 or LED indicators are inoperative. For the remaining feature keys listed, an indication on the adjacent LCD or LED 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 Mitel 5340 IP Phones and Superset 4025 telephones because most features are provided via softkeys.

Purchasable System Options

The software packages include all the available features. The packages are purchased using the MOSS system options. Refer to the following for more information:

- “Purchasable System Options (MOSS)” on page 147
- “DSP configuration options for MX controller” on page 151
- “DSP Configuration Options for CX/CXi Controller” on page 152
- "DSP Configuration Options for the AX Controller" on page 154

Table 7: Purchasable System Options (MOSS)

Option Number On CDE Form 4	System Options	Description
19	DID Server Application	Enables automatic DID number assignment.
83	Internet Gateway	Allows an Ethernet connection to the Internet through the WAN interface of the CXi controller.
84	Multiple Guest Suite Phones	Groups a number of telephone lines through interconnected hotel or motel rooms (suites), for the purposes of billing and shared telephone services.
85	Speak@Ease Integration	Enables you to have access to Mitel Speech Server directly (offhook) or indirectly (softkey). The Speech Server is a speech recognition application that routes incoming calls to a specific destination based on spoken commands.
86	PRI Card: Q.SIG	Allows you to connect PBXs from different vendors together to form a private network. Q.SIG currently supports incoming calls and incoming Calling Name. **Note**: Applicable to PRI Card, NSU, and IP Trunking.
87	Record a Call	Allows you to record an internal or an external two-party conversation and save the conversation in a voice mailbox.
89	CLASS functionality for ONS Sets	Allows analog telephone sets to show the calling party name and number during the ringing state and the talking state (campon).
90	ACD Real Time Event	Enables text strings to represent call events as they happen in ACD.
91	PRI Card: NFAS (Non-Facility Associated Signaling)	NFAS allows you to use a single D-channel to handle the signaling requirements for a group of PRI links that use the same Protocol. This feature eliminates the need to purchase a D-channel for each link. NFAS is mainly for North America. (PRI Card and NSU only)
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Table 7: Purchasable System Options (MOSS) (continued)

Option Number On CDE Form 4	System Options	Description
92	PRI Card: D-Channel Backup	Used for signaling to establish and maintain the circuit, and to send user data. D-channel Backup provides an alternate D-channel for calls related to NFAS. If the active D-channel fails, the system switches to the backup D-channel to support call processing. This functionality is mainly for North America. NFAS is required in order to program D-channel Backup. (PRI Card and NSU only)
93	PRI Card: Remote LAN Access	Provides LAN access to the wide area network (WAN) for both incoming and outgoing calls through LAN servers (for example, routers, bridges) using Mitel's PRI Gateway interface. (PRI Card and NSU only)
94	PRI Card: Min/Max	Controls the number of simultaneous incoming and outgoing calls. The level of control ranges from generic minimums and maximums on all calls to minimums and maximums for particular directory numbers. (PRI Card and NSU only)
95	PRI Card: Auto Min/Max	Works in conjunction with Min/Max, and increases Min/Max configurations by providing time of day programming. Time of day programming allows you to have consistent traffic control without having to frequently reprogram Min/Max. You can program call control for an entire week, and the system will automatically change Min/Max settings based on the time of day and the day of the week. Min/Max is required in order to program Auto Min/Max. (PRI Card and NSU only)
96	Number of Links (0-8)	Limits the number of T1 type links from the NSU or from a PRI card in a Peripheral cabinet, or for the onboard Dual T1/E1 Framer module(s). This option is not required for links on the T1 trunk card. The total number of links available in the system is 8 (combination of purchasable links and non-purchasable links).
98	Support 3DN, 4DN, and 400 Series Set Types	Support 3DN, 4DN, and 400 Series Set types: Enables the programming for Superset 3DN, Superset 4DN and Superset 400 series telephones.
99	Fax Tone Detection	Enables the system to recognize the FAX tone on incoming calls to the Automated Attendant. System Option 106, Automated Attendant, must be enabled as well.
102	Feature Level	Allows customers to access selected features in the release. The Feature Level option (incremented for each Feature Level) is a purchasable MOSS option, number 102 in the System Options and Timers, CDE Form 04. The feature level is obtained with all new installs and software upgrades but not with software fixes.
103	Maximum Devices (the number of user devices enabled)	Displays the maximum number of user devices that can be programmed in the system, from 24 through 768.
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Table 7: Purchasable System Options (MOSS) (continued)

Option Number On CDE Form 4	System Options	Description
104	Maximum ACD Agents (the maximum number of ACD agents enabled)	Displays the maximum number of ACD agents enabled, from 0 through 100, in increments of 5. This is the maximum number of agents that can be logged in concurrently.
105	MITEL Application Interface	Enables the MITEL Application Interface Package (MAI).
106	Automated Attendant	Enables the Automated Attendant Feature Package which supports FAX Tone Detection.
107	Lodging (Hotel/ Motel)	Enables the Hotel / Motel feature package. Lodging and Property Management System are mutually exclusive.
108	Property Management System	Enables the Property Management System (PMS). Lodging (Hotel/Motel) and Property Management System are mutually exclusive.
110	Maximum BNIC Cards	Specifies the quantity of BNIC cards that have been purchased as part of a package. No additional BNIC cards can be purchased separately. The quantity entered must exactly match the quantity on the MOSS sheet.
111	Maximum BONS Cards	Specifies the quantity of BONS or BONS CLASS cards that have been purchased as part of a package. No additional BONS cards can be purchased separately. The quantity entered must exactly match the quantity on the MOSS sheet.
112	SS 4000 Series Sets	Allows programming of Superset 4000-series telephone sets.
113	Centralized Attendant / Voice mail	Allows the system to program and access centralized attendant or voice mail facilities.
114	Maximum IP Sets	Sets the maximum number of IP phones per SX-200 ICP system. The MX controller supports a maximum of 248 IP phones and the CX/CXi a maximum of 100.
115	Maximum IP Trunks	Sets the maximum number of IP Trunks per SX-200 ICP system. The maximum number is 24. The default is 0.
120	Number of Compression Resources (0-24)	This purchasable MOSS option controls the number of G.729 codecs available to IP devices in the system. Compression enables more devices to share available bandwidth. The option is purchasable in increments of one codec to a maximum of 16 (CX/CXi) or 24 (MX) per system.
121	Voice Mail License for Bilingual Prompts	This purchasable MOSS option provides simultaneous prompts in two of the following languages: English (alphabetic prompts), English Overlaid (numeric prompts), French, and Spanish. Callers reaching the auto attendant or a subscriber’s mailbox can dial the Language Change Mailbox number to hear subsequent prompts in the alternate language.
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Table 7: Purchasable System Options (MOSS) (continued)

Option Number On CDE Form 4	System Options	Description
122	Voice Mail License for Personal Contact Numbers	If the Personal Contacts Numbers option has been purchased (listed on MOSS sheet) then mailbox owners can program alternate numbers (cell phone, pager, fax etc.) where callers can contact them instead of leaving a message. Callers reaching the owner's mailbox will hear the owner's greeting followed by prompts such as "...to reach this person's cellular phone, press C, the 2 key." Use of this feature can be enabled or disabled for the entire system.
123	Reserved for future use	
124	Voice Mail Property Management System	This purchasable MOSS option adds PMS Integration with Voice mail and voice mail-related Hospitality Features to the standard SX-200 ICP system features.
125	Licensed Embedded Voice Mail Boxes (0-748)	Determines the number of voice mailboxes available for programming. The SX-200 ICP system includes 20 mailboxes; licenses for more can be purchased. The number does not include the Administrator's Mailbox and the Operator Mailbox which are provided at no charge.
126	SMTP (Email)	Allows users' voice mail messages plus E911 and system alarm notifications to be sent to e-mail addresses specified via CDE. This purchasable MOSS option enables the SX-200 ICP to forward e-mail using IMAP and/or SMTP. IMAP allows users to manage their voice messages with one or more e-mail clients, and to synchronize the status of messages throughout the system. SMTP allows users to download their voice messages to an e-mail client. It also allows administrators to receive automatic notification of alarms and E911 calls, and to manually send system logs to an e-mail address.
128	Phonebook	Enables the Dial-by-Name (Phonebook) feature. **Note**: Phonebook and Speak@Ease cannot both be enabled.
131	PC (2nd) Port on IP Phone	Activates the second network port on IP phones which allows an attached PC to access the network.
133	TDM Bays (0-7)	This option controls the number of digital bays that boot up -- 0 -7. Note that this option applies to digital bays and NOT to PRI bays. PRI bays are controlled by System Option 96, Number of Links.
134	Recorded Announcement Device	The embedded voice mail ports can provide recorded announcement device (RAD) service, eliminating the need for external tape machines or other audio-playing devices. Enabling this option allows the RAD ports to be placed in RECORDING type hunt groups.
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DSP configuration options for MX controller

The information in the following table is provided as a guideline for determining DSP requirements. Actual requirements vary depending on the intended system usage. For more information, see the SX-200 ICP Technical documentation and Engineering Guidelines.

	Number of DSPs Installed			
Option Type	**Base Dual DSP (2 DSPs total))**	**2 Dual DSP (4 DSPs total)**	**1 Dual + 1 Quad DSP (6 DSPs total)**	**2 Quad DSPs (8 DSPs total)**
Business Option 1	3 three-party conf. 4 voice mail ports 8 G.729 channels 6 ONS/DNIC 48 IP phones 12 LS/Class trunks	8 three-party conf. 12 voice mail ports 8 G.729 channels 6 ONS/DNIC 96 IP phones 12 LS/Class trunks 24 T1 trunks	12 three-party conf. 18 voice mail ports 16 G.729 channels 96 IP phones 96 ONS/DNIC 12 LS/Class trunks 48 T1 trunks	12 three-party conf. 24 voice mail ports 24 G.729 channels 192 IP phones 192 ONS/DNIC 12 LS/Class trunks 96 T1 trunks
Business Option 2	8 three-party conf. 8 voice mail ports 0 G.729 channels 48 IP phones 12 LS/Class trunks	12 three-party conf. 18 voice mail ports 0 G.729 channels 96 IP phones 12 LS/Class trunks 24 T1 trunks	18 three-party conf. 24 voice mail ports 8 G.729 channels 96 IP phones 96 ONS/DNIC 12 LS/Class trunks 48 T1 trunks	21 three-party conf. 24 voice mail ports 16 G.729 channels 192 IP phones 288 ONS/DNIC 12 LS/Class trunks 96 T1 trunks
Hospitality Option (IP+TDM)	8 three-party conf. 8 voice mail ports 0 G.729 channels 96 ONS/DNIC 12 LS/Class trunks	8 three-party conf. 12 voice mail ports 8 G.729 channels 48 IP phones 96 ONS/DNIC 12 LS/Class trunks 48 T1 trunks	12 three-party conf. 18 voice mail ports 16 G.729 channels 96 IP phones 192 ONS/DNIC 12 LS/Class trunks 48 T1 trunks	12 three-party conf. 24 voice mail ports 16 G.729 channels 192 IP phones 384 ONS/DNIC 12 LS/Class trunks 96 T1 trunks
Analog Option 1 (MX controller)	2 three-party conf. 6 voice mail ports 0 G.729 channels 24 IP phones 288 ONS/DNIC 12 LS/Class trunks 48 T1 trunks	8 three-party conf. 18 voice mail ports 0 G.729 channels 48 IP phones 288 ONS/DNIC 12 LS/Class trunks 72 T1 trunks	12 three-party conf. 24 voice mail ports 0 G.729 channels 96 IP phones 288 ONS/DNIC 12 LS/Class trunks 72 T1 trunks	21 three-party conf. 24 voice mail ports 0 G.729 channels 192 IP phones 384 ONS/DNIC 12 LS/Class trunks 96 T1 trunks
Analog Option 2 (MX controller)	2 three-party conf. 4 voice mail ports 0 G.729 channels 24 IP phones 384 ONS/DNIC 12 LS/Class trunks 48 T1 trunks	10 three-party conf. 12 voice mail ports 0 G.729 channels 48 IP phones 384 ONS/DNIC 12 LS/Class trunks 48 T1 trunks	12 three-party conf. 16 voice mail ports 0 G.729 channels 48 IP phones 480 ONS/DNIC 12 LS/Class trunks 48 T1 trunks	21 three-party conf. 24 voice mail ports 0 G.729 channels 96 IP phones 480 ONS/DNIC 12 LS/Class trunks 48 T1 trunks
				Page 1 of 2

	Number of DSPs Installed (continued)			
Option Type	**Base Dual DSP (2 DSPs total))**	**2 Dual DSP (4 DSPs total)**	**1 Dual + 1 Quad DSP (6 DSPs total)**	**2 Quad DSPs (8 DSPs total)**
Analog Option 3 (Requires Quad DSP) (MX controller)		8 three-party conf. 12 voice mail ports 0 G.729 channels 96 IP phones 576 ONS/DNIC ports 12 LS/Class trunks 96 T1 trunks		
				Page 2 of 2

Note: The base MX contains an Analog Main Board (AMB) that supports 2 ONS, 2 DNIC and 6 LS circuits. By installing an Analog Option Board (AOB), capacity increases to 4 ONS, 2 DNIC and 12 LS circuits. Additional circuits are available by connecting peripheral cabinets to the MX. The AOB does not require extra DSP resources.

DSP Configuration Options for CX/CXi Controller

The information in the following table is provided as a guideline for determining DSP requirements. Actual requirements vary depending on the intended system usage. For more information, see the SX-200 ICP Technical Documentation and Engineering Guidelines.

DSP Configuration	Without Compression	With Compression
Base System (2 total)	3 three-party conf. 4 voice mail ports 0 G.729 channels 24 IP phones 8 ONS phones 12 LS/CLASS trunks 0 T1 ccts	
Base + T1/E1 Combo (3 total)	10 three-party conf. 16 voice mail ports 0 G.729 channels 64 IP phones 24 ONS phones 12 LS/CLASS trunks 24 T1 or 23 PRI ccts	
Base + Dual DSP (4 total)	10 three-party conf. 16 voice mail ports 0 G.729 channels 40 IP phones 40 ONS phones 16 LS/CLASS trunks 0 T1 ccts	3 three-party conf. 4 voice mail ports 8 G.729 channels 40 IP phones 8 ONS phones 12 LS/CLASS trunks 0 T1 ccts
		Page 1 of 2

DSP Configuration	Without Compression	With Compression
Base + T1/E1 Combo + Dual DSP (5 total)	10 three-party conf. 16 voice mail ports 0 G.729 channels 80 IP phones 150 ONS phones 12 LS/CLASS trunks 24 T1 or 23 PRI ccts	10 three-party conf. 16 voice mail ports 8 G.729 channels 80 IP phones 100 ONS phones 12 LS/CLASS trunks 24 T1 or 23 PRI ccts
Base + Quad DSP (6 total)	10 three-party conf. 16 voice mail ports 0 G.729 channels 80 IP phones 150 ONS phones 36 LS/CLASS trunks 0 T1	10 three-party conf. 16 voice mail ports 8 G.729 channels 80 IP phones 56 ONS phones 36 LS/CLASS trunks 0 T1
Base + T1/E1 Combo + Quad DSP (7 total)	10 three-party conf. 16 voice mail ports 0 G.729 channels 100 IP phones 150 ONS phones 16 LS/CLASS trunks 24 T1 or 23 PRI ccts	10 three-party conf. 16 voice mail ports 16 G.729 channels 100 IP phones 130 ONS phones 16 LS/CLASS trunks 24 T1 or 23 PRI ccts
		Page 2 of 2

Note: The base CX/CXi Controller contains an Analog Main Board (AMB) that supports 4 ONS and 6 LS circuits. By installing an Analog Option Board (AOB), capacity doubles to 8 ONS and 12 LS circuits. The AOB does not require extra DSP resources.

Note: Not all maximum values for lines and trunks can be realized simultaneously.

DSP Configuration Options for the AX Controller

The information in the following table is provided as a guideline for determining DSP requirements. Actual requirements vary depending on the intended system usage. For more information, see the SX-200 ICP Technical documentation and Engineering Guidelines.

Business Option 1	
Configuration	**Feature/Resource**
Base	10 three-party conf. 20 voice mail ports 0 G.729 channels 200 IP phones 192 ONS phones 48 LS/CLASS trunks 0 T1 ccts
Base + Dual T1/E1	10 three-party conf. 20 voice mail ports 0 G.729 channels 248 IP phones 288 ONS phones 48 LS/CLASS trunks 48 T1 or 46 PRI ccts
Base + T1/E1 Combo	10 three-party conf. 20 voice mail ports 0 G.729 channels 248 IP phones 288 ONS phones 24 LS/CLASS trunks 24 T1 or 23 PRI T1 ccts
Base + Dual T1/E1 + Quad DSP	10 three-party conf. 20 voice mail ports 24 G.729 channels 248 IP phones 288 ONS phones 48 LS/CLASS trunks 48 T1 or 46 PRI ccts

Note: G.729 Compression requires that additional DSP resources must be installed, either a T1/E1 Combo card or a Dual or Quad DSP card.

System Parameters

This section contains specifications for the following:

- "Environmental Requirements" on page 155
- "Shipping and Storage" on page 156
- "Electrical Requirements" on page 156
- "Feature Capacities" on page 157
- "System Parameters" on page 159
- "Tone Plan Support" on page 160
- "Traffic Parameters" on page 160
- "System reliability and availability standards" on page 163
- "Physical Characteristics of SX-200 ICP" on page 166
- "Power and Grounding" on page 166
- "Regulatory Compliance" on page 167

Environmental Requirements

The SX-200 ICP system should be located in an area that is dry, clean, well ventilated, well lit, and readily accessible.

	Controller	ASU	NSU	Peripheral Node
Temperature	41° to 122°F (5° to 50°C)	41° to 122°F (5° to 50°C)	41° to 122°F (5° to 50°C)	32° to 122°F (0° to 50°C)
Humidity	40-90% Relative Humidity, non condensing	34-95% Relative Humidity, non condensing	34-95% Relative Humidity, non condensing	5-95% Relative Humidity, non condensing
Max Heat Dissipation (fully loaded	750 BTUs per hour	170 BTUs per hour	170 BTUs per hour	724 BTUs per hour
Air Flow	46 cubic ft/min at maximum output of fans			150 cubic ft/min at maximum output of fans
Acoustic Emissions	Maximum 50dBA continuous, 75 dB intermittent (<10% duty cycle)			Maximum 50dBA continuous, 75 dB intermittent (<10% duty cycle)

Shipping and Storage

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 the following table.

Condition	Specification
Temperature	-40× to 120× F (4× to 49× C) for all components
Humidity	SX-200 ICP - 34-85% relative humidity non-condensing All other components - 34-85% relative humidity non-condensing

Electrical Requirements

Table 8: Power Supply

	SX-200 ICP - CX/CXi	SX-200 ICP - MX	ASU	NSU	Peripheral Node
Input / disconnect	IEC 320 - C14 Class 1 AC Receptacle	IEC 320 - C14 Class 1 AC Receptacle	Connector is a Standard Male IEC-320 AC input	IEC 320 - C14 Class 1 AC Receptacle	IEC 320 - C14 Class 1 AC Receptacle
Operation	100-120/200-240 V ac auto selectable	100-120/200-240 V ac auto selectable	Universal input design, operating input voltages from 90-132/180-264 Vac	100-120/200-240 V ac auto selectable	For North America: 100-120/200-240 V ac auto selectable. For Europe: 200-240 Vac
Maximum input power	300 W	100 W	60 W full rated load	60 W	300 W
AC source	90 - 132 Vac; 47 - 63Hz in North America	90 - 132 Vac; 47 - 63Hz in North America	AC input frequencies from 47Hz to 63Hz	90 - 132 Vac; 47 - 63Hz in North America	47 - 63Hz

Feature Capacities

SX-200 ICP systems offer a wide range of features through software packages. These are outlined in the Features Description section of this guide. A full description appears in the E-docs under Program Features. Maximum capacities that apply to system features are listed in the following table. See also Voice Mail Capacities for information about the capacities of the embedded voice mail application.

Table 9: Feature Capacities

Feature	SX-200 ICP (CX/CXi)	SX-200 ICP (MX)	SX-200 ICP (AX)
Maximum number of simultaneous calls	90	248	248
Maximum number of Call Park keys	24		
Maximum number of Mailbox keys	748		
Max number of speech paths or channels used by any call	2		
Maximum number of simultaneous consultations	5		
Maximum number of System Park Orbits	25		
Maximum number of Specific Park Orbits	25		
Maximum number of Embedded VoiceMail Ports	16	24	20
Maximum number of Embedded VoiceMail Mail Boxes	748		
Maximum number of simultaneous add-on (3-way) calls	DSP configuration dependent		
Maximum number of simultaneous station-controlled conference calls	DSP configuration dependent		
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	247		
Maximum number of simultaneous callbacks that can be enabled	100		
Maximum number of simultaneous “Dial 0” calls	48		
Maximum number of ONS telephones ringing simultaneously per bay	8	32	96
Maximum number of messages queued in the system	750		
Maximum number of hunt groups	99		
Maximum number of ring groups	25		
Maximum number of hunt groups in ACD	99		
Maximum number of ACD agents that may be defined	50	999	40
Maximum number of active agents in ACD per bay	25		
Maximum number of calls that can be simultaneously connected to Music-on-Hold	unlimited		
Page 1 of 3			

Table 9: Feature Capacities (continued)

Feature	SX-200 ICP (CX/CXi)	SX-200 ICP (MX)	SX-200 ICP (AX)
Maximum 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	36	200	36
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 attendant consoles on a Digital Line Card	4		
Maximum number of calls that can be simultaneously held by one attendant	8		
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 stations of Superset 4001, Superset 4015, Superset 4025, Superset 4125, Superset 4150, Superset 401+, Superset 410, Superset 420, Superset 430, Superset 3DN and 4DN telephones, DSS /BLF Interface Units, and ONS ports.	650		
Maximum number of user devices (all sets, stations, trunks, consoles, stand alone datasets, and DMP units)	150	768	536
Maximum number of IP devices and other resources per system - IP trunks - IP phones	 16 100	 30 248	 30 248
			Page 2 of 3

Table 9: Feature Capacities (continued)

Feature	SX-200 ICP (CX/CXi)	SX-200 ICP (MX)	SX-200 ICP (AX)
Compression channels	16	24	32
Maximum number of music sources	9	25	9
Maximum number of Door relays	4	3	Not supported
Maximum number of ASU	Not supported	6	Not supported
Maximum number of ASU II	3	6	Not supported
Maximum number of lines: SX-200 Peripheral cabinet	Not supported	96	Not supported
Maximum number of TDM bays	Not supported	7	Not supported
Maximum number of ISDN bays	Not supported	4	Not supported
Maximum number of T1 links, including T1 D4 links, PRI links, and NSU links	2	8	2
Maximum Number of T1 links per system	2	8	2
Maximum Number of Page Groups	50		
Maximum Number of Paging Zones	9		
Maximum Number of Stations in a Page Group	32	64	32
Maximum Number of Sub-attendants	25		
Maximum Number of LDN Appearances	16	16	16
Maximum Number of Line Appearances	64		
			Page 3 of 3

System Parameters

The sale and installation of any communications equipment is subject to various local and national regulations covering a number of parameters including electrical characteristics, tone plans, and loss and level plans. In addition, the traffic capacity of the system must be considered - will the system handle the expected traffic at cut over and can it be expanded to cover expected growth in the future?

This section of the guide lists many of these system parameters that can be used for preliminary planning. For additional SX-200 ICP planning information, refer to the 200 ICP Engineering Guidelines.

Tone Plan Support

The SX-200 ICP system supports tone plans for North America, Asia, and Latin America only.

Traffic Parameters

Traffic engineering is a statistical method used to ensure that you have provisioned your system to give the level of service to which your users are accustomed. Understanding these traffic engineering concepts is important when purchasing or configuring your system.

Early analog PBXs usually had a large number of voice ports contending for a low number of speech paths. Once all the speech paths were in use, anyone trying to place a new call was forced to wait for the next free path. To reduce the chance of being unable to complete a call, customers with high traffic could give users enough speech paths by installing large PBXs with more ports and thus more speech paths.

The introduction of digital systems replaced speech paths with call connections and channels. Using time division multiplexing, a single piece of wire could now carry up to 32 simultaneous conversations. The result was a system that was physically smaller but able to carry many more calls. These systems were often referred to as non-blocking, implying that all users of the system could be placing calls at the same time, and a lack of system resources would not prevent any of these calls from being completed.

To improve system efficiency, PBXs are normally engineered so callers are competing for limited system resources such as trunks. This contention allows the system to make better use of trunks by scheduling callers on each trunk. Most users are unaffected by this as a higher percentage of traffic in a system is traditionally with inside parties. Key Systems force users to manually select outgoing lines as most of their traffic is external calls.

Another factor which must be considered is traffic peaks. Although most system analysis is done using average traffic, maximum peaks must also be identified. If traffic in any period exceeds these specified maximums, system performance will likely degrade, and over-competition for resources may result. Once traffic drops below this peak, the system will provide normal performance. When purchasing a system, ratings for system peak capacities should be determined for your configuration rather than using the average figures for the product line.

Here are some facts about the SX-200 ICP system relating to traffic.

- **a.** The systems provide for at least 200 simultaneous call connections. This means that 200 stations can talk to 200 other stations or trunks before call connections could create blocking.
- **b.** In peripheral digital bays the concentration of ports to channels is 96:90. This means that if only 90 devices are installed in a digital bay there is no possibility of blocking on channels.
- **c.** To withstand peaks in traffic, Mitel rates its switches according to line size using the very heavy traffic patterns stipulated in ATT0048.

Mitel also tests the systems to ensure that they can withstand twice the traffic specified for the line-size of the switch. This is to ensure that peak traffic will not impact system performance under normal conditions.

The SX-200 ICP system also contains a traffic measurement package to help monitor actual traffic patterns. This traffic information must be considered when additional lines and trunks are added to an existing system. For example, information on dial tone delays may indicate a need for additional receiver modules. Console pegs can indicate the need for additional console positions. Trunk usage reports can indicate the need for additional trunks.

Use the traffic report figures as guidelines. Specific departments or trunks may not follow the averages of the rest of the system. This should be understood and analyzed to ensure that your system can meet the needs of all users.

To aid in configuring your system, the following chart can help you in determining the number of trunks required, as well as using the System Engineering Tool. For this chart to be effective certain assumptions have been made. If your system does not fit with these assumptions, then consult your dealer or sales representative.

Assumptions:

a. Traffic patterns are approximately
- 33% internal
- 33% outgoing
- 33% incoming.

b. Trunks are both-way, as these are most efficient for carrying traffic.

c. An adequate number of receivers are present. Rule of thumb is 4 receivers per 60 telephones.

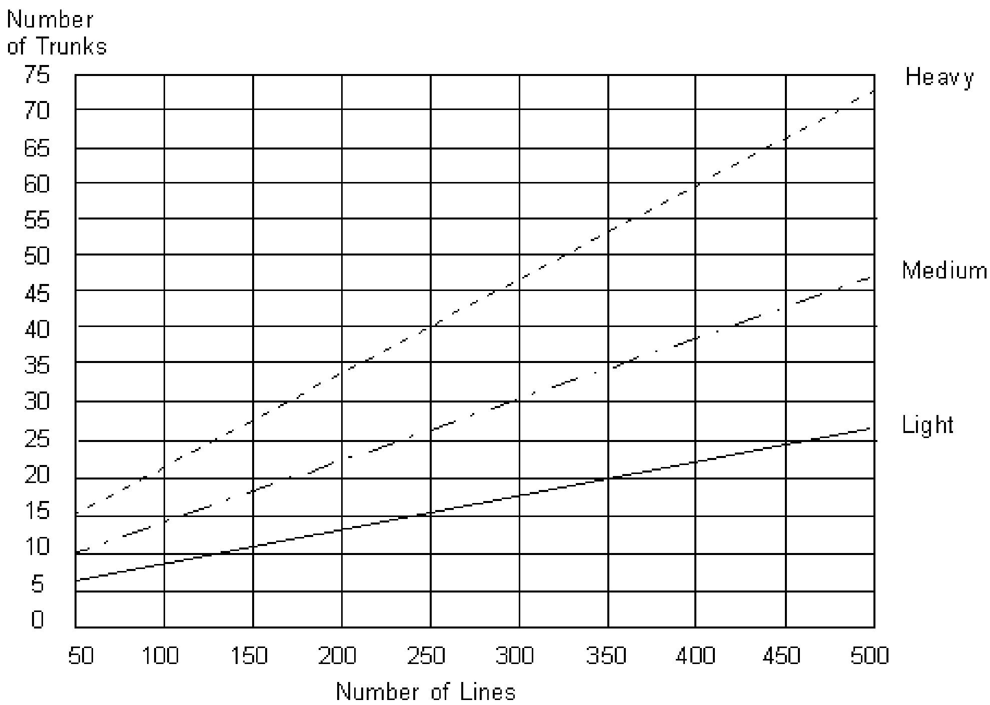
d. Target grade of service is P.01 or the same as the level which most telephone companies provide.

To use the chart you must determine your average traffic levels for the busiest hour of the day. Divide the number of calls for this hour by the number of telephones on your system.

- 1 call per hour = light traffic
- 2 calls per hour = medium traffic
- 3 or more calls = heavy traffic

On the bottom axis of the chart select the number of telephones in your installation. Choose the line on the graph that represents your traffic level. The left axis indicates the recommended number of trunks.

Special applications (such as ACD) should be highlighted to your dealer or sales representative.



Grade of Service

The SX-200 ICP Grade Of Service (GOS) in terms of blocking is outlined below.

Table 10: SX-200 ICP 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

Traffic Limitations

Traffic capacities are specified on a per line basis in terms of calls per hour and Erlangs.

The tabulated voice traffic call rates defined in the tables below are based on actual laboratory traffic tests with Superset 430 telephone sets originating and terminating calls. Set type, trunk/ARS programming, and maintenance reporting options all affect overall system performance. Therefore, system performance will vary depending on customer selected options.

Table 11: Voice Reference Call Rates (per hour)

Configuration	Rated	Peak
96 ports and 19 trunks	730	1222
Two Peripheral Bays with 180 ports and 29 trunks	1422	2087

Table 12: Maximum Calls Per Hour and Performance Index

Call Type	Relative P.I.
ONS	1
IP to IP	3
IP to T1/D4	9
ONS to IP	8
IP to ONS	6

Detailed calculations on traffic, performance and physical capacity of a system can be made using the Mitel System Engineering Tool. For more information, contact a sales engineer or dealer.

System reliability and availability standards

An Enterprise voice solution demands stringent reliability and availability standards. Mitel SX-200 ICP provides 99.999% reliability. The SX-200 ICP is built around a secure, real-time Unix-like operating system that is not vulnerable to MS Windows OS virus attacks.This is important because, although Enterprises might be able to carry on business when E-mail servers are temporarily put out of service by virus attacks, voice communications require ’five nines’ (99.999%) reliability.

The standard used by Mitel, to calculate Mean Time Between Failure (MTBF) is the Bellcore Standard TR-NTW-000332 "Reliability Prediction Procedure for Electronic Equipment". This standard combines predicted failure rates (based on US Military standard MIL-STD-217-F) with actual failure rates (based on field returns). This combined failure calculation provides a more accurate prediction of the serviceability of a product.

Defining terms

The term five-nines refers to availability more than reliability, although reliability is integral to availability. Availability is determined by two basic factors; Mean time Between Failures (MTBF), sometimes referred to as Mean Time Between Outages (MTBO), and Mean Time to Repair (MTTR). Both of these are commonly measured in hours. Availability is described by the following equation:

Availability = MTBF/(MTBF + MTTR) = .9xxxxx

where:

Mean Time Between Failure (MTBF): term used to estimate the reliability of a product's hardware.

Mean Time To Repair (MTTR): the mean time to restore service rather than to repair a component. MTTR includes the following five activities:

- Failure Detection
- Failure Notification
- Vendor/User Response
- Repair/Replacement
- Recovery/Restart/Reboot

Note: The first four items in this list can be significantly reduced or eliminated by redundant components and automatic reconfiguration.

MTBF provides a measure of a system's reliability. However, over the system's lifetime, this metric does not necessarily identify everything you need to know. Your system could have 99% availability and still suffer a disaster (one huge outage or many short outages) and still produce the same availability. Although these metrics do not take the impact of outages into consideration, they still provide a frame of reference.

The metrics for five-nines includes performance for the following system elements and components:

- Hardware components (CPUs, NSUs, ASUs)
- Power supplies
- Any other hardware component that can cause a total failure

The calculation does not include the following items:

- Shutdown of the operating system software
- Loss of electrical power
- Network loss
- Time required for application software upgrades and fixes
- Time required for preventative maintenance

- Shutdown of some call servers when line cards, trunk cards or gateways are installed
- Complete server shutdown to install operating system changes or new releases

Reliability calculations

Although some industry standards exist regarding the calculation of MTBF, few manufacturers will state which specification is used to determine their MTBF figures. To prevent any misconceptions with respect to the MTBF figures referred to here, this section considers both the MTBF figure and the items involved in its calculation.

Note: The following are estimations based on detailed engineering and statistical research and are made available by Mitel's Corporate Engineering Department. This information constitutes average reliability and may differ from actual reliability due to particular usage, environment factors and other conditions. These estimates assume proper maintenance by factory-trained technicians using genuine new spare parts. MTBF figures do not take into account terrorism, neglect, misuse, malicious damage or Acts of God.

Reliability is measured using the following values:

- Hardware: Mean time between failures (MTBF)

System failure is measured using the following values:

- Hardware: Critical resource failure (MTBF: 15 years)
- Software: Mean time to failure (MTTF: < one unplanned system reboot per 5 years)

Down time is measured using the following values:

- Hardware: Mean time to repair (MTTR: 2 hours)
- Software: Software recovery time (SRT: 25 minutes)

Availability is measured using the following values:

- Hardware: MTBF/(MTBF + MTTR) = 99.999%

Software: MTTF/ (MTTF + SRT) = 99.999%

Physical Characteristics of SX-200 ICP

	SX-200 ICP MX Controller	SX-200 ICP CX/CXi Controller	SX-200 ICP AX Controller
Height	2.7 in. (7 cm)	3.5 in. (8.9 cm) (2 U)	12.25 in. (31.1 cm) (7 U)
Width	17.3 in. (44 cm) (19" rack- mountable)	17.75 in. (45.1 cm) (19" rack mountable)	17.76 in. (45.1 cm) (19" rack mountable)
Depth	19.6 in. (50 cm)	16.5 in. (41.9 cm)	14.6 in. (37.1 cm)
Weight	14 lb (6.39 kg)	19.8 lb (8.98 kg)	39.69 lb (18.0 kg)
Network Services Unit			
Height	1.75 in. (4.454 cm)		
Width	17.75 in. 45.1 cm (19" rack- mountable)		
Depth	15.5 in. (39.4 cm)		
Weight	8.41 lb (4.27 kg)		

Power and Grounding

The system requires a single-phase, 115 Vac, 15 A circuit. A separate ground wire (size 6 AWG) must be installed between the equipment cabinets and the building ground. The ground wire should be a separate ground wire.

Grounding conductor

The grounding conductor must be an insulated grounding conductor, sized according to the National Electrical Code (NEC) in the United States (NFPA/ANSI 70 Section 250-95, Exception No. 1, and Section 240-4, Exception No. 1).

The grounding conductor is provided as part of the three-wire, 15-Amp, AC-power cord set included with the equipment. If the power cord must be replaced, use a power cord of the same gauge that has the same insulation, number of conductors, and usage ratings. The grounding conductor must be:

- Not smaller in size than, and equivalent in insulation material and thickness to, the grounded and ungrounded branch circuit supply conductor
- An insulated green wire with yellow stripes
- Part of the circuit that supplies that product or system
- Connected to ground at the service equipment.

Protective grounding conductor

The protective grounding conductor must comply with the general rules for grounding contained in Article 250 of the National Electrical Code, NFPA 70, or Section 10 of the Canadian Electrical Code, CSA C22.1. The protective grounding conductor must not depend on the power cord and plug of the product. The protective grounding conductor must be:

- An insulated wire, #6 (13mm2) to #14 (2mm2) AWG, with green and yellow stripes
- Connected to the grounding stud on the back of the cabinet.

In addition to proper grounding, an AC surge suppressor is recommended between the SX-200 ICP and the 115 Vac outlet for each cabinet. This will protect the equipment from power surges.

All trunks and off-premises extensions should be protected against lightning by gas tubes.

Regulatory Compliance

The SX-200 ICP system meets the following regulatory requirements:

EMC - United States:FCC part 15 subpart B - Class "A"
EMC - Canada:IC ICES-003 - Class "A"
Safety - United States:ANSI/UL1459
Safety - Canada:CAN/CSA-C22 No. 225
Network - United States:FCC 47 CFR part 68
Network - Canada:IC CS-03

The FCC Registration Numbers for the SX-200 ICP equipment are: BN2KF10BKTS and BN2MF10BKTS.

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