MITEL

Communications Director

General Information Guide
Release 4.1

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About this Document

Overview

This guide provides an overview of the Mitel® Communications Director (MCD) call-processing software and its host hardware platforms, the Mitel® 3300 IP Communications Platform (ICP), Industry Standard Servers (ISS) from HP and IBM, and Sun Microsystems® servers (as a component of the Mitel Communications Suite). The topics covered in this guide include

- description of the system architecture and components
- migration strategies
- supported applications.

Audience

This guide is for

- end customers
- sales executives
- consultants
- industry analysts
- media analysts
- sales engineers
- system engineers.

Related Documentation

Go to the Mitel Customer Documentation web site at http://edocs.mitel.com to access Mitel documentation. You require a Mitel Online account username and password to view and download technical documentation. However, you do not need a username and password to view and download end user documents (such as telephone user guides).

The following guides provide complete information about MCD and the 3300 ICP:

- *General Information Guide* provides an overview of the system.
- *Site Planning Guide* provides site planning and site preparation guidelines.
- *Technician's Handbook* provides installation, upgrade and maintenance instructions.
- *Hardware Technical Reference Manual* provides hardware specifications.
- *System Administration Tool Online Help* provides programming, maintenance, and trouble-shooting procedures.

- *Troubleshooting Guide* provides information on diagnosing and resolving common problems with the 3300 ICP.
- *Resiliency Guidelines* provides a comprehensive overview of the Mitel Resiliency solution and offer customers the tools to understand, plan, and implement a resilient network.
- *Engineering Guidelines* provides the information that is required to engineer a 3300 ICP system for a customer site. The guidelines are intended to highlight specific areas of the product that need to be considered before installation.

For information on Mitel Communications Suite, refer to the following documents:

- *Mitel Communications Director Installation and Administration Guide for Mitel Communications Suite (MCS)* describes the installation, administration, maintenance, and troubleshooting of the MCD software blade.
- *Mitel Communications Director for MCS System Administrator Online Help* provides administration and programming procedures for the MCD software blade.
- *Mitel Communications Suite Engineering Guidelines* provides information on implementing MCS. Topics covered include licensing, system requirements, capacities, performance, and supported configurations.

Overview

MCD provides businesses of all sizes with a highly scalable, feature-rich communications system. It's designed to meet the needs of businesses from 5 to 65,000 users whether they are single-site deployments or multi-site networks that span many countries.

Platforms

MCD runs on the following hardware platforms:

- Mitel 3300 ICP controllers, including MXe (Standard and Expanded), MXe Server, AX, CX II and CXi II
- Industry Standard Servers (ISS)
- Sun Microsystems® servers (as a component of Mitel® Communications Suite)

Modular Platform Design Provides Scalability and Flexibility

Whether deployed on the 3300 ICP, an ISS or a Sun server as part of MCS, MCD has a modular and scalable system design that allows customers to invest in a hardware platform to meet their current requirements and then increase the size of the system as their business expands. The core call control features are the same regardless of the hardware platform, and functionality, such as trunk support, can be provided through field-installed modules for some platforms. This hardware commonality ensures that as a business grows the majority of a customer's investment is protected when a controller chassis is upgraded.

About the Mitel Communications Director (MCD)

The MCD can be deployed to support a broad spectrum of site configurations. For example, a highly centralized solution can be implemented at the head office with the call control and IP telephony services delivered over wide-area-network connections to small branch offices. Larger branch offices can be configured with a main controller on site to provide local support. Finally, an entire network of controllers can be clustered to function as one large system.

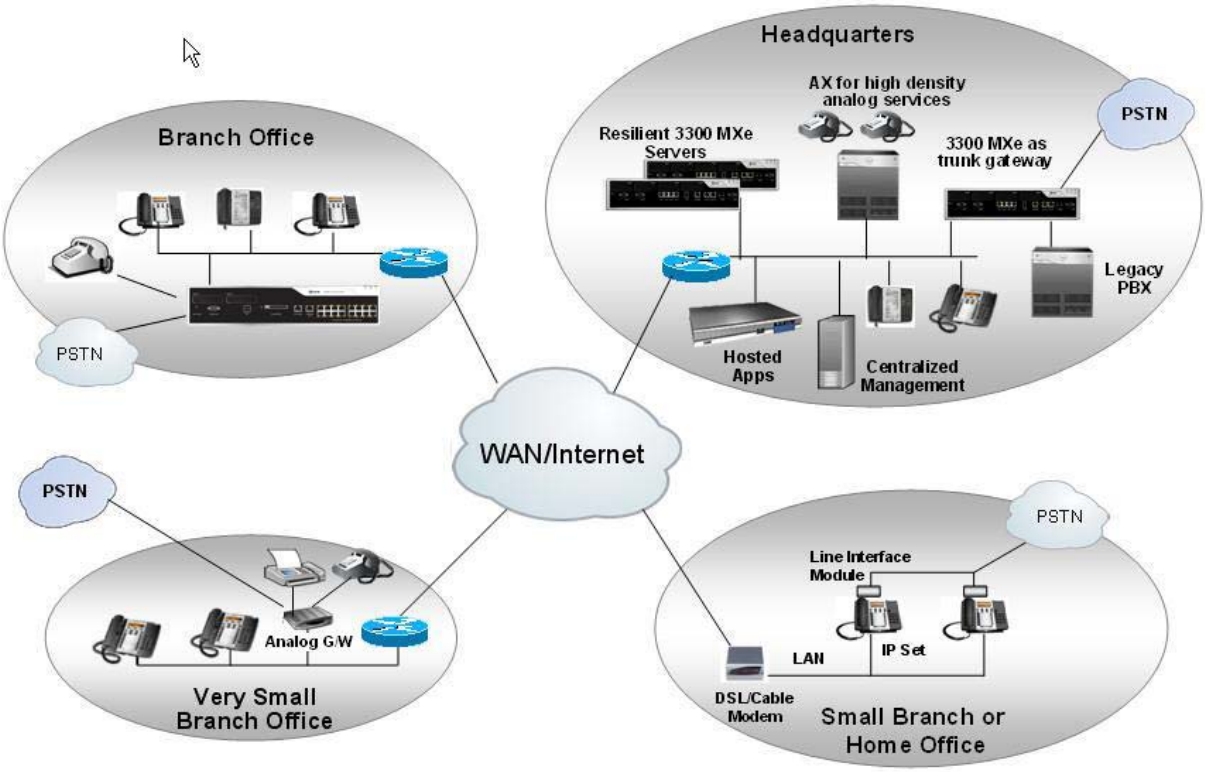


Figure 1: MCD Site Configuration

For smaller organizations, Mitel delivers a 3300 ICP system which incorporates a powered Ethernet switch and an embedded processing card that hosts applications and services such as the Mitel Border Gateway (MBG), Unified Communicator Mobile and Live Business Gateway, essentially delivering all the voice communications capabilities a small business needs in a single chassis. For example, a CX Controller can support a 40/150-person business with embedded voicemail and auto attendant, as well as house a connection to the public network and host applications like UC Mobile – all from a single chassis.

For large organizations or multi-site deployments, up to 999 controllers can be deployed in a cluster to deliver extensive features, services, and applications. These controllers use a peer-to-peer communication protocol to ensure management and administration data is shared between systems thus ensuring consistency of features and applications for the users without incurring high management costs.

On large sites, key functionality is typically hosted by dedicated “task-specific” controllers with all users connected over an IP network; for example a large organization might have

- 5000 users on an MXe Server or ISS (primary) with a backup MXe II Controller to provide resilient support
- an MXe II Controller acting as a trunking gateway with connections to the traditional telephone network for outside access, and
- a dedicated NuPoint Unified Messaging Server for voicemail, automated attendant, and unified messaging.

Networking With Industry-Standard Protocols

Use of open standards, such as SIP, interconnects next generation network services and applications to support new desktop devices. Customer can be assured that their investment in a Mitel solution will be developed and expanded into the future because the solution is not limited by proprietary protocols. And while our focus is to deliver a complete communications solution that meets the needs of today with potential to deliver more in the future, our solution also supports an extensive list of legacy protocols and devices.

Reliability Through Redundancy and Resiliency

For mission-critical environments, you can provision some models of the 3300 ICP with hardware redundancy and resiliency. Resiliency automatically transfers support for an IP phone to an alternate controller in the event that the phone cannot communicate with its primary controller. By taking advantage of IP-networking, resiliency provides an extremely flexible solution to enhance system reliability. It uses resources that are spread across the network to optimize hardware resources and ensure that there is no single point-of-failure.

Mitel's resiliency solution uses Spanning Tree Protocol and Rapid Spanning Tree Protocol (STP/RSTP). These protocols allow physical path redundancy between Ethernet switches. They place redundant network paths into standby mode by blocking traffic on redundant ports. Then, if a currently active network path fails due to a Bridge/Switch failure or a network cabling failure, STP/RSTP enables a standby network path and network connectivity is restored.

Hardware reliability is enhanced by the use of solid state hard drives in some models of the 3300 ICP.

About Mitel Communications Suite for Sun Servers

Mitel Communications Suite allows Mitel's Mitel Communications Director software to reside on Sun Microsystems® servers along with other Mitel voice applications. This solution preserves the functionality and features of the traditional MCD as well as the other embedded applications and allows organizations to carry voice-based solutions on a Sun data infrastructure. It is the perfect solution for all businesses looking to reduce install times; to get secure, centralized management; and to lower their total cost of ownership.

Aimed at reducing the number of deployed servers as well as the complexity of the installation, Mitel Communications Suite integrates:

- Call control: Mitel Communications Director
- Voicemail: Mitel NuPoint™ Messenger
- Microsoft® integration: Mitel Live Business Gateway
- Thin client integration: Mitel Unified IP Client for Sun Ray™

Applications that Enhance Productivity

MCD boasts an extensive number of applications that provide significant value to an organization and its employees. Applications are available to enhance communication, productivity, accessibility, mobility, as well to support the specialized site requirements of businesses and institutions, such as hotels, hospitals, schools, military sites, and call centers.

Mitel Networks also supports the integration of third-party applications through the Mitel Solutions Alliance (MSA). If your business requires custom applications or features to achieve higher productivity, this program enables you to develop them.

Devices that Support the User

Mitel has a wide selection of attractive, easy-to-use IP devices to meet the needs of employees, managers, executives, and attendants. These IP devices provide quick access to the system's powerful features through programmable feature keys, softkeys, and menu-guided applications such as Call Forwarding and Call History. Mitel provides:

- Basic IP Phones
- Display Phones
- Desktop Application Phones
- Wireless Phones
- Session Initiation Protocol (SIP) Phones
- Consoles
- Conference Units
- Video Conferencing Devices
- Digital Phones
- Phone Accessories.

Powerful Tools Minimize Configuration and Support

Installation, configuration, and administration are minimized and simplified through a series of powerful tools. These tools are designed to ensure that the target audience — end-user, group administrator, system administrator, or installer can perform their functions quickly and easily.

- End user tools allow users to maximize the value of the system features.
- Administrator tools simplify system and user configuration.
- Management tools automate the tasks required to support large scale installations.
- Maintenance tools reduce the time and costs associated with system support.

Extensive System Feature Set

MCD has an extensive list of end-user and system features that support effective and efficient communications. The system administrator can enable or disable features through the System Administration Tool and can create Classes of Service to define levels of feature support for each different group of users. For example, a Class of Service can be created to provide executives with advanced calling privileges, such as Executive Busy Override.

Administrators can also enable or disable system settings across the entire system or network. Although the features and system settings are configurable to allow maximum flexibility, usable default settings minimize configuration requirements.

Migration Made Easy

Because of Mitel's long history in voice communications, we continue to support a host of protocols which facilitate a smooth migration to Voice over IP support – whether your legacy PBX is from Mitel or another supplier.

You can deploy the 3300 ICP as a network gateway to link multiple traditional PBX's together over a WAN connection, eliminating costly private circuits. Or, you can deploy it as an applications gateway that delivers critical functionality to a defined user community without disrupting the broader organization. These deployment models allow organizations to migrate at their own pace, when it suits their needs.

3300 ICP System Description

This chapter describes the 3300 ICP system functionality. For details on system configurations, refer to the Engineering Guidelines. For detailed descriptions of the hardware components, refer to the Hardware Technical Reference Manual.

3300 ICP System Architecture

The 3300 ICP delivers 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. It also switches calls between traditional telephone devices:

- For IP telephony, it provides call setup, tear down, and signaling between Ethernet IP connected telephones.
- For traditional telephony, such as POTS and PSTN trunks, it handles calls 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

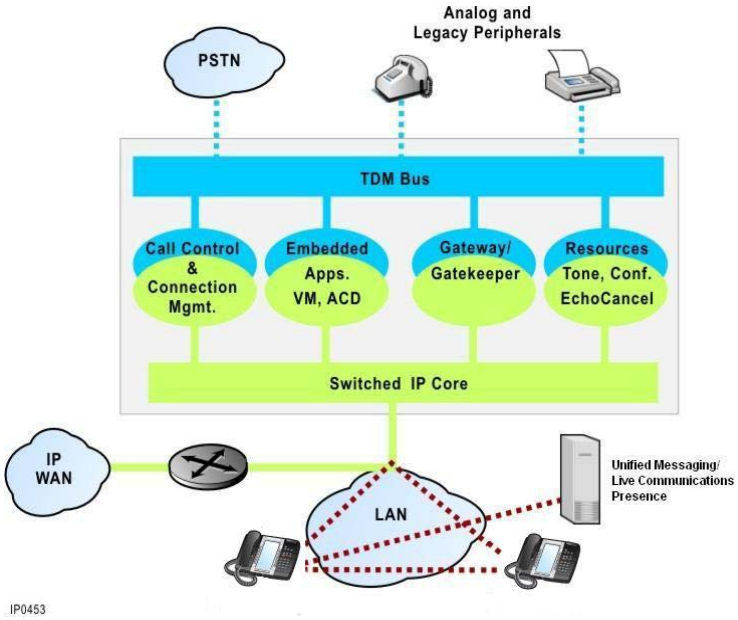


Figure 2: 3300 ICP System Architecture

Mitel Communications Director Software

The Mitel Communications Director software provides the call control features and applications that enhance business communications. The software is packaged for a variety of site capacities – but ultimately all sites, regardless of size, have access to the same extensive feature set.

Controllers

The 3300 ICP Controllers provide the voice, signaling, central processing, and communications resources for the system. Mitel offers a selection of five controllers and a server to meet the overall solution requirements of any site.

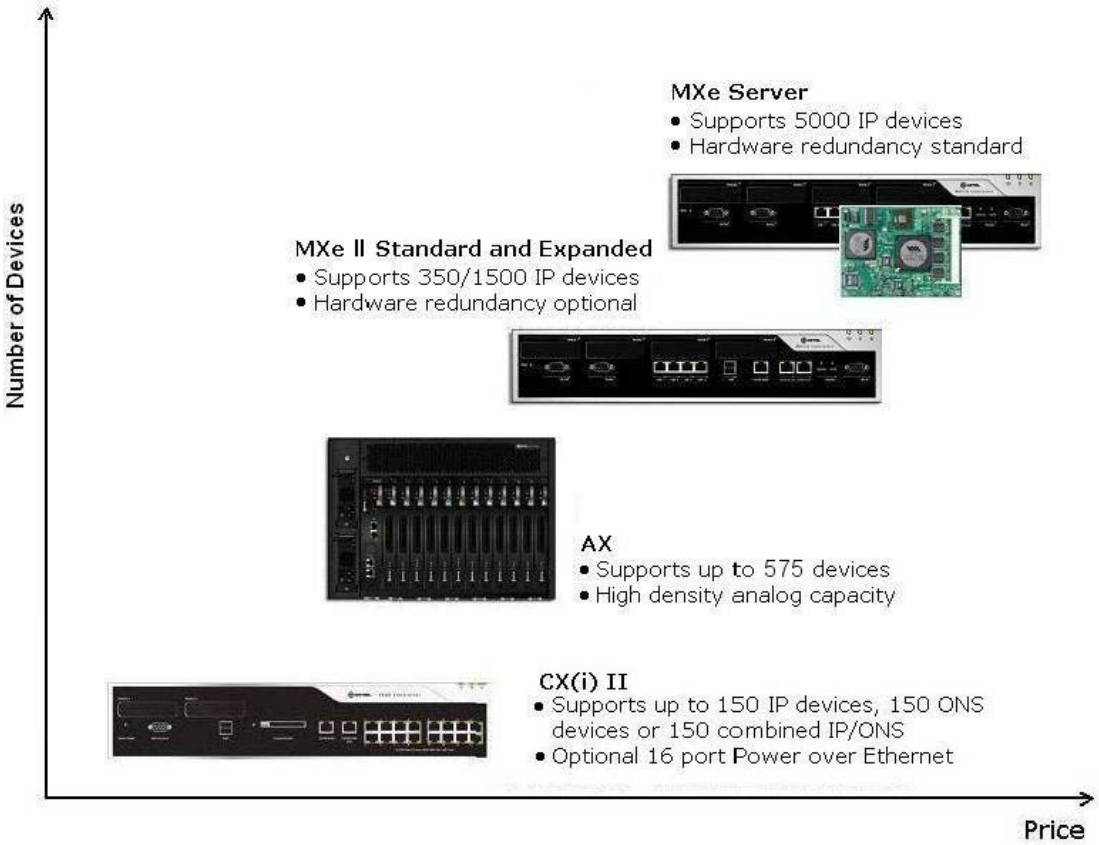


Figure 3: 3300 ICP Controllers — Scaling to Site Requirements

The controllers scale to meet the requirements of small to large scale sites.

- **CX II and CXi II Controller** — 2nd generation versions of the CX and CXi controllers that provide support for up to 150 devices without the need for additional DSP resources.
- **AX Controller** – optimized for analog devices, this unit supports a maximum of 100 IP devices or a maximum of 288 ONS devices (or a combined maximum of 300 devices). Note that when installed in a low traffic environment (for example, Hospitality), the AX can support 288 analog sets and 300 IP sets, for a combined total of 588 devices. Up to 300 IP devices can be supported under low traffic conditions.

- **MXe II Standard Controller** – supports a maximum of 300 IP devices or 350 ONS devices (or a combined maximum of 350 IP/ONS devices).
- **MXe II Expanded Controller** – supports a maximum 1400 IP devices or 1500 ONS (or a combined maximum of 1500 IP/ONS devices).
- **MXe Server** – supports up to 5,000 IP devices. This controller must be deployed in conjunction with other 3300 controllers because it is a dedicated call control engine for IP devices only. It connects over the LAN/WAN to other 3300 controllers which act as media gateways.

Controllers can be networked together over an IP infrastructure to deliver solutions for large or multi-site organizations.

Modules are field replaceable units (FRUs) that expand the functionality and capacity of the controller. Modules are installed in external and internal slots in the controller. The number of available slots depends on the controller model. Communication interface modules— such as the Dual FIM, Dual T1/E1 module, T1/E1 Combo Card, and Quad BRI Framer—are installed in slots that are accessible externally from the front or rear panel of the controllers.

The controllers have the following common physical features:

- External casing: All of the components may be stacked or rack-mounted (in a 19-inch rack).
- Power supply: Each unit has its own standard male IEC AC input connector for power.
- LEDs: All LEDs are located on the front or rear of the units for visual indication of circuit status.
- LAN/WAN ports: RJ-45 connectors
- Maintenance port: DB-9 (RS-232)

The following sections provide an overview of the controller variants. For detailed information on the controller capacities, refer to the controller configuration tables in the *3300 ICP Engineering Guidelines*.

CX II and CXi II Controllers

The CX II/CXi II ships with an embedded Analog Main Board that supports 6 analog trunks and 4 analog extension ports. The CX II/CXi II includes all the required DSP in the base configuration. You only need to add cards and DSP resources for additional functionality, not for performance scaling.

The CX II and CXi II Controllers support:

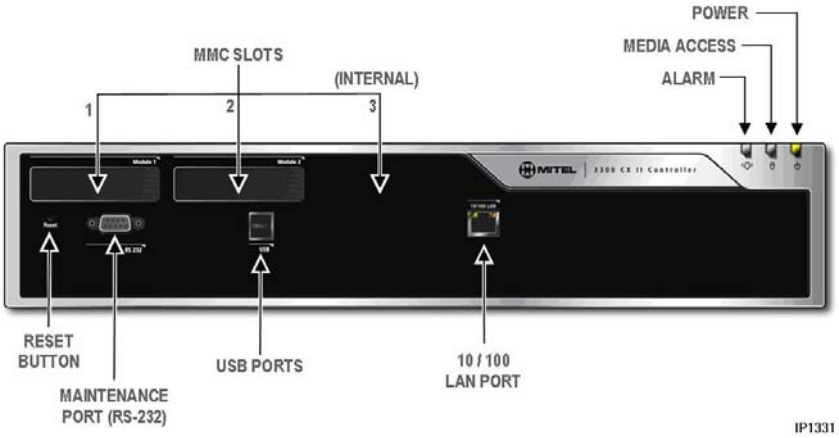
- up to 150 IP devices, or up to 150 combined IP/ONS devices
- the Analog Main Board (AMB) - provides six LS trunk ports with CLASS support (CLASS is available in North America and Latin American only), four ONS ports, a single Music-on-Hold port (one source supported), a single Paging port (one paging zone), and two System Fail Transfer circuits

- the Analog Option Board (AOB) - provides six LS trunks ports with CLASS support, four ONS ports, one System Fail Transfer circuits and one paging circuit
- one 10/100 BaseT WAN port (RJ-45 connector)
- one 10/100/1000 BaseT LAN port (RJ-45 connector)
- sixteen 10/100 BaseT LAN ports connected to an internal Ethernet Layer 2 switch (CXi II Controller only)
- SATA solid state drive or SATA hard drive for software storage.

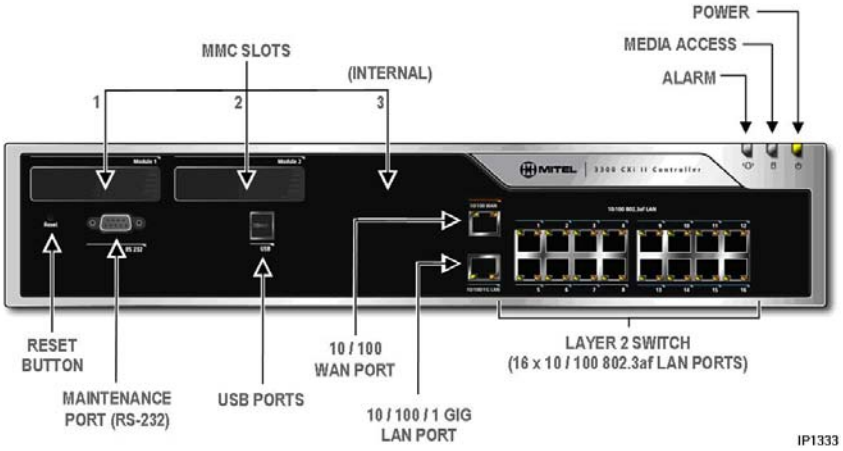
Optionally, you can install:

- one DSP II module for FAX Relay (T.38) / compression
- one or two T1/E1 Combo modules for digital trunking
- one or two Quad BRI Framer modules for BRI trunks
- Analog Option Board (AOB) for additional analog trunks and lines
- APC-CXi II assembly for running supported applications and services (such as the Mitel Border Gateway, Unified Communicator Mobile, and Live Business Gateway).
- a Quad Copper Interface Module (CIM) for connection of up to three Analog Service Unit IIs (ASU IIs).

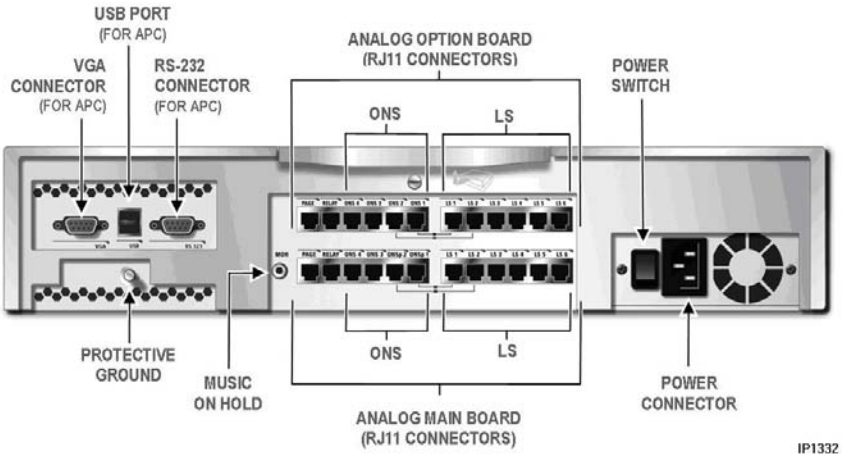
Note: The CX II and CXi II Controllers do not support Network Service Units (NSUs) or peripheral cabinets.



CX II FRONT PANEL



CXi II FRONT PANEL



REAR PANEL

Figure 4: CX II/CXi II Controllers

AX Controller

The AX Controller provides support for IP devices and analog devices. It's 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 288 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 288 IP sets, for a combined total of 576 devices.

The AX Controller provides

- 12 line card slots to support analog phones and trunks. The following cards (all hot-swappable) 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 FIM (external)
  - Quad DSP (external or internal)
  - Echo Canceller (external or internal)
  - Dual T1/E1 (external)
  - T1/E1 Combo (external)
  - Quad BRI (external)
  - Quad CIM (external)
  - DSP II (internal or external).

Optionally, you can install:

- 4 GByte flash card (required to support voice mail)
- 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.

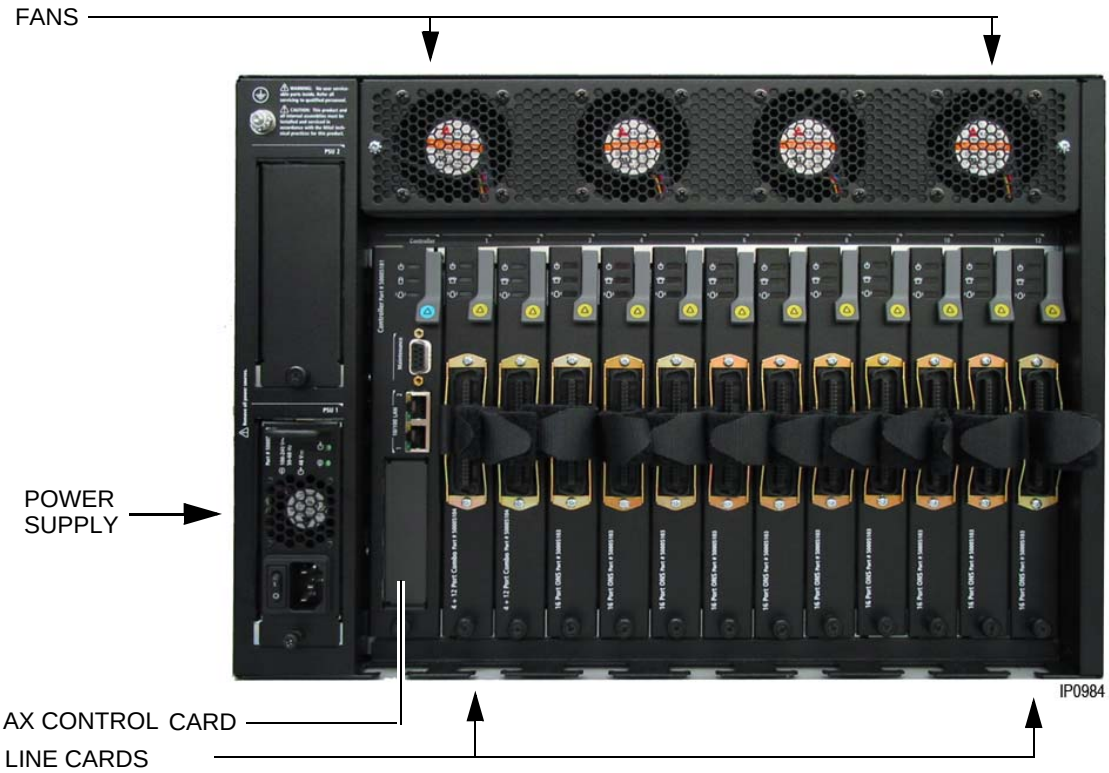


Figure 5: AX Controller Rear View

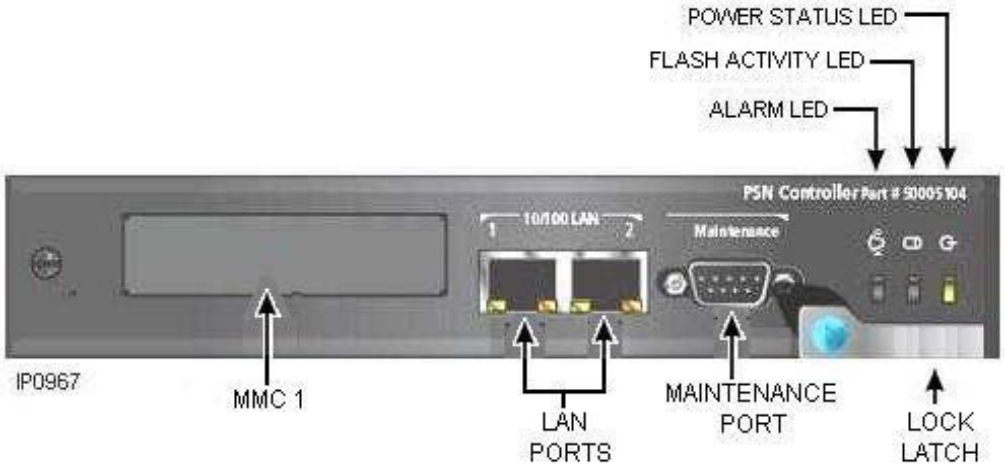


Figure 6: AX Controller Control Card

MXe II Controller

The MXe II Controller is available in two capacities: standard and expanded. Both versions include an embedded Analog Main Board and redundant cooling fans.

The MXe II Controller supports:

- up to 350 devices (combined IP/ONS) in the standard configuration
- up to 1400 IP devices and 1500 ONS devices (1500 combined IP/ONS) in the expanded configuration
- up to 1400 SIP devices/users.

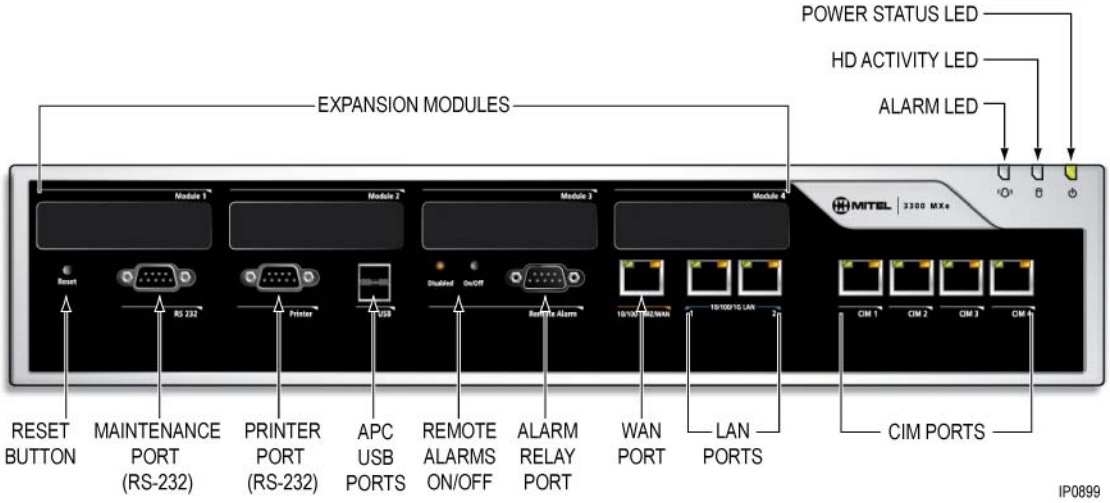
The MXe II Controller can also host up to seven SX-200 Bays providing connectivity for 96 ONS or OPS devices per bay. Only BCCIII-equipped bays are supported. Trunk cards are not supported.

The MXe II Controller provides:

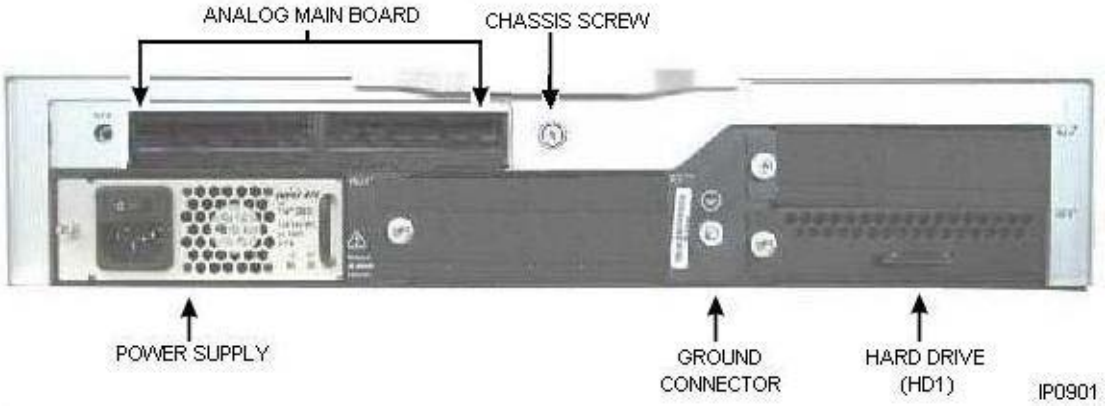
- two 10/100/1000 BaseT Ethernet LAN ports (RJ-45 connector)
- one 10/100 BaseT Ethernet WAN port (RJ-45 connector)
- four externally accessible slots and two internal slots for optional modules
- four CIM ports
- an Analog Main Board that provides 6 analog trunks and 4 analog extension ports
- an alarm relay port
- SATA solid state drive or PATA hard drive for software storage.

Optionally, you can install

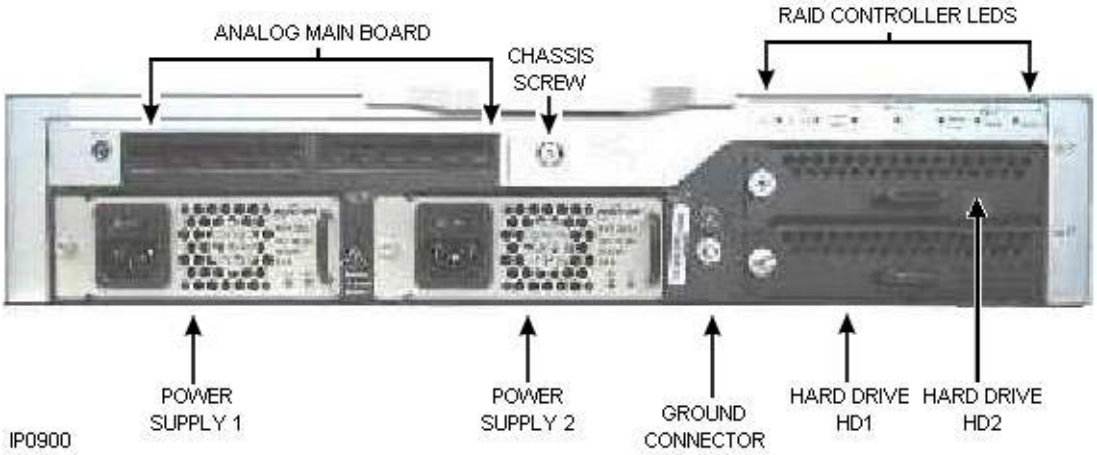
- MXe II Expanded Processor Package to upgrade from standard capacity (350 devices devices and 64 E2T channels) to expanded capacity (1500 devices and 128 or 192 E2T channels)
- two Quad DSP modules for G.729a compression
- two octal DSP II modules for G.729a compression and T.38 FAX support
- up to four Dual FIMs for connecting NSUs, peripheral units, and bays
- up to four Dual T1/E1 Framer modules
- up to three T1/E1 Combo modules
- up to three Quad BRI Framer modules
- up to two Quad CIMs for connecting ASUs and bays
- power and disk drive redundancy with the addition of a RAID (Redundant Array of Independent Disks) controller, a second hard disk, and a second AC PSU.



FRONT PANEL



REAR PANEL - SINGLE HARD DRIVE



REAR PANEL - REDUNDANT HARD DRIVE

Figure 7: MXe II Controller

MXe Server

The MXe Server is purely a call control server for IP devices. It supports up to 5,000 IP devices and connects over the LAN/WAN to other 3300 ICP controllers to access digital trunks and analog devices. The MXe Server uses the same chassis and many of the same components as the MXe II Controller, but it includes an additional application processor card (APC). The MXe Server is shipped with a RAID controller, dual hard drives, and dual power supplies.

The MXe Server provides the same call processing features and applications as the other controllers, running MCD software on the second application processor card.

The MXe Server supports:

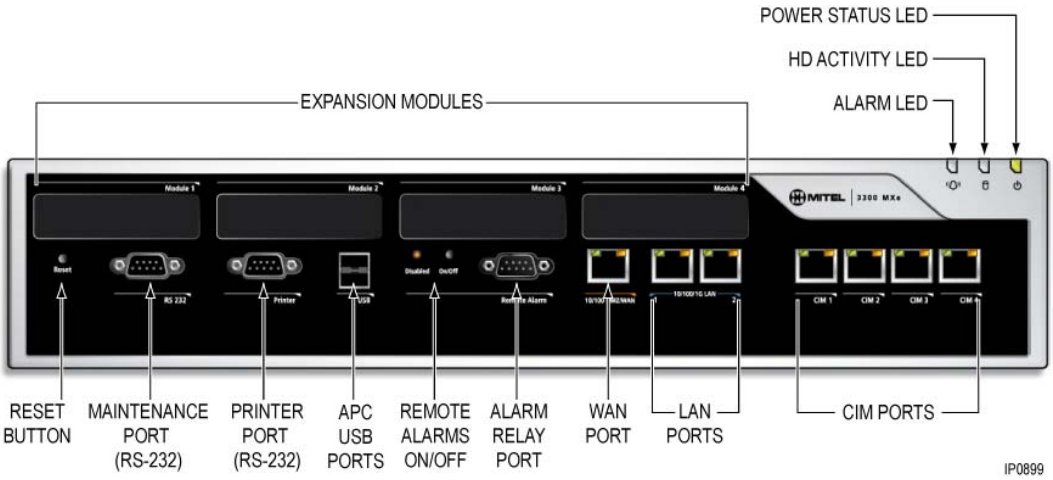
- up to 5000 IP or SIP devices
- hardware redundancy as a standard feature — the MXe Server ships with a RAID controller, two hard disks, and redundant AC power supplies and cooling fans.

Note: Though the MXE Server does not directly support ONS devices or trunks, it can, however, connect to other 3300 ICPs that can provide ONS & Trunk connectivity.

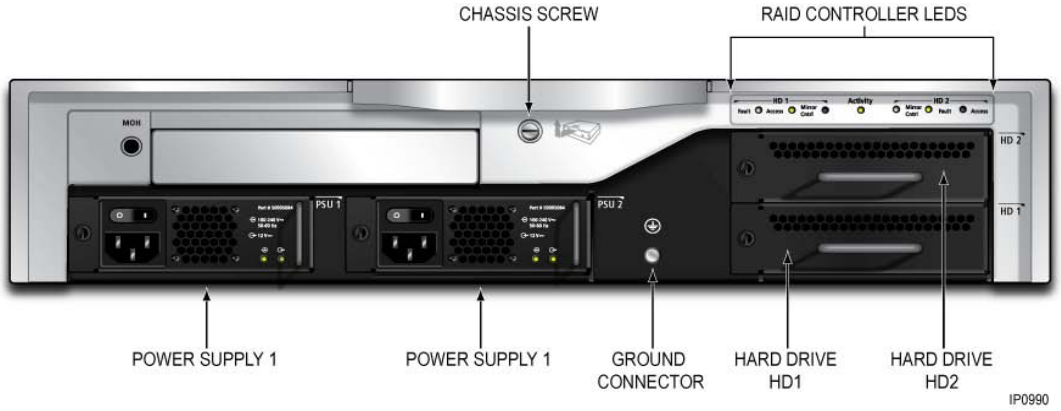
The MXe Server provides:

- expansion slots for up to two optional Quad DSP or octal DSP II MMCs
- two 10/100/1000 BaseT Ethernet LAN ports (RJ-45 connector)
- one 10/100 BaseT Ethernet WAN port (RJ-45 connector)
- two 128-channel Echo Canceller MMCs (installed internally as standard components)
- two PATA hard drives.

Note: The four CIM ports are non-operational and the printer port is used only for communications with the APC.



MXe SERVER FRONT PANEL



MXe SERVER REAR PANEL

Figure 8: MXe Server

Analog Support

You can add analog support to a controller with an Analog Services Unit II, Analog Main Board, or Analog Options Board. The following table summarizes the analog support for each controller type:

Table 1: Analog Support

Controller	Quad CIMs	ASU IIs	Analog Main Board	Analog Option Board
CX II / CXi II	1	3 with one Quad CIM installed	1	1 (optional)
AX	1	4 with one Quad CIM installed	Not supported	
MXe II	2	4 without any Quad CIMs installed 8 with one Quad CIM installed 12 with two Quad CIMs installed	1	Not supported
MXe Server	0	Not supported		

Quad Copper Interface Module (CIM)

A Quad CIM MMC provides four CIM ports that allow you to connect ASU IIs to the following 3300 ICP controllers:

- CX II / CXi II Controllers support one Quad CIM module. Only the first three ports on the Quad CIM are functional, the fourth is not supported. Therefore, you can only connect up to three ASU IIs.
- AX Controller supports one Quad CIM module for the connection of up to four ASU IIs.
- MXe II Controller: This controller has four embedded CIM ports allowing the connection of up to four ASU IIs. You can add up to two Quad CIM MMCs to increase the number of supported ASU IIs to 12.
- MXe Server: Although the MXe Server has 4 CIM ports, they are not operational. The LEDs will continue to flash, but they do not support ASU IIs.

The CIM ports require standard 8-pin modular jacks (RJ-45) consisting of 2 balanced signal pairs on Unshielded Twisted Pair (UTP) crossover cable. The CIM supports a distance of up to 100 feet or 30 meters between the controller and the ASU II.

Analog Services Unit II

The ASU II is a common platform to deliver analog trunks and extension services to all markets. It comprises a chassis with two card slots. Depending on how you configure the unit with line cards, the ASU II chassis can support up to 48 ONS phones and up to eight LS trunks.

Two card variants (both hot-swappable) are available to support analog phones and trunks:

- 24-port ONSp card provides 24 ONS lines for provisioning extensions outside the building. The ports on this card are protected against surge and lightning.

- 12-port ONS/ 4-port LS Trunk Combination card provides analog line and trunk capability in a single card:
  - 12 On-Premise Station (ONS) Lines for analog phones and four Loop Start (LS) trunks for analog connection to a central office. The ONS ports on this card are protected against lightning.
  - Four System Fail Transfer (SFT) relays that provide direct connection between an analog telephone and a Loop Start trunk in the event of a system or power failure.
  - Custom Local Access Signaling Services (CLASS) is supported on the ONS circuits. CLASS allows the 3300 ICP system to pass Calling Line ID digits and CLASS name information to display sets that support Caller ID functionality.

Any card can fit into either slot and the cards can be inserted while the unit is operational.

Note: ASU IIs support DTMF telephones only; pulse or rotary dial phones are not supported.

Analog Main Board/Analog Option Board

The MXe II and CX/CXi Controllers support the Analog Main Board (AMB). Additionally, the CX/CXi can support the Analog Option Board (AOB).

The Analog Main Board supports:

- six Loop Start (LS) trunks
- four On-Premise (ONS) lines
- two Power Fail Transfer (PFT) ports
- one Music On Hold (MOH) circuit
- one Loudspeaker Paging circuit.

The AOB provides the controller with an additional

- six LS trunks
- four ONS lines
- one Music On Hold (MOH) circuit
- one Loudspeaker Paging circuit.

Custom Local Area Signaling Services (CLASS) is supported on embedded LS trunks and ONS lines. CLASS allows the 3300 ICP system to pass Calling Line ID digits and CLASS name information to display sets that support Caller ID functionality.

Digital Trunk Support: Units and Modules

The Network Service Units and digital trunk modules provide connectivity to digital trunks for public or private networks. The following table summarizes the digital trunk support for the controllers.

Table 2: Digital Trunk Support

Controller	Available External Slots	Dual FIMs	R2 Network Services Units	Dual T1/E1 Framer	T1/E1 Combo Modules	Quad Basic Rate Interface Framer Modules
CX II / CXi II	2	Not supported			maximum of any 2 modules	
AX	1	1	2	maximum of any 1 module		
MXe II	4	4	8	4	maximum of any 3 modules	
MXe Server	4	Not Supported				

Dual Fiber Interface Module (FIM)

The Dual Fiber Interface Module (FIM) converts:

- optical signals received over a fiber optic cable to electrical signals
- electrical to optical signals for transmission over the cable.

The Dual FIM module allows you to connect

- NSUs to the MXe II Controller and AX Controller
- Peripheral Cabinets to the MXe II Controller (the AX Controllers do not support Peripheral Cabinets).

Each Dual FIM allows you to connect up to two NSUs or two Peripheral Cabinets to the controller. The NSUs or Peripheral Cabinets connect to the controllers via fiber optic cabling. The fiber optic cabling allows units or cabinets to be located up to 14 km from the controller.

Note: The CX II / CXi II Controllers and the MXe Server do not support Dual FIMs or NSUs.

R2 Network Services Unit

R2 is a protocol converter that allows the R2 NSU to access an R2 National Public Switched Telephone Network (PSTN) using MF-R2 digital trunk signaling. The 3300 Controller also receives and processes Calling Line Identification (CLI) and allows the information to be displayed on the user's telephone display screen.

The R2 NSU supports the CCITT Blue Book, Volume VI, Fascicle VI.4, Specifications of Signaling System R2, Recommendations Q.440 to Q.490 (with the exception of Echo Suppression (Q.479), Test Calls (Q.490) and international signals).

The R2 NSU converts the following:

- Incoming MF-R2 signals from the PSTN into Digital Private Network Signaling System (DPNSS) signals for the system
- Outgoing DPNSS signals from the system into MF-R2 signals for the PSTN.

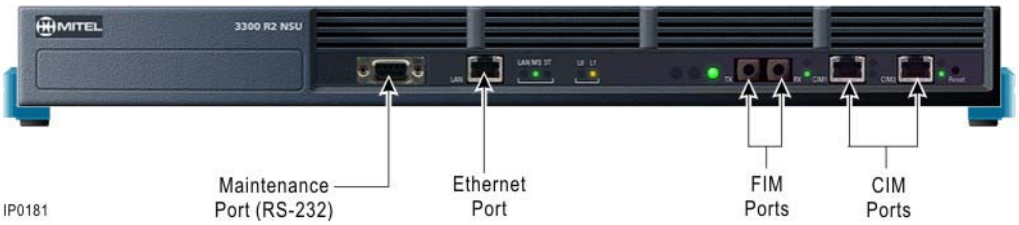


Figure 9: R2 Network Services Unit

Dual T1/E1 Framer Module

The Dual T1/E1 Framer module is a digital trunk interface that supports the direct connection of ISDN-PRI, T1/D4, QSIG, MSDN/DPNSS, and IDA-P trunks to the controller. This module has two ports supporting two digital links. Each port can support a different protocol.

T1 interfaces (1.544 Mbits/sec) support:

- ISDN PRI and QSIG links composed of 23 B-channels (bearer channels) for voice or data plus one D-channel (data channel) for signaling
- T1/D4 links composed of 24 B-channels
- IDA-P.

E1 interfaces (2.048Mbits/sec) support ISDN PRI and QSIG links composed of 30 B-channels plus one D-channel.

Note: The D-channel backup, NFAS, and min/max capability features are not supported for embedded ISDN PRI links.

Note: The CX II / CXi II Controllers and the MXe Server do not support the Dual T1/E1 Module.

T1/E1 Combo Module

The T1/E1 Combo Module combines trunk support, DSP functionality, and resiliency support in a single module. T1/E1 trunk resiliency is designed for businesses where resilience is critical, but where only a single digital link to the PSTN is required. If a site's primary controller fails, this feature automatically transfers the support for the T1/E1 trunk from the T1/E1 Combo Module in the primary controller to a T1/E1 Module in the secondary controller.

The digital trunk port can be configured as a T1 interface (1.544 Mbps) that provides 24 B-channels for T1/D4 or an E1 interface (2.0 Mbps) that provides 30 B-channels for E1. The DSP provides resources for voice echo cancellation. Embedded PRI is available for the 3300 ICP via the T1/E1 module.

Quad Basic Rate Interface (BRI) Framer Module

The Quad Basic Rate Interface (BRI) Framer Module is a digital trunk interface that supports the direct connection of BRI trunks to the controller eliminating the need for a BRI NSU.

The BRI Framer Module has four ports supporting four digital links. Each port may be configured as either of the following:

- T (trunk) interface for links from a BRI Central Office.
- S (subscriber) interface for connecting up to eight BRI devices to the controller.

Note: S interfaces support only basic call features such as calling number display for BRI devices. BRI call handling such as Hold or Transfer is not supported. BRI devices are not line powered from the Quad BRI Framer. The U interfaces are not supported.

The Quad BRI Framer is supported in Europe, Middle East, Africa, and Australia.

SX-200 Bay Support

SX-200 Cabinet

The MXe II Controller can support up to seven SX-200 Bay cabinets. The SX-200 Bay cabinet holds up to 12 card slots: eight slots support line (ONS, OPS or DNI) cards, and four slots support the control cards and the FIM or CIM carrier cards. The bays are connected to a 3300 ICP using FIM or CIM cables.

Note: SX-200 Bays are supported by MXe II Controllers only.

Note: SX-200 Bays are **not** supported in Europe.

SX-200 Peripheral Cards

The SX-200 Bay cabinet consists of Peripheral Interface Cards. The peripheral interface cards connect peripheral devices (such as SUPERSET™ telephones) to the system. They are located in slots 1 through 8. They include

- Digital Line Card
- On-Premise (ONS) Line Card
- Off-Premise (OPS) Line Card.

Each ONS port requires an ONS licence on the 3300 ICP, which is counted against the 3300 ICP software device limits.

Digital Network Interface Line Card

The Digital Network Interface (DNI) line card provides an interface from the system to Mitel SUPERSET telephones and SUPERCONSOLE 1000 attendant consoles.

Note: Only 4000-series SUPERSET telephones are supported.

The DNI line card supports voice/data transmission at the rate of 128 kb/s (64 kb/s on each of two voice channels) and 16 kb/s on one signaling channel over a loop length of up to 3280 ft (1000 meters), using 24 - 26 AWG wire (25 - 27 IWG). The DNI line card provides full duplex digital transmission of both voice and data.

ONS/CLASS Line Card (North America Only)

The ONS (On-Premise)/CLASS (Custom Local Area Signaling Service) line card has 12 circuits that connect up to 12 standard telephones with line loop resistance, usually not exceeding 600 ohms including the telephone. As such, the card is used to connect internal telephone extensions close to the system. It also supports modems and fax machines. These cards install in any peripheral interface card slot and are hot-swappable.

OPS Line Card (North America Only)

The OPS Line Card contains six off-premises line circuits. An Off-Premises (OPS) line circuit is used where the line goes outside the building that houses the PBX and the line may be exposed to extraneous high voltages or induced currents.

Peripheral Interface Module Carrier Card

The Peripheral Interface Module Carrier Card (PIMCC) holds a FIM II or CIM to provide the fiber or copper interface between a peripheral cabinet and the 3300 ICP. The FIM II and CIM interface modules on the BCC III also provide the connectivity needed to the 3300 ICP, making the PIMCC dispensable.

The Peripheral Interface Module Carrier card plugs into slot 12 of an SX-200 cabinet.

System Resources: Processors, Cards, and Modules

This section describes the cards and modules that support the system. To meet site requirements, you may need to add additional system resources to the controller. When planning a site, refer to the Configuration Tables in the Engineering Guidelines to determine if additional system resources, such as compression, echo cancellation, or Ethernet-to-TDM (E2T) channels are required.

Processors (E2T/RTC)

The CX II/CXi II, AX, and standard MXe II Controllers use a single processor to perform the Real Time Controller (RTC) functions and the Ethernet-to-TDM (E2T) functions. The expanded MXe II Controller and MXe Server have separate processors for these functions.

The E2T converts voice streaming between Internet Protocol and Time Division Multiplexing (TDM) signals. The RTC runs the call control for the controller and acts as a gateway for the IP signals/packets.

Digital Signal Processor Modules

The Digital Signal Processor (DSP) Modules perform basic telephony and compression functions including:

- Conferencing
- Voice Mail playout and recording
- Call Progression tone generation and detection
- Auto-attendant support
- G.729a compression (for IP trunking and wireless phones)
- FAX over IP (T.38) and additional G.729a compression (provided by the high-density DSP II MMC)

The system allocates DSPs for

- conferencing (at startup)
- voice mail depending on the number of ports programmed in the customer database (at startup)
- tone generation and detection as required by traffic conditions (on a per call basis).
- auto-attendant features.

You can add additional DSP resources to a controller by adding a Quad DSP module, a Dual DSP Module, or a DSP II Module. The Dual DSP module is only available for CX/CXi systems. Instructions on how to calculate system DSP requirements are provided in the *3300 ICP Technician's Handbook*.

Echo Cancellation Module

The Echo Canceller (EC) module provides echo cancellation on E2T channels. Each bi-directional E2T channel requires one bi-directional EC channel. The EC module provides 128 EC channels.

Applications Support

Application Processor Card - MXe Server

The Application Processor Card (APC-MXe) is an embedded PC card that is standard in the MXe Server. It runs the Mitel Communications Director call control software. In the MXe Server, the RTC performs the role of Media Server, with the APC-MXe performing call control functions.

The Mitel Communications Director software and the controller software are pre-installed on the hard drives that are shipped with the MXe Server.

Application Processor Card - CXi II

The APC-CXi II is an optional card that you can install in the CXi II controller. The APC-CXi II hosts the Mitel Standard Linux (MSL) Server software that supports applications such as:

- **Mitel Border Gateway (MBG)** - The MBG is the evolution of the Mitel Teleworker Solution. It functions as a platform for the secure deployment of multiple network connectivity services in a number of network edge scenarios:
  - Teleworker Service - A secure plug & play solution for remote workers
  - SIP Trunk Proxy Service - A SIP-aware firewall at the edge of the company network.
  - Application Web Proxy Service - A secure teleworking solution for remote and home-based employees. This solution is supported on standard Mitel IP Phones.
- **Unified Communicator Mobile** - Enables users to "twin" their desk phone to another internal or external phone such as a cell phone.
- **Live Business Gateway** - enables communication between Microsoft® Office Communicator (MOC) and the 3300 ICP allowing MOC to take advantage of the basic and supplementary phone services offered by the 3300 ICP.

For information on how to program and use software blades and services, refer to the Mitel Standard Linux documentation at http://edocs.mitel.com.

The Application Processor Card (APC) requires a dedicated hard drive.

Firewall (CXi II, MXe II and MXe Server only)

For security, the CXi II, MXe II and MXe Server can function as a firewall, dropping or rejecting unknown packets, allowing or disallowing IPSec and PPTP pass-through, and performing many-to-one NAT (IP masquerading).

By default, all inbound traffic is blocked by the firewall with the exception of packets for establishing PPTP VPN tunnels and ICMP requests to the IP address of the WAN interface.

3300 ICP Network Support

Voice Networking

Lines

The 3300 ICP supports the following internal voice connections:

- Ethernet connections are provided on the 3300 ICPs to connect the system to the customer's Ethernet LAN. IP phones communicate with the 3300 ICP over the Ethernet LAN.
- On-Premises (ONS) lines are for industry-standard DTMF telephones. These lines are supported by the ONSp line card in the ASU II. ONSp card variants have "protected" ports that protect the circuitry against lightning and surges. ONS ports are also available on the Analog Main Board (AMB), or Analog Option Board (AOB).

Note: Pulse and rotary dial telephones are not supported.

- Digital Network Interface (DNI) lines provide an interface for Mitel legacy digital telephones and consoles. These lines are supported by the DNI Line card connected into the SX-200 Bay.
- On the client side, SIP phones are starting to proliferate across the full spectrum of telephony vendors.

Trunks

The system can connect to the Public Switched Telephone Network (PSTN) or to private networks over digital and analog trunks. The following digital links are supported:

- **DS1 Links** - The system supports the following protocols: D4, Q.Sig, MSDN/DPNSS, Primary Rate ISDN (DM-250, DMS-100, Bellcore National ISDN, 4ESS, NI-2, 5ESS NI2), and XNET over PRI protocols. The system connects to DS1 links through the T1/E1 Modules or Universal NSU.
- **E1 Links** - The system supports DASS II, MSDN/DPNSS, Q.Sig, Primary Rate ISDN (Euro ISDN (CTR4)), and XNET over PRI protocols. The system connects to E1 links through the T1/E1 Modules or Universal NSU.
- **R2 Links** - The system supports the CCITT Blue Book, Volume VI, Fascicle VI.4, Specifications of the Signaling System R2, and Recommendations Q.440 to Q.490 (with the exception of Echo Suppression (Q.479), Test Calls (Q.490) and international signals). The system connects to R2 links through the R2 NSU. Note that many countries use R2 signaling but do not adhere to the CCITT recommendations in their entirety. The 3300 ICP is completely flexible and supports regional variations of the R2 protocol. Line signaling, tone interpretation, and timing parameters for the converter can be adapted to suit any national or regional requirement. For example:
  - Line signaling features allow you to program up to four bits to define the incoming and outgoing patterns for line signals such as Idle and Answer
  - Register signaling features allow you to program the type of address signaling

termination (signaled or timed) and whether signaling should be fully-compelled or semi-compelled. These features allow the individual definition of each register signaling tone.

- **PRI Links** - The system supports DM-250, DMS-100, Bellcore National ISDN, 4ESS, NI-2, 5ESS NI2, NI13, Euro ISDN (CTR4) protocols, and IDA-P PRI protocol used in Hong Kong. The system connects to PRI links through the T1/E1 Modules or Universal NSU.
- **BRI Links** - The system supports Euro ISDN 2B + D, Basic Rate Interface, or the North American ISDN-1 and ISDN-2 protocols. The system connects to BRI links through BRI NSUs or through Quad BRI Framer modules:
  - The BRI NSU provides connectivity for Basic Rate ISDN (BRI) transport of both data and voice traffic. This unit is available only in the North American variant, which supports U-Interface (user-side interface).
  - The Quad BRI Framer module supports the S-Interface for the European market. This module is also supported in North America where the S-Interface is available.
- **SIP Trunks** - Allow the 3300 ICP to connect to the Service Provider through the SIP protocol over the IP network. Service Providers offer SIP trunks that provide flexible and cost-effective WAN solutions for the 3300 ICP. The SIP Trunking solution provides basic feature functionality, billing capability, Emergency Services support, FAX support, and more.

The following analog trunks are supported:

- **Analog CO trunks** - Loop start (LS) ports on the ASU II can be used to interface analog CO trunks with the system.
  - LS CLASS trunks are available on the Analog Main Board, Analog Option Board, ASU II, and AX Controller line cards.

IP Networking

IP Networking provides customers with an option for networking systems together. Instead of leasing dedicated voice circuits, customers can route voice traffic over the existing LAN/WAN infrastructure. Mitel's implementation of IP Networking can use

- a point-to-point topology to optimize network resources, or
- a fully meshed topology to maximize inter connectivity between systems.

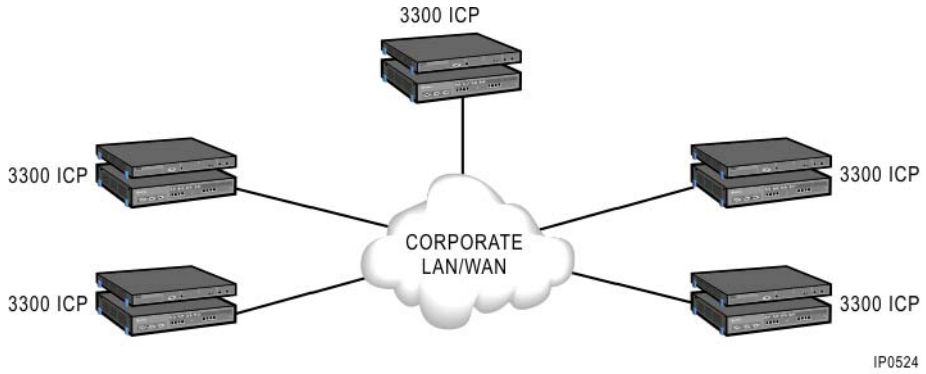


Figure 10: IP Networking - Point to Point Topology

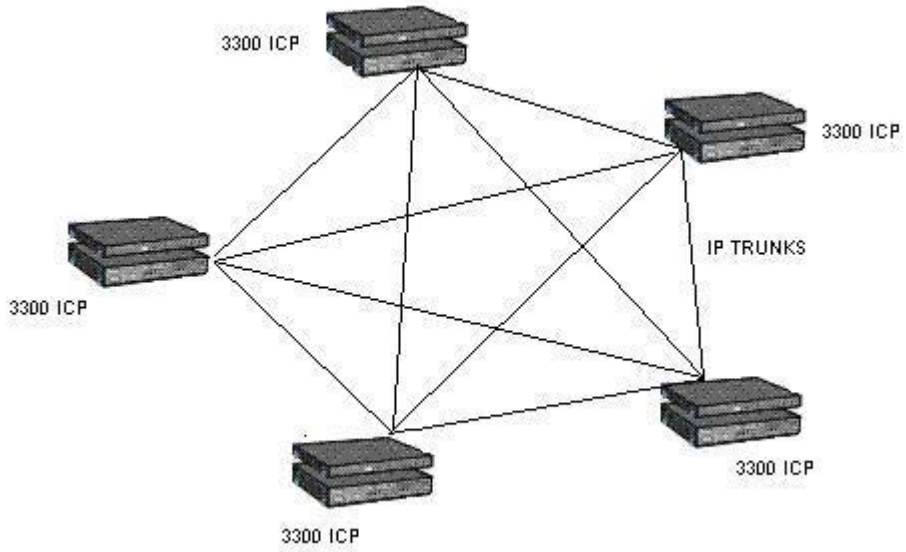


Figure 11: IP Networking - Fully Meshed Topology

IP Networking supports the MSDN/DPNSS protocols over the IP infrastructure. Controllers can be clustered in a single location to provide greater resiliency than a single controller operating autonomously. 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.

A 3300 ICP with IP Networking enabled can be configured to act as an IP Networking gateway for SX-2000 and SX-200® PBXs, or third-party PBXs.

Figure 12: 3300 ICP as Trunking Gateway for PBXs

The IP Networking feature supports G.711 and G.729a encoding. Connections with up to 999 other network nodes are supported. A total of 2000 IP network connections are supported from any one node and up to 200 connections can be defined between any two nodes. IP Networking is enabled via the IP Networking license. One license is required per controller in the network.

SIP Trunking

To manage costs within an organization, many businesses are considering replacing their traditional PSTN connections with new SIP services deployed by service providers. Mitel expects to see a proliferation of these services in the future and SIP trunking has the ability to support these new network services.

The 3300 ICP connects to the service provider's network using the SIP protocol over the IP network. The SIP Trunking solution provides basic calling features, billing capability, Emergency Services support, FAX support, and more.

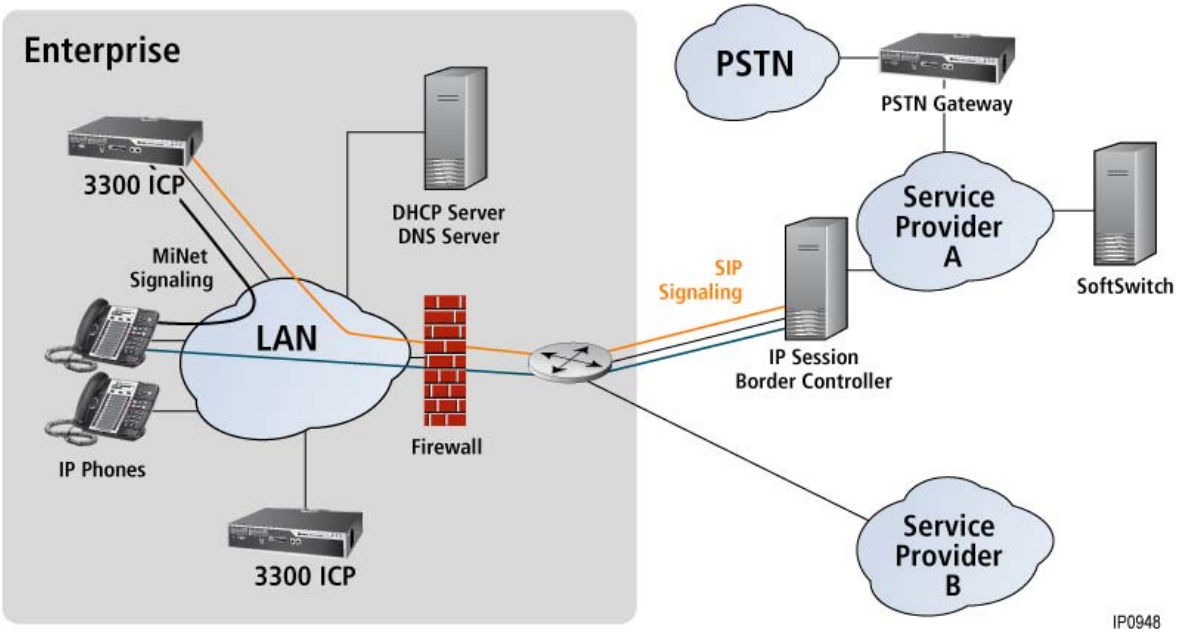


Figure 13: SIP Trunking

9-1-1

SIP Trunking supports 9-1-1 emergency service. The SIP service provider can be chosen as the outgoing emergency route.Caller's Emergency Service Identification (CESID) information must also be programmed.

DNS Support

Communication between SIP Service Providers and the 3300 ICP can be configured to use either Fully Qualified Domain Names (FQDN) or IP Addresses.

Configurable Real-time Transport Protocol (RTP) Packetization

3300 administrators can configure the voice stream packet rate for SIP trunks to their service provider's, choosing a rate between 10ms to 80ms (with 10ms increments). IP sets supporting this feature include the 5302, 5304, 5212, 5224, 5235, 5312, 5324, 5320, 5330, 5340, 5360.

Billing and SMDR

The Service Provider bills calls based on the peer connection to the 3300 ICP. The 3300 ICP records are created with a special SMDR tag entered in the SIP Peer Profile form. An SMDR tag can be enabled for outgoing and incoming calls.

Malicious Call Trace

For incoming SIP calls that are tagged for Malicious Call, the 3300 records the Media IP address and port used remotely. As well, the SIP signalling information is captured. This information cannot be sent to the SIP Service Provider, but the information is recorded if needed.

> **Note**: Malicious Call SMDR records are logged on the 3300 ICP. SIP endpoints cannot invoke Malicious Call Trace, but it is recommended that SMDR be enabled for SIP devices and gateways.

FAX Support

You can configure the 3300 ICP network to allow facsimile to be sent over the IP network using G.711 pass-through or IP network using FAX Relay (T.38).

Real-time, Group 3 facsimile (FAX) communication over IP networks using FAX Relay (T.38 standard protocol) allows you to transmit and receive facsimile over IP trunks between FAX machines on 3300 ICP (Release 9.0 or later) systems.

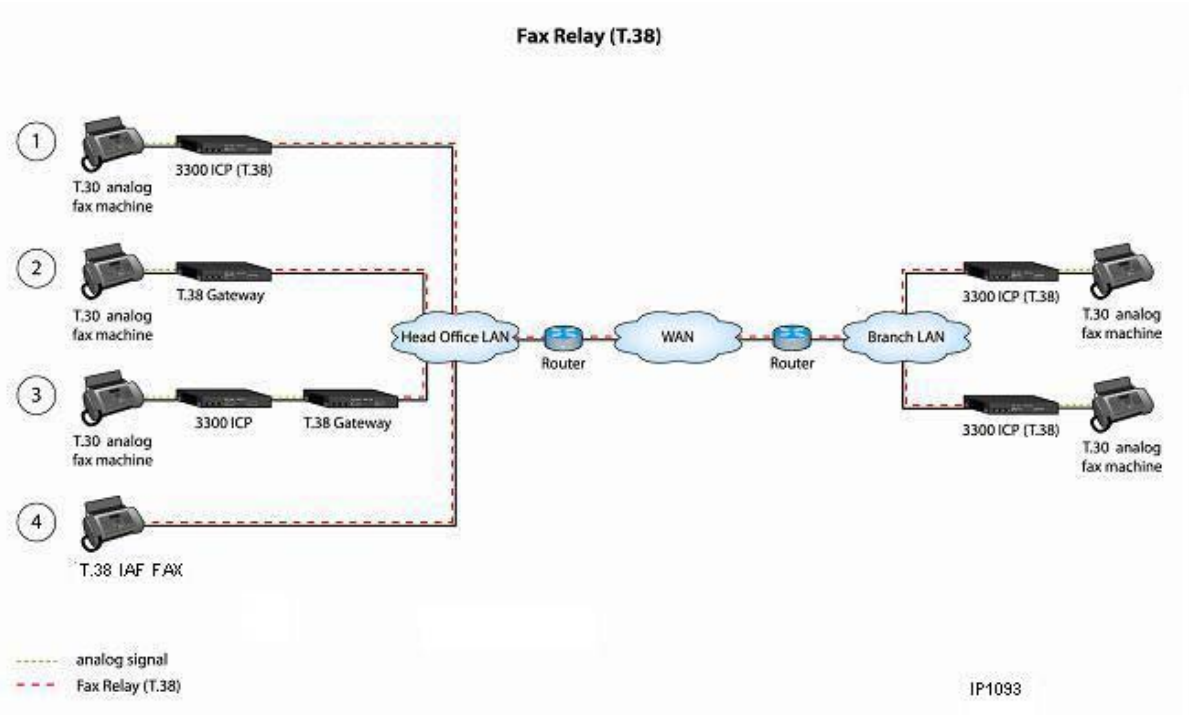
FAX Over IP Using T.38 FoIP

FAX Relay (T.38) is the preferred method because the quality and performance of FAX transmission is superior to that of the G.711 pass though method, especially over WAN links. The LAN and WAN do not have to be engineered specifically to support this method of FAX transmission.

The following diagram illustrates four different ways that FAX Relay (T.38) can be employed to support fax transmission:

- FAX machine connected to a T.38 protocol enabled 3300 ICP
- FAX call transferred by 3300 ICP to a T.38 protocol enabled gateway
- FAX machine connected directly to a T.38 protocol enabled gateway
- Internet Aware FAX (IAF) machine with built in T.38 gateway functionality

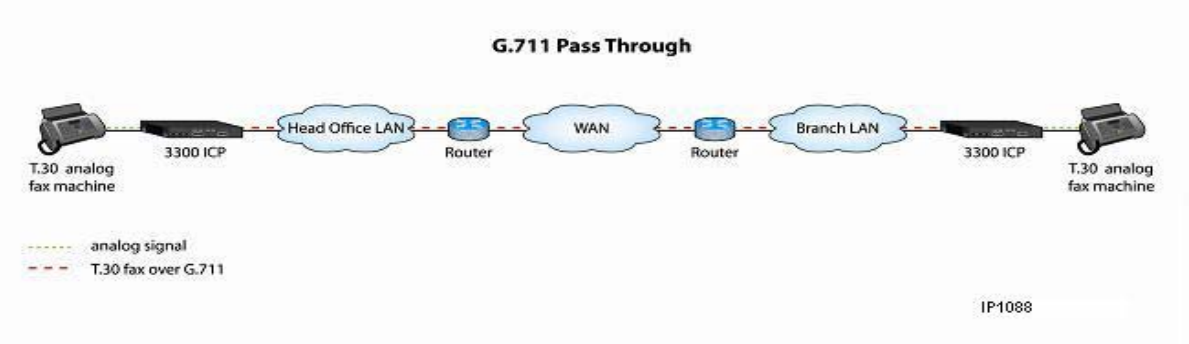
Figure 14: FAX Relay (T.38)



FAX Over IP Network Using G.711 Pass-Through

This method uses T.30 protocol with G.711 pass through to send FAX over IP trunks using the LAN and WAN. The 3300 ICP systems in the network must be running Release 7.0 or later.

Figure 15: G.711 Pass Through.



This method has the following advantages:

- allows you to send faxes between 3300 ICP systems over the IP network
- reduces the cost of sending long-distance faxes because the data is sent over the IP network and not the PSTN

However, this method has the following disadvantages:

- LAN and WAN must be engineered to meet the LAN quality guidelines specified in the *3300 ICP Engineering Guidelines*.

- Voice packets carrying fax on the network cannot be compressed.
- Quality and performance of fax transmission using G.711 pass through is inferior to the quality and performance provided by FAX Relay (T.38) protocol, especially over WAN links because the likelihood of experiencing network issues is higher.

Compression

Optimization of bandwidth is a key requirement in a VoIP system. In order to meet this requirement, the 3300 ICP supports G.729a voice compression. G.729a compression reduces the bandwidth required for a call from 64 kbps to 8 kbps plus packet overhead. By using voice compression across the LAN/WAN infrastructure, customers can ensure that they are able to optimize their bandwidth usage for voice calls. The mechanism for managing this feature is based on a zone concept. Groups of devices on the 3300 ICP can be placed in a zone so that calls between zones can be compressed while calls within the same zone are not. Zones can be defined within a controller's LAN infrastructure, between remote IP devices and the controller and across the WAN for multiple controller networks.

Most Mitel IP phones inherently support G.729a voice compression. Calls between IP Phones on the LAN/WAN infrastructure can thus be compressed to G.729a as required. For example, a call between IP Phone B and IP Phone D (over the LAN or WAN) can be compressed without system compression resources.

G.729a compression is also supported for calls that have TDM (Digital or Analog) endpoints that cross the LAN/WAN infrastructure. For example, a call from TDM phone A to IP Phone D can be compressed using compression resources in controller A to compress the LAN/WAN segment between Controller A and IP Phone D. The same compression occurs if TDM Phone A called TDM Phone C over the LAN/WAN, except that in this case compression resources would be required on both controllers.

Optional compression licenses and DSP modules can be purchased to enable TDM-to-IP compression on the 3300 ICP.

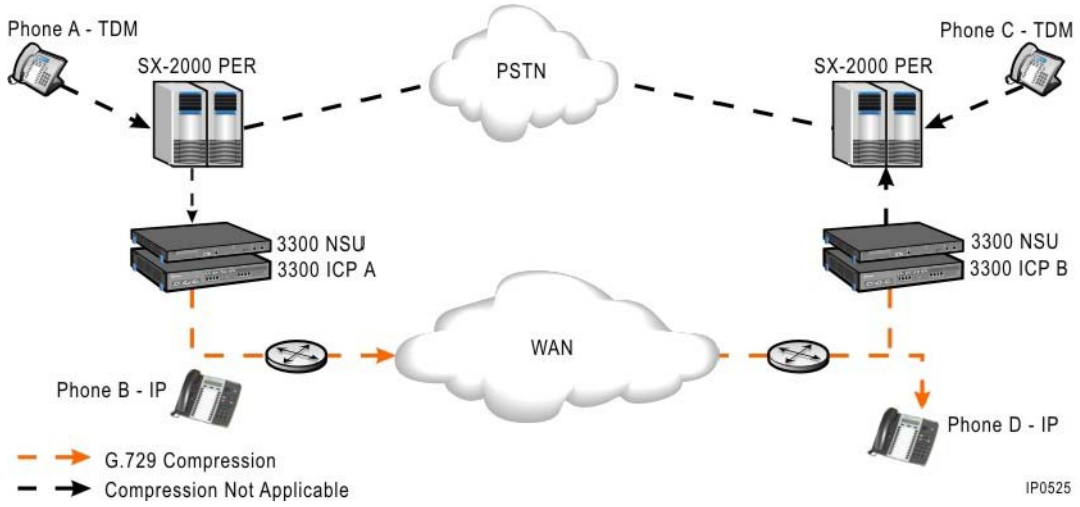


Figure 16: Voice Compression Between 3300 ICPs

Bandwidth Management

For many customers, one of the key benefits of IP telephony is the opportunity to reduce costs and ongoing management by eliminating controller hardware at small remote sites. IP phones can simply be deployed across the WAN (or Internet using the Mitel Border Gateway teleworker service) hosted off a centralized 3300 ICP controller with gateways for remote survivability. When remote sites are deployed, organizations must ensure that non-voice data has adequate bandwidth and that quality of voice can be preserved.

Where bandwidth between locations is restricted, you can reduce consumption by applying compression to the voice calls between IP Phones. Compression reduces the bandwidth demands of a standard voice call (G.711) by compressing the call using the G.729 codec. Compression is applied to calls between zones of IP Phones (see Compression on page 35 for details).

In addition, Mitel provides a bandwidth management feature that helps IT managers plan and justify network capacity expansions. The Bandwidth Management feature allows IT managers to perform the following tasks at predetermined zone access points (ZAPs) between the zones in the network:

- Measure and report consumed and available bandwidth
- Establish maintenance alarms when bandwidth consumption exceeds configured threshold levels
- Provide Call Admission Control, that is, reject new calls through a specific bottleneck point if the consumed bandwidth exceeds the maximum configured levels.

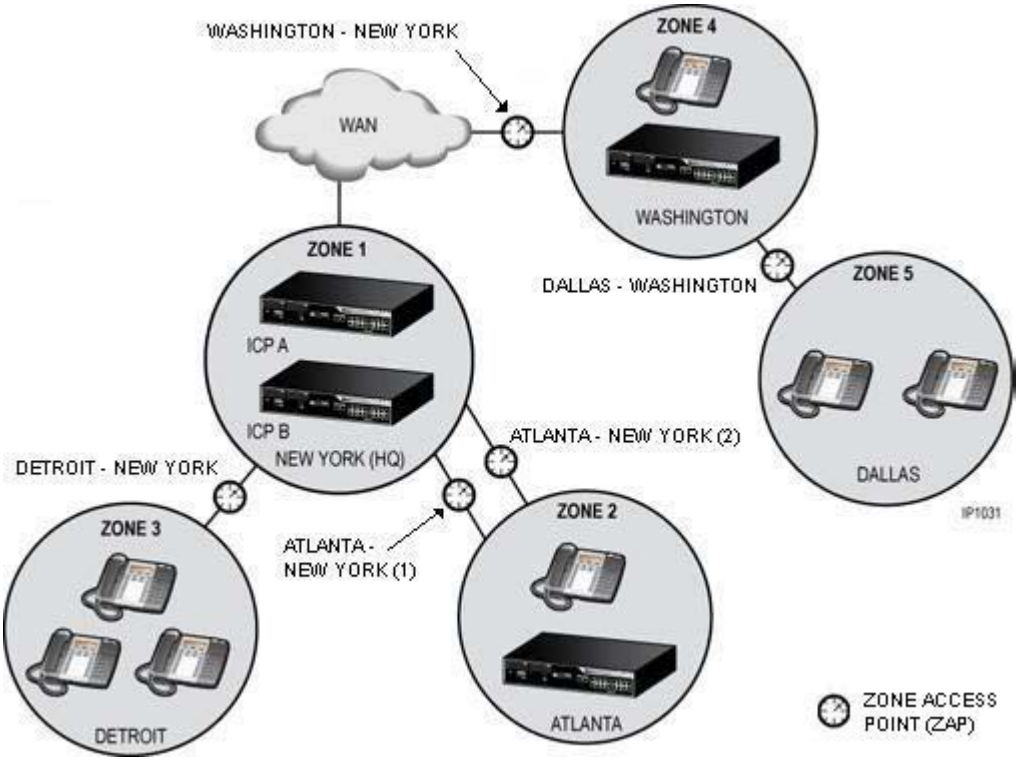


Figure 17: Bandwidth Management at Zone Access Points

Resiliency

Reliable communication is critical to businesses. Resiliency on the 3300 ICP increases communications reliability by maintaining calls in progress, handling new incoming and outgoing calls, and continuing to provide voice mail services in the event of a 3300 ICP or network failure. The Resiliency solution preserves system functionality in the event of network difficulties by distributing network intelligence throughout “resilient” clusters that anticipate and pro-actively mitigate system failure.

By taking advantage of IP-network characteristics of location independence, resiliency provides an extremely flexible solution to enhance system reliability. By using resources that are spread across the network, resiliency ensures that there is no single point of failure and that hardware resource utilization is optimized. Resiliency provides an obvious advantage over many other competing alternatives where solutions involve costly hardware redundancy for every controller (see See “Advantages Over Redundancy” on page 39.).

Resiliency is an IP-enabled capability that builds on existing 3300 ICP Voice Clustering (Portable Directory Number [PDN]), and call-routing principles. Existing clustering techniques are used to set up a cluster, which is then made resilient through the programming of boundary nodes, transit nodes, call routing ARS, and resilient devices. Resilient devices within the cluster are programmed on a primary and secondary controller. Should the primary controller experience a service outage, support for the resilient devices is automatically transferred to the

secondary controller. During the transfer of phone service between the primary and secondary controllers, calls in progress are maintained, ensuring that IP phone users are not affected by the controller outage. The figure below shows how a site can be configured with fully resilient devices. Node B is the secondary controller for the phones on Node A, and Node A is the secondary controller of the phones on Node B. Should either controller experience an outage, support for the phones is transferred to their respective secondary controller.

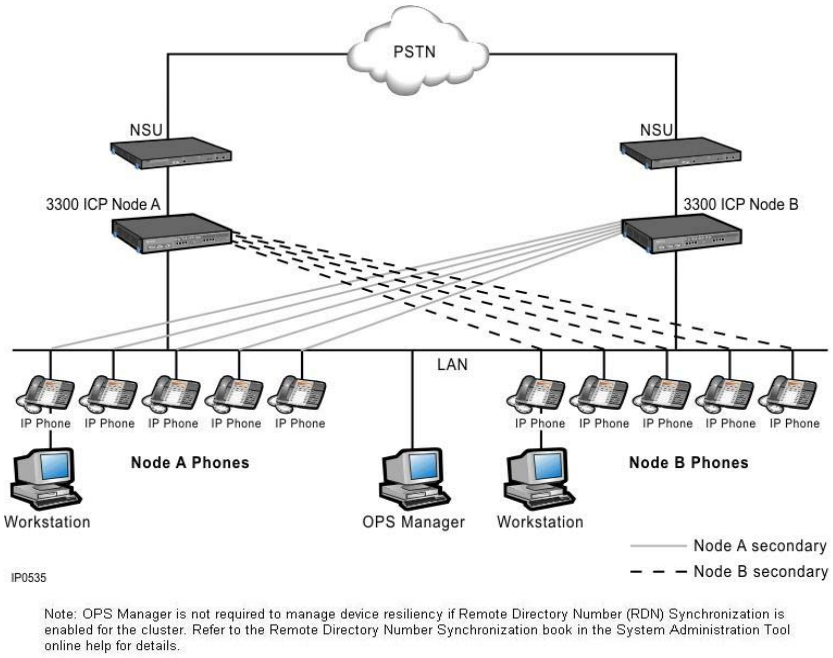


Figure 18: Resilient Configuration

As a further example, on a site where there is a MXe Server to support the users and an MXe II Controller to act as a media gateway to the public telephone network, the IP phones can be supported by the MXe Server (primary controller) and have backup support available from the media gateway controller (secondary controller). Resiliency is supported for:

- **IP Devices** - A secondary ICP provides service to phones in the event of a failure on the primary ICP or in the event of a networking failure between the phone and the primary ICP. The secondary ICP directs phones to transfer back to the primary ICP once the failure is corrected.
- **SIP Devices** - A secondary ICP provides service to phones in the event of a failure on the primary ICP or in the event of a networking failure between the SIP endpoint and the primary ICP. SIP device resiliency is dependent on the particular SIP device deployed. SIP endpoint resiliency applies to Mitel's 5300 series dual-mode IP Phones, IP DECT phones and other commercially available SIP devices.
- **Trunks** - A T1/E1 trunk connection can be critical to site operation if it is the only link to the network. If the primary ICP fails, a relay in the T1/E1 module physically transfers the trunk termination point to a T1/E1 module on the secondary ICP.
- **Calls** - Calls in progress over the IP infrastructure are maintained.
- **Features** - Most features are resilient and are available to a user while the user's phone is in service on its secondary ICP. Also any programming changes that users make to their

phones while the phones are on the secondary controller are automatically updated on the primary controller after the phone re-homes to the primary controller.

- **Voice mail** - Embedded voice mail continues to be provided to a device when it moves to a secondary ICP during failure.
- **Automatic Call Distribution** - ACD Agent calls, agent status (for example, logged in, make busy, Do Not Disturb, and so forth) and Agent Skill Groups are resilient.
- **Hunt groups-** Resiliency is supported for voice, voice mail, or recorder hunt groups. The 3300 ICP supports only voice hunt groups. Currently, only NuPoint Unified Messaging Release 10.0 and later supports voice mail or recorder group hunt group resiliency.
- **Ring groups**- Support for ring groups passes to the secondary controller if the primary fails.

The Mitel system management tool or the System Administration Tool can be used to provision resilient users and devices on primary and secondary 3300 ICPs.

Advantages Over Redundancy

Resiliency is less costly and more flexible than a redundant solution because it uses self-correction techniques that take advantage of the IP-network characteristics of location independence and network element distribution. While the redundancy model is highly effective and reliable, it is unnecessarily costly for some customers.

Distributed resilient networks offer the ability to route around failed or otherwise inaccessible portions of an IP network. This feature provides the following distinct advantages over the centralized 1+1 hardware requirements of a redundant solution:

- No single point of failure
- Lower hardware costs because of the efficient use of existing hardware.

Because any controller in the network can act as a secondary controller, Mitel Resiliency can be referred to as an "any +1" solution for system reliability. Rather than dedicating expensive, robust hardware to solving temporary and often infrequent system failures, Mitel Resiliency makes efficient use of a system’s existing capacity.

In resilient networks, a secondary controller is not limited to acting as a dedicated backup call-control host. In many cases, the secondary controller can also function as any one of the following:

- Full service controller (in a configuration where resiliency support is distributed among multiple controllers in the network)
- Group controller
- Wireless access controller
- Call center controller
- Video conference controller
- IP network gateway
- PSTN gateway

- Voice mail server.

Devices that Support Resiliency

The following Mitel IP devices support resiliency:

- All 5000 series, 5100 series, 5200 series, and 5300 series IP Phones
- 5540 IP Console
- 5550 IP Console
- 5560 IPT
- IP PKM 12 and IP PKM 48
- 5310 IP Conference Units
- Teleworker service sets.

Note: Resiliency is an IP solution that does not support ONS, DNIC and older 4000-series IP Phones.

Resilient clusters can contain pre-4.0 3300 ICPs and Mitel legacy SX-2000 PBXs. These devices cannot function as secondary controllers, but they can be part of a resilient solution as boundary nodes and transit nodes.

For detailed information on Resiliency, see the *3300 ICP Resiliency Guidelines*.

Rapid Spanning Tree Protocol

Both Rapid Spanning Tree Protocol (RSTP) and Spanning Tree Protocol (STP) are supported on the CXi II/CXi II Controller, MXe II Controller, AX Controller, and MXe Server:

- Spanning Tree Protocol (STP) is a Layer 2 Link Level protocol specified by the IEEE (802.1D) that runs on bridges and switches.
- Rapid Reconfiguration of Spanning Tree Protocol (RSTP) is also a Layer 2 Link Level protocol specified by the IEEE (802.1w) that runs on bridges and switches.

STP and RSTP serve the same purpose. The difference between the two protocols is how quickly the algorithms converge on a network. RSTP "reconverges" networks faster than STP.

In an Ethernet network that is not using STP/RSTP, multiple active paths between devices are not allowed since multiple paths will cause network loops.

Network loops will cripple a network because a broadcast or multicast packet sent from Station “A” to Station “B” will be forwarded by Switch “B” to Station “B” and also back to Switch “A”, when Switch “A” receives the packet it will then forward the packet back to Switch “B” and the cycle will repeat for infinity causing a broadcast storm.

Both STP and RSTP allow for physical path redundancy by placing redundant network paths into a standby mode by blocking traffic on redundant ports. Should a currently active network

path fail due to a Bridge/Switch failure or a network cabling failure, STP/RSTP will enable the network path that was previously held in a standby mode and network connectivity will be restored. The following figure depicts how STP/RSTP breaks a potential network loop by blocking traffic on one of the ports on Switch “B”.

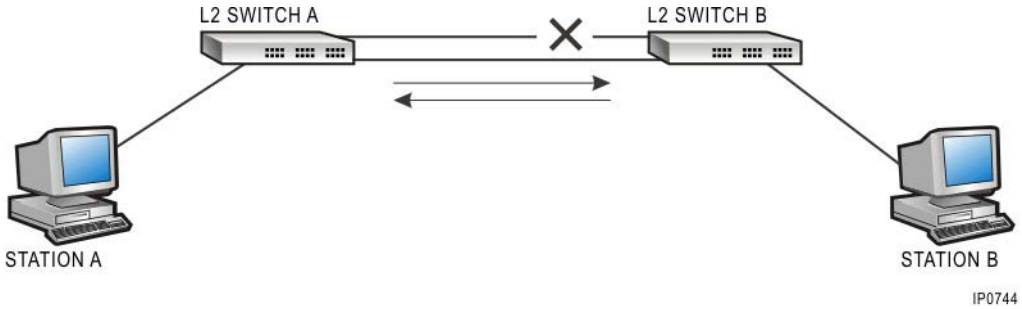


Figure 19: Spanning Tree Protocol (RSTP/STP)

Gateway Solutions

Industry’s acceptance of IP telephony has resulted in vast improvements in cost savings and in the number and quality of voice-related applications. Many end customers recognize the benefits of these enhancements but are sensitive about replacing their entire voice infrastructure just to obtain benefits at a particular site. In such situations, using the 3300 ICP as a gateway into IP telephony has many advantages.

By integrating a 3300 ICP into an existing third-party PBX, customers retain their previous investment in communication equipment while also taking advantage of the benefits of a superior IP telephony solution. The 3300 ICP can connect to the third-party PBX using a variety of methods, building the gateway to IP telephony. For customers, this model allows them to use features like IP Networking, Collaboration, Mobility, and virtual Contact Center applications.

The diagram below illustrates these various solutions.

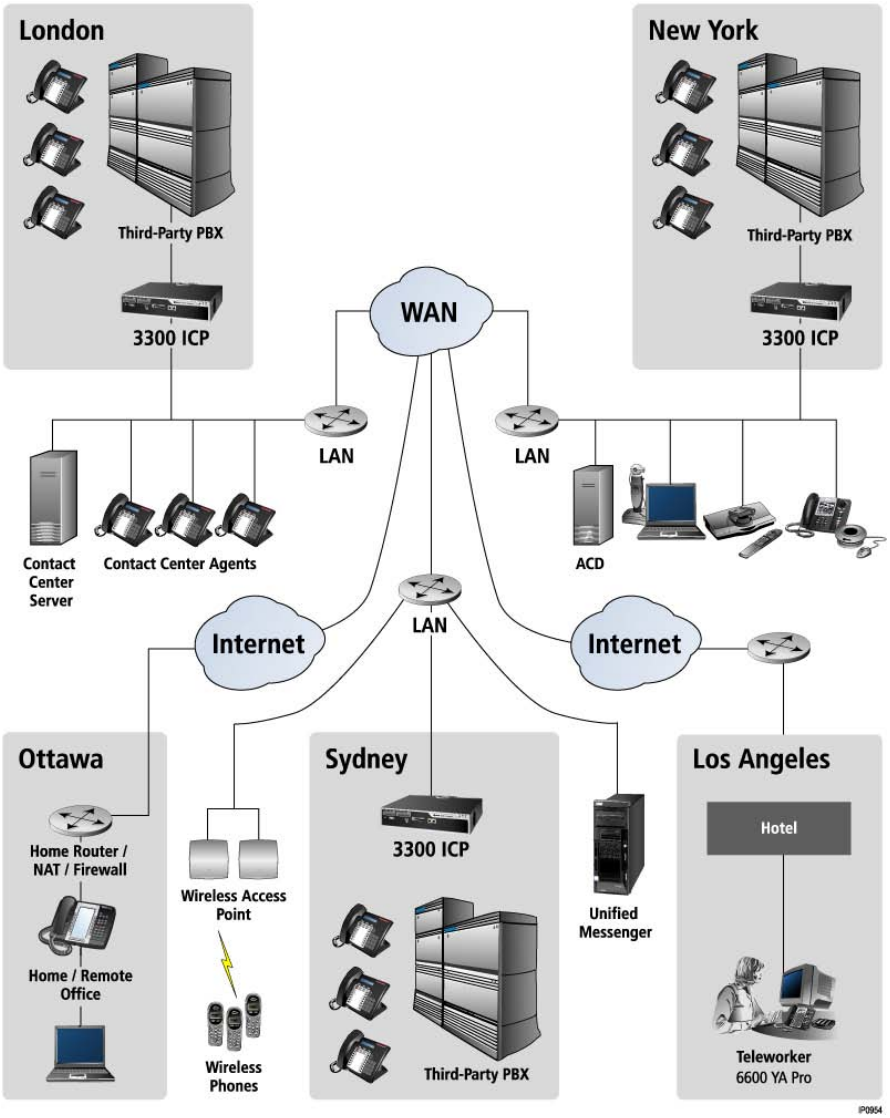


Figure 20: 3300 ICP as Gateway

Live Business Gateway

Mitel Live Business Gateway allows the Microsoft® Office Communications Server 2007 to communicate with a Mitel 3300 IP Communications Platform (ICP). The Live Business Gateway enables a user to place and receive calls from the Microsoft Office Communicator 2007 via a Mitel IP desktop phone. It also provides Communicator 2007 and the Microsoft® Office system with the telephony status of business users.

Note: Live Business Gateway continues to fully support Microsoft Live Communications Server 2005 and Communicator 2005.

Communications Server 2007 is an enterprise real-time communications server that supports convergence of real-time multi-modal communications. Office Communicator is an integrated enterprise communications client that integrates online presence, instant messaging, telephony conferencing and video at the desktop. Office Communicator integrates into Microsoft Office (Word, Excel, SharePoint). Through SIP/CSTA, the Live Business Gateway extends voice capabilities beyond the basic SIP point-to-point communications of Communications Server to enable the rich business telephony features and capabilities of the Mitel 3300 ICP.

The Live Business Gateway allows Office Communicator to take advantage of basic and supplementary phone services offered by the 3300 ICP. Using Communications Server 2007, Microsoft Office Communicator is able to control the phones connected to the 3300 ICP.

The Live Business Gateway:

- Allows users to dial contacts using Office Communicator and Office Suite – either by lifting the Mitel IP phone handset when the call is connected or by answering from a desktop with Office Communicator.
- Further extends Office Communicator and Office Suite with the online presence and availability status features of Online, Busy, DND, Be Right Back, and Appear Offline. It also includes telephone presence for In Call and DND, allowing corporate or personal contacts to quickly and easily know the preferable communication medium when immediacy is required.
- Allows desktop applications to easily use standard call control features such as placing, transferring, forwarding, and holding calls.
- Allows users to create up to an eight-party conference call without the need for a separate Media Conferencing Unit (MCU) by clicking from contacts through the Office Communicator GUI.
- Uses the industry standard SIP/CSTA protocol which removes the need to understand Mitel 3300 ICP call control API. The Mitel solution allows an Office Communicator client to extend past a basic 1:1 SIP call within the enterprise, thus enabling calls through the Mitel 3300.

The Live Business Gateway integrates into a typical 3300 ICP office environment as shown in the following figure. In this diagram, the call placed from Office Communicator signals the Live Communication Server, which then signals the Live Business Gateway. The Live Business Gateway conveys a message to the 3300 ICP that then rings the two phones. Once the call is connected, the 3300 ICP signals the Live Communications Server via the Live Business Gateway to tell all Communicator clients that those two parties are now in a call.

For more detailed information on Live Business Gateway, see See “Mitel Live Business Gateway” on page 60..

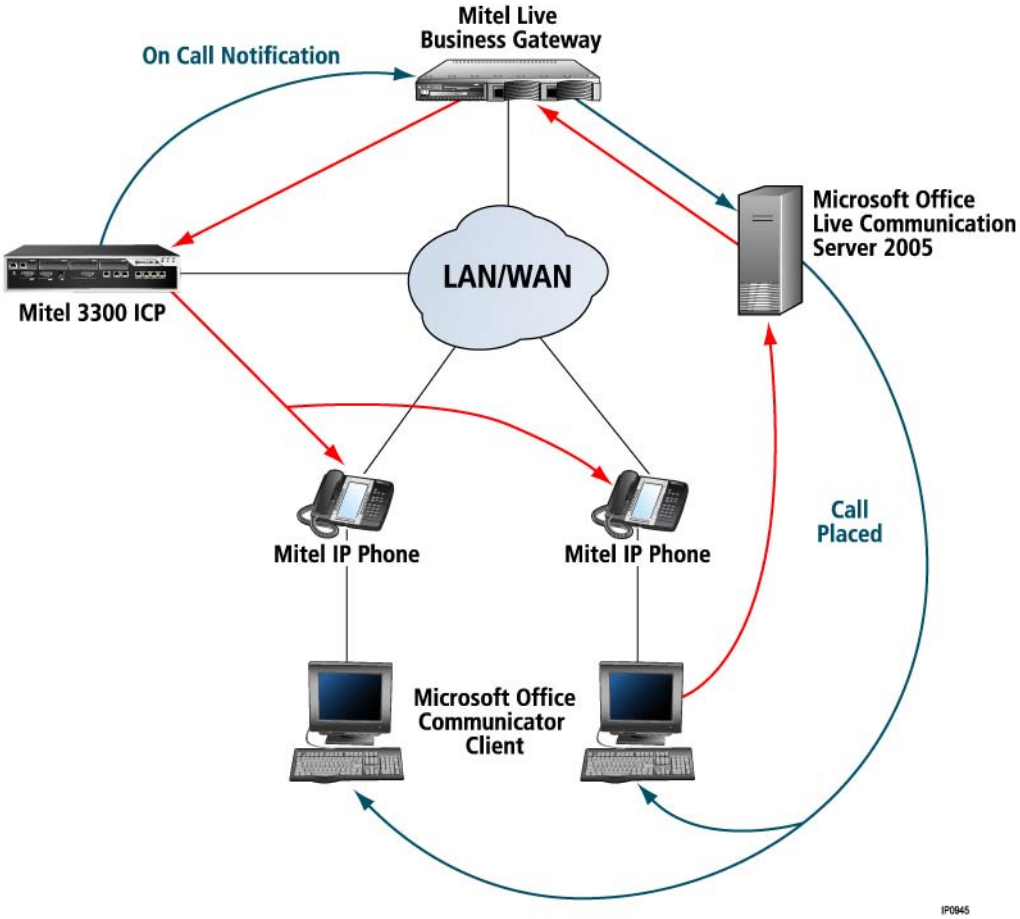


Figure 21: Live Business Gateway Solution

PC-to-Phone Support and the 3300 IP Communications Platform

PC-to-Phone SIP (Session Initiation Protocol) integration is supported between Microsoft® Office Live Communications Server (LCS) 2005 or Microsoft Office Communications Server (OCS) 2007 and the Mitel® 3300 Integrated IP Communications Platform (ICP). An end user can establish calls from a Microsoft Office Communicator client, acting as a basic SIP softphone, to or from a Mitel desk phone, the Public Switched Telephone Network (PSTN), or any other network-connected PBX. The term PC-to-Phone encompasses calls between PC to PSTN, Phone to PC, and PSTN to PC.

Typically, mobile employees use the PC-to-Phone solution with the embedded softphone capability provided by the Microsoft Office Communicator client on their laptops. From a Microsoft Office Communicator client, a user can dial through the Find field, through a Contact list, or through the Microsoft Office suite of products as defined by the Microsoft server specifications. All telephony services for the Communicator softphone are provided by the Live Communications Server 2005 or Office Communications Server 2007. There is no integration between the Communicator client and a Mitel desk phone.

PC-to-Phone provides Communicator users with access to the full range of features provided by the Office Communications Server (for example, Audio and Video Conferencing, Web Collaboration, and Simultaneous Ringing). Simultaneous Ringing is very similar to the Mitel Unified Communicator Mobile application. Simultaneous Ringing allows a user to program "twinned" numbers for his or her Communicator client. When the client is called, the twinned number also rings. Users can twin their Communicator clients with other devices such as their cell phones or home phones.

The following sections describe the supported integrations. A SIP trunk is required to support each concurrent call between the Communications Server and the 3300 ICP or PSTN.

Live Communications Server 2005

The following diagram illustrates a Live Communications Server 2005 integration:

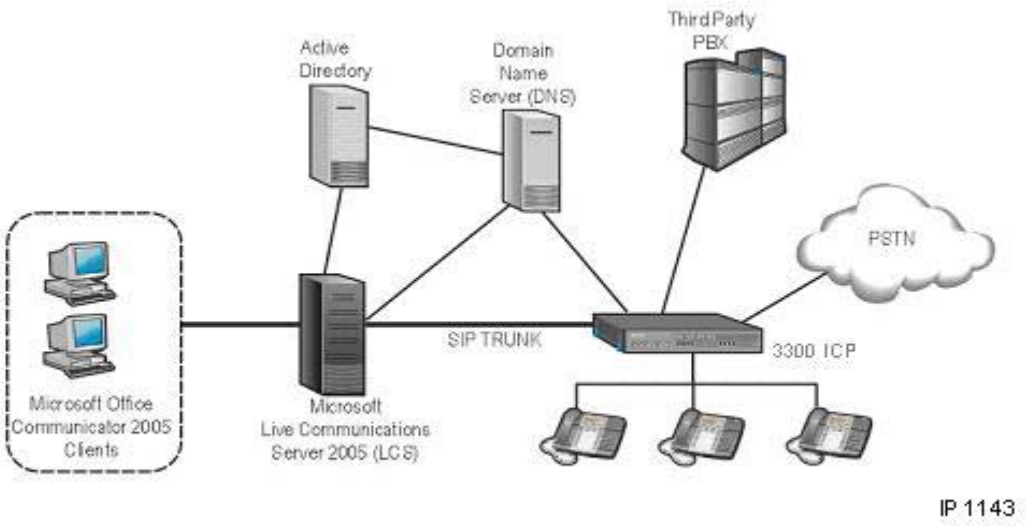


Figure 22: Live Communications Server 2005 Integration for PC-to-Phone Support

Office Communications Server 2007

In an Office Communications Server 2007 integration, the Communicator client operates as a SIP Phone and connects via SIP trunking through a Mediation Server to the 3300 ICP.

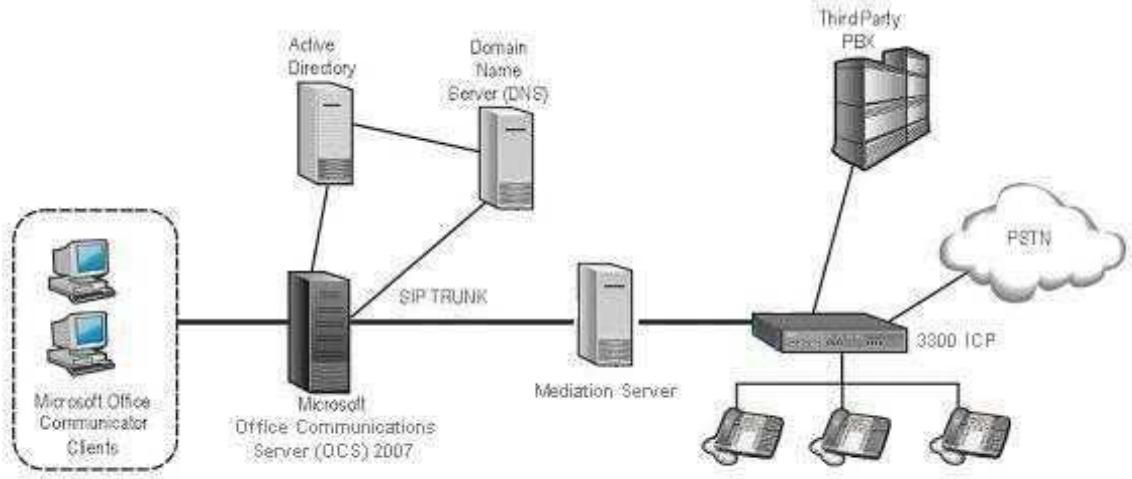


Figure 23: Office Communications Server 2007 Integration for PC-to-Phone Support

Migrating to the 3300 ICP

Mitel continually implements the latest technology advancements to increase the capacity, performance, and value of our products. However, at the same time, Mitel has followed the strategy of protecting our customers' communication investments by allowing them to transition their existing systems, whenever possible, to the latest Mitel solutions.

Customers can make a smooth transition from an existing SX-2000 LIGHT system to IP telephony without sacrificing the features and the investment of their existing equipment. A customer can make small incremental investments to move from the circuit-switched world to a full 3300 ICP while retaining their investment in customer premise equipment and desktop devices. IP telephony simplifies moves, adds, and changes of desktop devices. It also simplifies the building cabling because the IP telephone and the desktop PC can share the same Ethernet connection.

For both IP telephony and traditional circuit switched telephone users, migrating to IP telephony enables you to bypass long distance tolls. It allows you to route traditional PSTN calls across an unmetered IP network, reducing ongoing costs.

SX-2000 Adjunct

By integrating a 3300 ICP into an existing SX-2000 solution, our customer fully retains their previous investment in Mitel equipment.

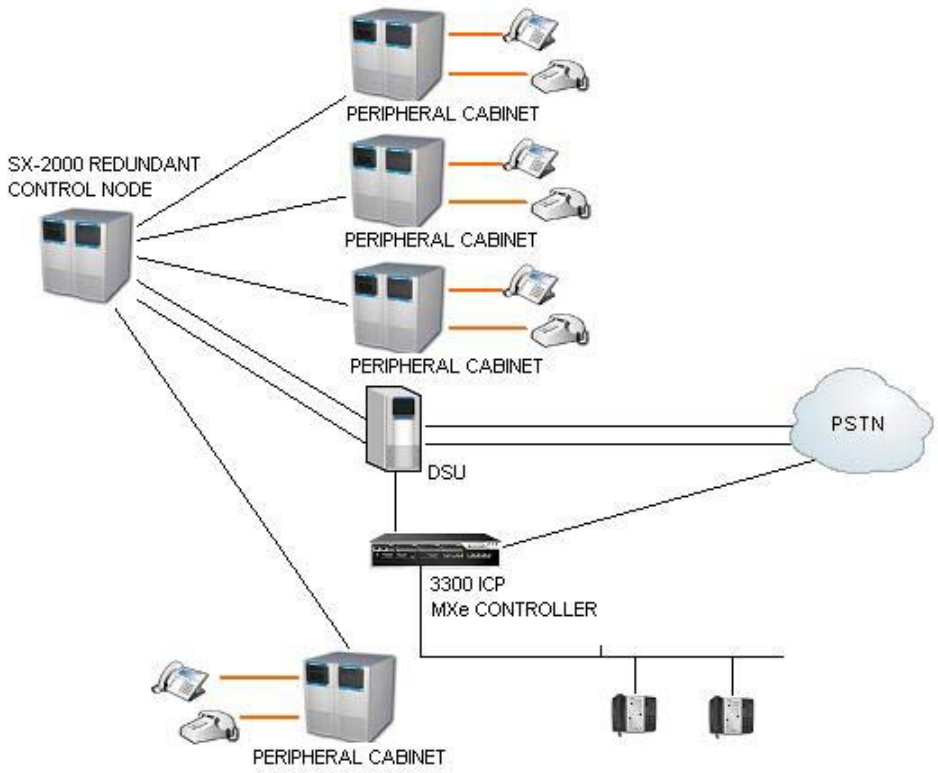


Figure 24: 3300 ICP Integrated with SX-2000 LIGHT

The 3300 ICP connects to the SX-2000 via a DPNSS (MSDN) connection. This connection allows many features to work transparently, and also delivers the customer a gateway to IP telephony.

SX-2000 Control Replacement

By replacing the SX-2000 control cabinet with the 3300 ICP Controller, the customer can retain their investment in desktop devices and connection equipment while allowing the benefit of IP telephony to be utilized immediately. This migration method has the advantage of retaining a single communication controller and hence reducing administration overheads.

Customers can migrate from an SX-2000 LIGHT or MicroLIGHT to a 3300 ICP. The Mitel Software Installer tool converts and restores an existing SX-2000 database to a 3300 ICP.

This replacement allows for conversion of a MicroLIGHT into another peripheral, with the ICP taking over as controller.

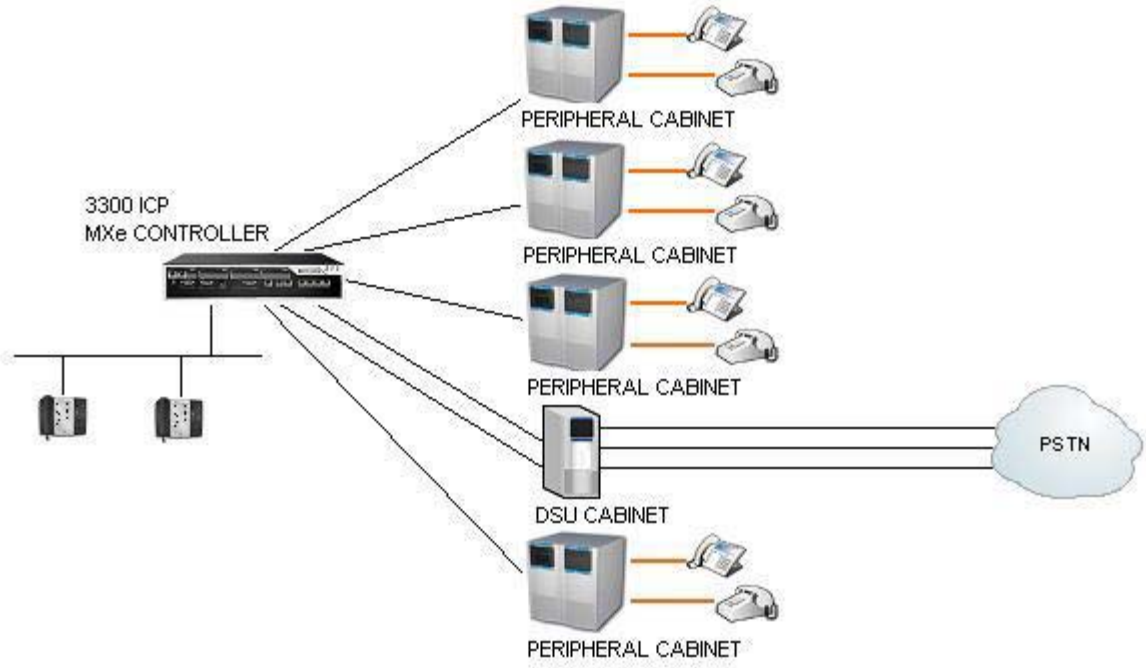


Figure 25: SX-2000 Control Replacement

SX-200 Control Replacement

In the same manner as with the SX-2000, by replacing the SX-200 control cabinet with the 3300 ICP Controller, the customer can retain the bays and desktop devices and connection equipment while allowing the benefit of IP telephony to be utilized immediately. All of the ONS, OPS, and DNI lines can be reused, but no trunks are supported on the bays. Only bays with the BCCIII controller are supported.

Non-Mitel Products

Non-Mitel PBX Adjunct

By integrating a 3300 ICP into an existing non-Mitel PBX, the customer retains their previous investment in communication equipment while being able to take advantage of the benefits of a superior IP telephony solution. The 3300 ICP can connect to the Non-Mitel PBX in a variety of methods (depending on which is available) including QSIG, DPNSS or SIP.

The 3300 ICP delivers the customer a gateway to IP telephony. This method is typically utilized when an end customer wants to take advantage of Mitel's many features and applications but does not want to completely replace their existing investment.

Applications

The following applications are available for all Mitel Communications Director deployments, except where noted. These applications include Unified Communications solutions, solutions for vertical markets, General Business Solutions and Third Party Developer Support.

- Unified Communications
  - “Mitel Unified Communicator” on page 51
  - “Mitel 5300 Intelligent Directory Application” on page 59
  - “Conferencing and Collaboration” on page 60
  - “Mobility Solutions” on page 63
  - “Messaging” on page 72
  - “Customer Interaction Solutions” on page 75
  - “Mitel Applications Suite” on page 79
- “Hospitality” on page 80
- “General Business Solutions” on page 82
- “Third-Party Developer Support” on page 85.

Some applications are embedded in the system software and others are supported externally. For more information, see the sections that follow. For detailed information on the Unified Communications solutions, see the *Unified Communications General Information Guide*.

Mitel Unified Communicator

Mitel Unified Communicator enhances business communication and collaboration with co-workers, customers and partners, helping employees to make better, faster decisions, be more responsive and deliver greater value to their clients.

Mitel Unified Communicator Express

Mitel Unified Communicator (UC) Express is a cost-effective, server-less desktop unified communications client that provides productivity enhancements like click-to-call, incoming caller ID pop-up, call history, speed call list, plus personal (Microsoft® Outlook®) and Corporate (Microsoft Active Directory) directory integration with public instant message presence engines. UC Express is tightly integrated with Mitel’s 5312, 5324, 5320, 5330, 5340 and 5360 IP Phones, resulting in a converged infrastructure to enhance the user experience and the effectiveness of "in the moment" communications.

UC Express is designed for easy configuration and installation, providing IT personnel with a number of implementation options – from simple end-user downloads to large-scale pull or pushed-based mass deployment.

UC Express is a fast and easy way to simplify routine communications and help users maximize their operational efficiency.

UC Express features include:

- Click to dial on Telephony Toolbar
- Incoming caller ID displayed in Telephony Toolbar with click to answer
- Calling line ID pop-up
- Access to speed dials list, missed calls list and re-dial numbers
- PC access to Corporate Directory and Personal Outlook Contacts
- Context sensitive telephony functions on Telephony Toolbar (Mute, Hold, Retrieve, Cancel)
- Phone Button Programming Interface via intuitive Program Personal Key dialog
- Search and dial from Outlook contacts
- Presence integration (5312, 5325, 5320, 5330, 5340 and 5360 IP Phones)
- Multiple phone profiles (e.g. office profile and home office profile)
- Presence indication for speed dials on 5320, 5330, 5340, and 5360 IP Phone displays
- Alternate Number Dialing capability when IM Presence is Offline or Busy
- Auto IM presence state change based on call state (e.g., switches to "on the phone" while on a call)
- Search and dial from corporate contacts (Microsoft Active Directory integration)
- Open caller's contact record on incoming call
- Create new task with caller ID, date / time auto-filled
- Create new email to caller with 'send to' address and subject auto-filled
- Search desktop (using Microsoft Windows Desktop Search) for emails, documents, contact records, etc., containing the caller's name or number
- Create new tasks/e-mails, or even conduct a desktop search, with CallerID supplied information already filled into fields
- Click to dial highlighted numbers in emails, web pages, documents, contact managers
- Pause music on incoming calls - music being played on Windows Media® Player or iTunes® will automatically pause for an incoming or outgoing call.

Mitel Unified Communicator Advanced

Mitel Unified Communicator (UC) Advanced is a comprehensive unified communications client that integrates presence and availability, secure Instant Messaging (IM), audio conferencing, and video and data collaboration with the call control capabilities of MCD. UC Advanced provides a unique "launchpad" for commonly used Mitel and third-party applications, and an open API to enable tailored integration into business process software such as salesforce.com and Microsoft CRM.

UC Advanced enables users to manage contact information, determine the presence and availability of colleagues, and set their own call-handling policies at the desktop. With the UC Advanced Softphone, it becomes a feature-rich, fully functional softphone that can provide full

access to communications and collaboration from any location with a high speed internet connection – enabling those on the road or working from home to communicate and collaborate as if at the office. Fully modular, UC Advanced can support voice only; voice, video and data; and multi-media simultaneously, as required by the collaborative requirements of users and organizations.

UC Advanced features include:

Personalization

Users can personalize their UC Advanced interfaces by selecting from a wide range of color schemes and by choosing between the traditional “metal” or new “glass” appearance.

Simplified Call Management

The UC Advanced desktop control panel offers intuitive visual point and click access to the call management features of MCD. Conference calls can be managed while in progress by dragging and dropping the name of a participant into the conference (see “Directory Integration” on page 53). UC Advanced remembers the most frequently dialed numbers and makes them easily accessible from a centralized drop-down menu. The UC Advanced interface provides the following benefits:

- Access to any number of UC Advanced users across any number of servers
- A simple interface to initiate calls, transfer calls, and setup conferences
  - The interface is so simple to use that it gives the power to control the technology back to the user
- Ability to drag others into the call to create a conference
- Highlighted buttons during an active call present the user with options they may have never even thought were available.

Directory Integration

UC Advanced supports Corporate Directory for MCD and LDAP database interface. If the user selects the Corporate Directory option, their Corporate Contacts list is populated with data from the MCD telephone directory. If the user selects the LDAP database interface option, it enables integration with additional Personal Information Managers (PIMs) and databases that support LDAP. The LDAP interface utility within UC Advanced maps the data fields in the external database to fields within UC Advanced.

UC Advanced provides synchronization of contact data between UC Advanced and Microsoft® Outlook, IBM® Lotus Notes, and ACT! for contacts that are imported into the UC Advanced Personal Contacts list. Synchronization flows from Outlook to UC Advanced only. Therefore, it is best to update the PIM so that changes are then synchronized reflected in UC advanced.

Web Window

A smaller browser window is provided as a shutter within the UC Advanced 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 the company. For example, an RSS (Really Simple Syndication) feed can be displayed, or a URL link to a complementary application.

Caller Line ID-based Routing

This feature allows users 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 or to a administrative assistant. Benefits include:

- Provides call management and flexibility
- Sets up automatic call handling policies
- Forwards calls to your voice mail, cell phone, or any other phone number
- Stores and changes call handling policies from one profile to the next
- Business Advantages:
  - Always reachable by the people you specify
  - Empowers users to make full use of phone functionality.

Data and Telephony Presence and Availability

UC Advanced maximizes successful communication by indicating if people are on the phone, away from their desk, or available for a secure instant chat or data collaboration. The People Window provides:

- the state of the user's phone – available (on hook), busy (off hook)
- status advisory messages – In a Meeting, At my Desk, Out of the Office
- PC presence for secure IM – online/away/busy/offline. There is also integration with Outlook Calendar.

Corporate Secure Incorporated Instant Messaging

IM and file sharing features offer security as well as cohesive teamwork. Users have access to any number of UC Advanced users across any number of servers. Users can initiate private and secure real-time chat and drag and drop any number of individuals into a conversation. Documents can be shared by dragging and dropping files into chat sessions. Users can also view a history of the chat session.

The business advantages of secure chat and file sharing include:

- Less spam, more privacy
- Instantly send messages to colleagues
- Share large files directly by dragging the file into the chat window
- Archive and make easily accessible all of the messages exchanged

Versatile Call Forward Options

This feature allows users to set up multiple call forward profiles. UC Advanced also supports real-time call forwarding to other extensions, external phone numbers and voice mail via a simple interface.

Knowledge Management

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.

- Extends the benefits of screen pops to all knowledge workers
- When an incoming call is received, UC Advanced presents the user with a list of all known information about the caller
- Resources that can be presented include emails, contact entries, and documents
- Document types include Microsoft® Word, Microsoft Excel®, Microsoft PowerPoint®, Microsoft Outlook® and Adobe Acrobat PDF documents.

Communications-Enabled Business Process Integration

UC Advanced application programming interface (API) – UC Advanced API enables users or Mitel Professional Services to integrate the functionality of UC Advanced into an organization's in-house applications such as CRMs. UC Advanced provides a simple API that enables incoming and outgoing calls and dialling events to be associated with lookups in other applications, for example launching a Microsoft CRM contact record based on the caller line ID.

Enhanced Lotus Notes Integration

UC Advanced provides tight integration with Lotus Notes 8.0. Lotus Notes users can select a contact from the Notes contacts list and dial directly using UC Advanced. In addition to the basic dial facility, UC Advanced enables a user to select a contact from the Notes address book and easily establish a web collaboration session. The web collaboration session includes the ability to access video conferencing capabilities using USB web cams.

Enhanced Microsoft Office integration

Smart tag support makes it easier for Microsoft users to dial from within a Microsoft application such as Microsoft Outlook or Microsoft Office Suite (Word, Excel, PowerPoint etc.) using UC Advanced to establish the outgoing call. The user simply selects a name from within the Microsoft application and then selects "UC Advanced" followed by the "Dial user" option.

UC Advanced also supports dialing from Internet Explorer. The number should be identified as dialable on a hover over; clicking on the hover over will then dial the number via UC Advanced. As well, UC Advanced enables a user to select "Outlook calendar" as their advisory message. If this option is selected, the advisory message will be based on the user's Outlook calendar status e.g. "busy until 5pm"; "free until 2pm" The message will be updated as the Outlook

calendar status changes. The calendar integration option in the advisory selection is not shown if Outlook is not present on the PC.

Innovative “Launchpad” Shutter for Unified Communications

The Launchpad is a configurable shutter that a user can set up with the following clickable items:

- Individual contacts can be called with a single mouse click, including creating speed dials that will quickly navigate voicemail and conferencing service menus.
- Launch URLs to frequently accessed websites and web-based applications such as Unified Communicator Mobile or Audio and Web Conferencing.
- Launch frequently used applications such as Microsoft Office SharePoint or Outlook.
- Create shortcuts to frequently browsed folders or shared drives.

The user is able to create, edit, remove and re-position the buttons.

Federated Servers

UC Advanced servers in multiple locations can share IM and presence information between servers.These federated servers allow UC Advanced users in one office to view the presence and availability of UC Advanced users in another office in the same network.

Centralized Call Logging

The UC Advanced server can log incoming calls for UC Advanced clients while the client software is not running. When UC Advanced is started, the UC Advanced 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.

Unified Communicator Mobile Integration

UC Advanced supports integration with Unified Communicator (UC) Mobile. A UC Advanced user who also has UC Mobile enabled can answer an incoming call directed to their desk phone on their mobile device of choice. When the call is answered on the mobile device, UC Advanced is able to change the telephony presence status to “Off Hook”. This enables UC Advanced to display the correct telephony status for a user regardless of whether the call was answered on a user’s desk phone, softphone or mobile device.

Configuration Tool for USB Handsets and Headsets (UC Advanced Softphone required)

UC Advanced with the optional UC Advanced Softphone enables customers to create, answer and end calls via USB devices. These devices include handsets and headsets, as well as VoIP handsets. UC Advanced will provide USB support for devices that follow the USB HID standards, although support is specific to devices that do not require a custom integration to function. For additional details see “Mitel Unified Communicator Advanced Softphone” on page 57. The following USB headsets have been certified for use with UC Advanced Softphone:

- GN Netcom 8120
- Plantronics CS50 / CS60

- Voyageur 510 – USB Bluetooth
- VoIPvoice Cyberphone 654 USB handset

An intuitive configuration tool allows a user to configure their USB HID compliant headset or handset to their preferred configuration. This enables users to make, receive or manage calls using their preferred USB device while away from the PC that is running UC Advanced.

Key Line Appearance

UC Advanced has support for key line appearances. When a key line appearance is in use, all other users that have an appearance of this key are presented to them with a new telephony status icon indicating that this key line appearance is in use. UC Advanced also supports the barge-in feature on any key line extension that has privacy disabled. This is accessible by the communication window, communication shutter and tray icon. This feature allows UC Advanced to be used as a “Mini Console”, a very beneficial capability for small and medium sized businesses.

Consolidated Corporate Directory Across Multiple UC Advanced Servers

UC Advanced displays directories found on multiple UC Advanced servers as a single corporate directory.

Software Compatibility

- UC Advanced supports Windows 2008 server
- UC Advanced is supported in a VMware ESX Server 3.0.1 environment
- UC Advanced supports ACT! 2008 and Lotus Notes 8.0

Mitel Unified Communicator Advanced Softphone

Mitel Unified Communicator Softphone is an optional ad-on software module that provides telephone functionality including:

- PC Softphone – An IP-based software telephone running on a PC or laptop. This allows the remote user connected to the corporate network via a high speed internet connection to appear as though they are at their desk
- Record calls – Allows users to record calls and save them to their PC
- Feature-rich fully functional telephone
- All phone calls, prompts, and features are available from the laptop and the desktop phone simultaneously
- Access from any location with a high-speed connection
- Communicate and collaborate from anywhere
- Integrates with MBG teleworker service

Log into UC Advanced from Different PCs

UC Advanced enables softphone users who are logged in at one PC to move to another desk/station and log in again. UC Advanced then logs out the original session. When a user returns to their original desk/station a “log in” option is provided in the UC Advanced interface to enable the user to log back into UC Advanced at their original station. This log in function automatically logs out any other active UC Advanced sessions for that user. This new enhancement prevents multiple log-ins for a single user while providing a user with the flexibility of logging into a desk/station in another part of the building without having to first log out of their original session.

Customized Ring Tones

UC Advanced allows a UC Advanced Softphone user to change the ring tone for each line. This is an important feature where several UC Advanced Softphone users reside within the same office, and users require distinctive ringing on their PCs. Ring tones in the form of .wav files can be simply imported into the custom ring tone application. Users are able to add, remove, play and select ringtones.

Mitel Your Assistant Collaboration Option for UC Advanced

The Your Assistant Video/Data Collaboration module removes the complexity and cost of a hosted Web conferencing solution and ties the UC Advanced voice connectivity with full Web conferencing and collaboration capabilities. With this option, any user within the organization that has UC Advanced can create on-the-fly collaboration and conferencing sessions with their colleagues within the organization, or with anyone in the public wherever they might be. Users enjoy the full benefit of productivity enhancing, multi-party, multi-location, full collaboration and Web conferencing without paying for hosted services.

The Your Assistant Collaboration Option for UC Advanced provides

- Full desktop, browser, application, remote control, and video functionality
- Up to 20 videos in a single session
- Ability to place a call to a UC Advanced user and click "Start Web Conference"
- Capability to setup a session with non-UC Advanced users by simply scheduling the conference and sharing the link
- Desktop and application sharing

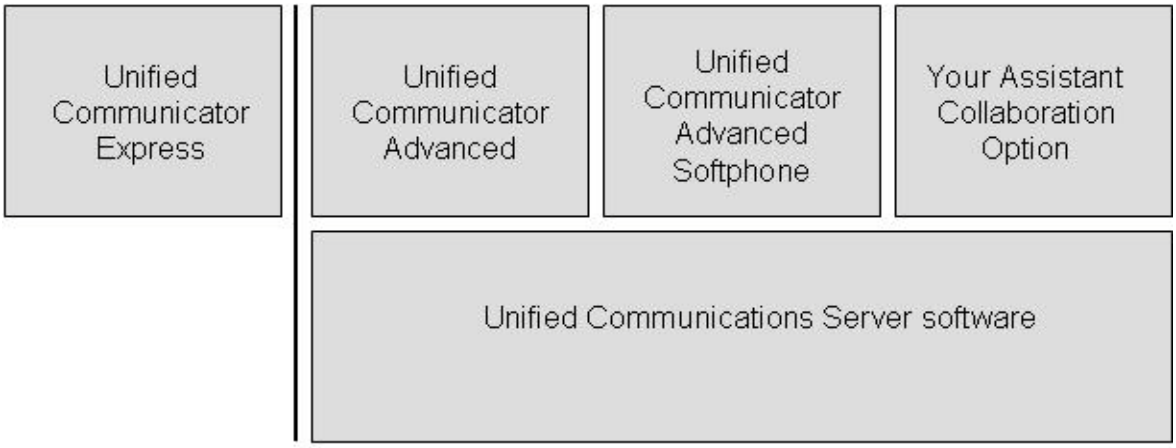


Figure 26: Unified Communicator Applications

Mitel 5300 Intelligent Directory Application

The 5300 Intelligent Directory application provides a simple, intuitive on-phone, searchable directory of both corporate (Microsoft Active Directory®) and personal contacts (Microsoft Outlook®). It allows users to search for contacts using their 5320, 5330, 5340 or 5360 IP Phone's familiar keypad and search the way that they are accustomed to using their cell phone and other handheld devices. The 5300 Intelligent Directory home screen can also be customized with a list of "favorites" selected from the corporate directory (Microsoft Active Directory) or a personal contacts list (Microsoft Outlook).

Features include:

- Extremely intuitive and easy to use – phone numbers are where you need them, on your phone.
- Instant phone number updates. When a new person is added to the centralized Microsoft Active Directory directory, their phone numbers are automatically available to all 5300 Intelligent Directory users.
- Less administration, less expense, more accuracy. No more need to separately maintain corporate phone books for internal use.
- Up to five phone numbers can be displayed per person (corporate, cell, home, etc.).
- One-touch dialing of numbers.
- Dynamic list of recent calls or searches displayed automatically on the default home screen.
- Addition of your favorite phone numbers to your home screen.
- Support for hot desking: 5300 Intelligent Directory requires users to enter their password to access personal Outlook contacts in Microsoft Exchange Server.
- On-phone access to corporate Active Directory directory.

- On-phone access to personal Outlook Contacts stored in Microsoft Exchange Server.

Mitel 5300 Intelligent Directory Presence Option

The 5300 Intelligent Directory Presence Option is an add-on software option to the 5300 Intelligent Directory application and provides at-a-glance presence information for the entire corporate directory list. Presence information is automatically fed from the Instant Messaging (IM) contact list (Microsoft Office Live Communications Server® 2005 or 2007) to the 5320, 5330, 5340, and 5360 IP Phone display. Presence icons that appear on the 5320, 5330, 5340, or 5360 IP Phone display provide presence indication for all corporate contacts.

Conferencing and Collaboration

Mitel's Conferencing and Collaboration portfolio includes:

- "Mitel Live Business Gateway" on page 60
- "Audio and Web Conferencing" on page 62

Mitel Live Business Gateway

Mitel Live Business Gateway allows the Microsoft® Office Communications Server 2007 to communicate with an MCD host platform. The Live Business Gateway enables a user to place and receive calls from the Microsoft Office Communicator 2007 via a Mitel IP desktop phone. It also provides Communicator 2007 and the Microsoft® Office system with the telephony status of business users.

The combination of Office Communications Server 2007 and the Mitel Live Business Gateway enhances information worker productivity and greatly improves business process efficiencies by combining a wide range of collaboration tools with Mitel's trusted IP telephony solution.

Note: Live Business Gateway continues to fully support Microsoft Live Communications Server 2005 and Communicator 2005; however, reference will only be made to Office Communications Server 2007 and Communicator 2007 in this document.

This integrated solution from Mitel and Microsoft includes:

- Microsoft Office Communicator 2007
- Microsoft Office Communications Server 2007
- Mitel Live Business Gateway 3.1
- Mitel 3300 IP Communications Platform (ICP)

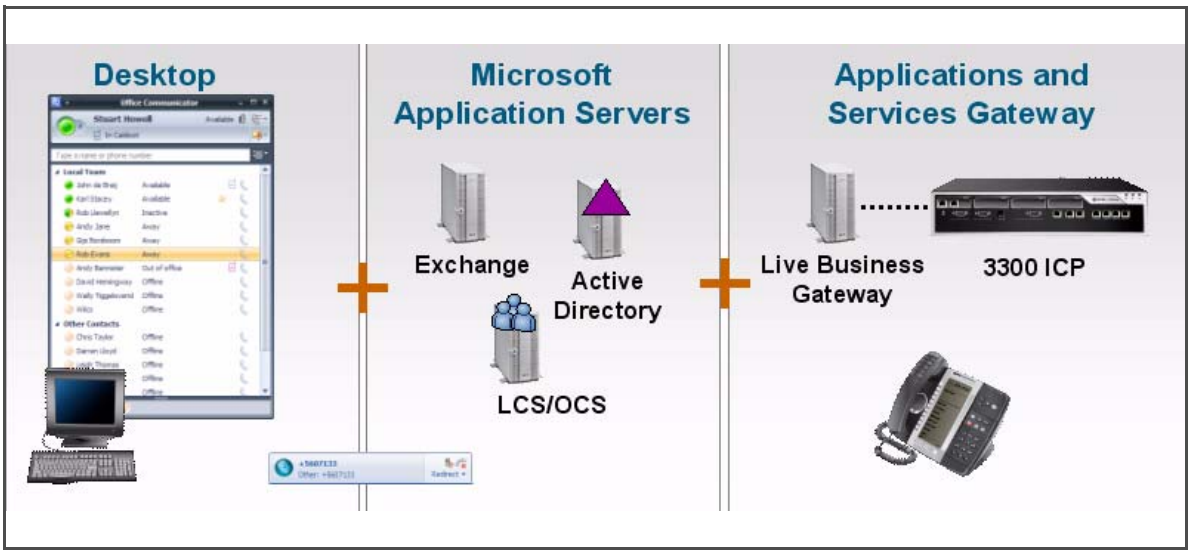


Figure 27: Live Business Gateway Integration

Voice Integration

The Live Business Gateway provides voice integration between the MCD host platform and Office Communications Server 2007. Communicator 2007 users are able to make calls via a Mitel IP desktop phone to any other internal or external telephone number.

Presence and Availability

Communicator 2007 users are able to get a visual indication on the status of other users on the network, including a user's telephone status. If a colleague or team member is busy on a call, the contact can be "tagged" so that when the user becomes free, a pop-up notification is provided to alert that this contact is now available.

Office Communications Server 2007 Enhancements

Live Business Gateway Release 3.1 users are able to take advantage of many of the non-telephony feature enhancements available with Communicator 2007 and Office Communications Server 2007. These enhancements include new presence states, presence management policy and call history, plus a range of contact management improvements. Full details on the new feature enhancements of Communicator 2007 and Office Communications Server 2007 can be found on the Microsoft website.

Integration with Microsoft Outlook and Microsoft Office System

Incoming calls to the Communicator 2007 user that are not answered will be flagged as a missed call in the user's Microsoft Outlook inbox. In addition, a conversation folder within Outlook tracks all voice and IM conversations. A user's telephony presence can be seen from within Outlook or the Microsoft Office system. Calls can be established directly from within these applications through the click of a mouse. Out of office messages and calendar information can be synchronized with Communicator 2007, enabling other users to get detailed information about a particular user's availability and whereabouts.

Flexible Deployment Options

Live Business Gateway is supported on either a Microsoft Window operating system or Mitel Standard Linux. Mitel Standard Linux is a free of charge Linux®-based operating system that is becoming the operating system of choice for Mitel applications. Live Business Gateway is supported on a wide range of commercially available dual processor servers.

All licenses are controlled and managed via the Mitel Applications Management Center (AMC), simplifying installation procedures and providing a cost-effective method to add additional licenses.

Support for Mitel Application Processor Card

The Mitel Application Processor Card (APC) is a CPU on a compact PCI card. The Mitel Standard Linux operating system and Live Business Gateway software can be deployed from an APC installed in a Mitel 3300 Controller.

Audio and Web Conferencing

Audio and Web Conferencing (AWC) provides a feature-rich, cost effective IP-based collaboration solution for conducting highly interactive online meetings, brainstorming and training sessions, and presentations. Its audio conferencing and web presentations capabilities facilitate better collaboration among internal and external employees and business partners. Audio and Web Conferencing key benefits are:

- **Better Communications:** High quality audio and video let people interact easily and effectively, no matter where they are located.
- **Faster Business Decisions:** Arrange meetings instantly to bring the right people together at the right time.
- **Easy Scheduling:** The simple, Web-based interface lets you send e-mail invitations with access codes, dial-in numbers, Web links and all the details participants need for effective meetings.
- **Lower Costs:** Reduce costly and inefficient travel, while avoiding the high costs of out-sourced conferencing services.
- **Easy Management:** Deployed as part of the Mitel Applications Suite.

Audio and Web Conferencing provides the following key features:

- Audio Conferencing features:
  - Conference Session Recording
  - Integrated Reporting Capabilities
  - Port Reservation
  - Outlook Integration
  - Browser-based User Interface
  - Conference Scheduling
  - Conference Management

  - Record and Playback
  - Controlling a Call in Progress
  - Ad-hoc Conference Calling
  - Spoken Name/Roll Call
  - International Callback
- Web Conferencing Features:
  - Desktop Sharing
  - Application Sharing
  - Internet Co-browse
  - Multi-Point Video Conferencing
  - Polling
  - Security
  - Hand Raising
  - Acknowledgements/Quick Polls

Mobility Solutions

Mitel Mobility Solutions include the following:

- “Hot Desking” on page 63
- “Mitel Border Gateway (MBG) Teleworker Service” on page 64
- “Unified Communicator Mobile” on page 66
- “Wireless Support (3300 ICP deployments only)” on page 67 (available on 3300 ICP deployments only)

Hot Desking

Hot Desking creates a more flexible work environment by providing users with the ability to share IP phones, making it ideal for businesses employing telecommuters, sales agents, and other employees who spend only part of their time in the office. Hot Desking allows a pool of shared phones to be made available to employees instead of requiring each employee to be assigned a dedicated phone.

When a user logs in to a hot desk set, the system applies the user's phone profile – phone settings such as directory number, COS/COR settings, display preferences such as language, line appearances and button programming – to the set. Once logged in, Hot Desk users can use or change the phone features associated with their profile; for example:

- Call forwarding (all types)
- Callback messages (message waiting indicator)
- Auto Answer
- Do Not Disturb (DND)
- Last Number Redial

- Timed Reminder
- Advisory Status Message.

Hot Desking is supported across clustered networks so users can log on to any Hot Desk-enabled set in the cluster. Upon successful login, the set is redirected to the user's host ICP. Figure 28 shows an example of a hot desk user in a cluster.

External Hot Desking

Hot Desking is also supported on external answering points such as a cellular telephone, a home phone, or even a remote telephone using a VoIP service. Once their user number is programmed to support External Hot Desking, users' calls will be routed to their External Hot Desk telephone number.

Note: An External Hot Desking license is required for each Hot Desk user configured with an external answering point.

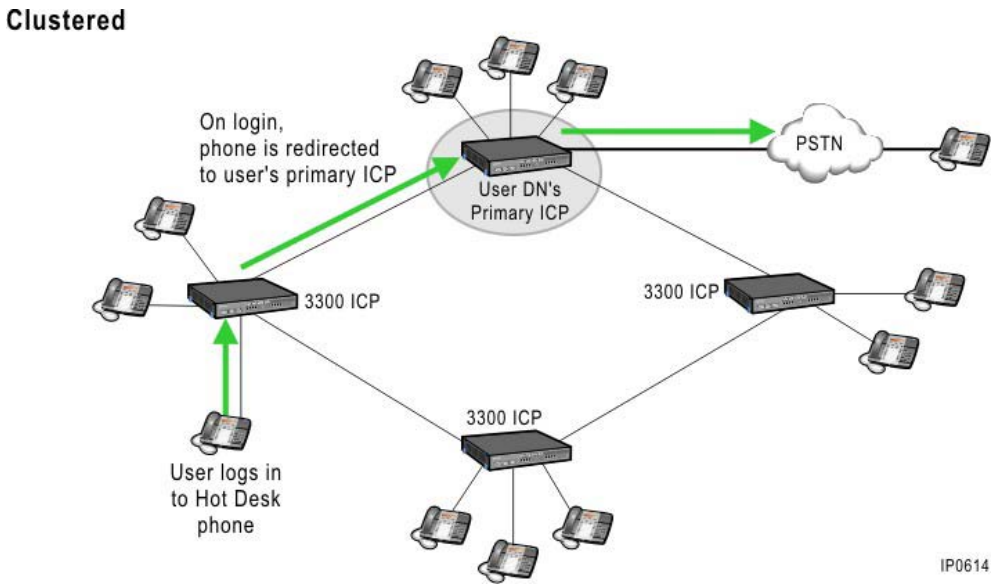


Figure 28: Clustered Hot Desking

Hot Desking can also be resilient. If both the set and user are programmed for resiliency, then the hot desk user does not lose service if the hosting controller fails. Instead, the hot desk phone registers for call service with the secondary controller and the user remains logged in with the current profile.

Mitel Border Gateway (MBG) Teleworker Service

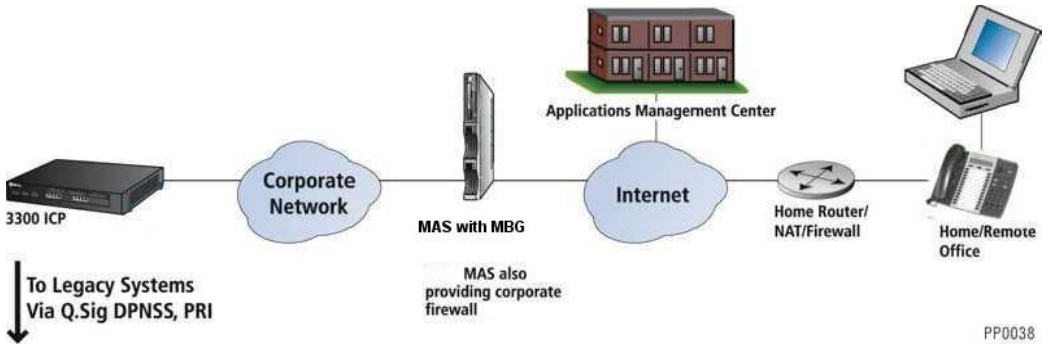
The MBG teleworker service connects a remote office to the corporate voice network to provide full access to voice mail, conferencing, and all the other features of the office phone system.

It requires the following:

Head Office	Remote Site
• Server installed with Mitel Applications Suite software and the MBG software blade or Server installed with Mitel Standard Linux software and the MBG software blade • Static IP address • Sufficient internet bandwidth (approximately 50 kbps is required per teleworker if G.729a compression is enabled)	• 5212, 5224, 5304, 5312, 5320, 5324, 5330, 5340 and 5360 IP Phones • DSL/cable router with Network Address Translation (NAT) and local DHCP • Broadband connectivity (static IP address is not required)

The MBG teleworker service can be completely configured at the head office using the 5212, 5224, 5304, 5312, 5320, 5324, 5330, 5340, or 5360 IP Phones. Using a two-click process, the phone is set to operate in teleworker mode. The telephone keypad is used to enter the IP address of the MBG installed at the head office. The phone can then be taken off-site and plugged into any broadband Internet connection. When the phone is powered up, it automatically establishes a connection with the MBG and is registered as a standard extension of the office phone system. The phone can also be returned to normal (non-teleworker) mode with the touch of a button.

The following illustrates possible MBG teleworker service configurations. In these configurations, the Applications Management Center (AMC) provides a range of downloadable applications and services to the remote office.



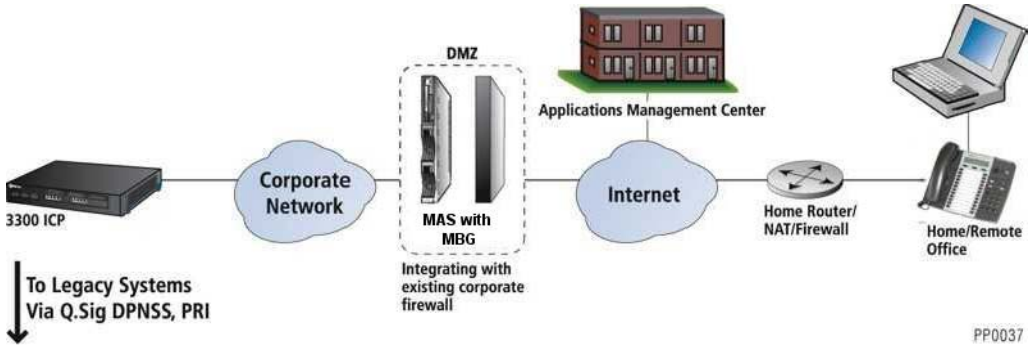


Figure 29: MBG Teleworker Service

Unified Communicator Mobile

Mitel Unified Communication Mobile (UC Mobile) is a software solution that lets users twin their desk phone with an internal or external PSTN-connected phone (a cell phone, for example). Calls arriving at the desk phone ring the cell phone simultaneously, until one or the other is answered, or if unanswered, are forwarded to voice mail.

UC Mobile allows users to extend commonly-used PBX features, such as hold and transfer, to their cell phone. This allows an enterprise's increasing mobile workforce to have access to Mitel's rich telephony features and applications while they are on the move. When configured as a twin, the cell phone acts as an extension of the enterprise desk phone, providing a single number contact and an integrated mobile and desktop experience.

The user also has the ability to easily change the device/number that is twinned to their primary extension and take advantage of the advanced capabilities delivered with the Mobile Clients for Windows Mobile and Symbian S60 smartphones. With the addition of the Mobile Client, outbound mobile calls are placed by the PBX, extending Single Number capability and delivering cellular long distance cost savings.

Administrators configure system settings using an administrative web interface. Users program their personal settings using a web interface, telephony user interface (TUI), or Mobile Client graphical user interface (GUI). For more information, refer to the Unified Communicator Mobile documentation available on the Mitel edocs website.

UC Mobile is available as a Standalone application or within Mitel Application Suite; UC Mobile release 2.0 and above will only be available within the Mitel Applications Suite (MAS).

Wireless Support (3300 ICP deployments only)

Mitel offers a choice of full-featured, integrated wireless IP solutions to suit your application, geographic location and technology preference. From DECT and Wi-Fi/802.11 solutions to Cordless Handsets and Cordless Headsets for the Mitel 5330, 5340, and 5360 IP Phones, Mitel's suite of wireless IP phone devices offer give users access to the complete range of MCD features.

MCD supports the following wireless devices:

Wireless phone	Wireless infrastructure	Comments
IP DECT OP27	DECT	EMEA and Australia availability only
NetLink i640 and e340/h340	Wi-Fi 802.11b	Direct IP integration support
Mitel 5602, 5603, 5604, 5606, 5606 Services and 5606 Alarm	IP-DECT	Available globally; integrated over SIP
Mitel 5610 Handset and IP DECT Stand	IP-DECT	Available globally; integrated over SIP

Additionally, for organizations who do not have a wired LAN, the Mitel Wireless LAN (WLAN) Stand is an accessory for the 5200 / 5300 series IP Phones that enables the phones to operate in a wireless LAN (WLAN) environment. When attached to a 5212, 5224, 5312, 5320, 5324, 5330, 5340, or 5360 IP Phone, the stand provides an 802.11b/g wireless interface.

Both the WLAN stand and the Cordless Handset/Headset are described later in this document.

IP Wireless phones offer the following benefits:

- **Integrated full-featured Call Control** – Includes caller name and number display, call hold and transfer, message waiting light and conference calls. Wireless softkeys provide users with single-button access to common telephony features such as call hold, call transfer, call waiting, call forwarding, call swap, multi-language support, voice mail control and Superkey functionality.
- **Complete IP Network integration** – When integrated with an MCD system, the phone offers the many features available in MCD.

IP-DECT System (Global)

The Mitel IP-DECT System (Global) is available globally for deployment where operation of devices in compliance with the European DECT or the North American DECT standards are permitted.

It consists of the following components:

- 3300 ICP Controller

- Base stations for wireless coverage
- 5602 – wireless handset for office environments
- 5603 – wireless handset for office environments
- 5604 – wireless handset for healthcare environments
- 5606 – wireless handset for healthcare and industrial environments
- 5606 (Alarm) – wireless handset for industrial/security applications
- 5606 (Services) – wireless handset for safety/security applications
- Services and Messaging gateway (WSM)
- Full Range of Accessories

The base stations connect to the 3300 ICP controller through the LAN and communicate to the 3300 ICP over the IP SIP protocol; the base stations communicate over the air to the Wireless Handsets using standard Digital Enhanced Cordless Telecommunications (DECT) protocol. One of the base stations functions as the Master Base station providing the management interface that enables configuration of the wireless system settings, base stations and handsets. The WSM connects to the system through the LAN and enables the Messaging and Alarm capabilities of the system. The handsets are programmed on the 3300 ICP as regular SIP extensions.

Each base station supports 8 simultaneous calls. The system supports up to 2,000 users and 1,000 base stations. See the diagram below for a typical configuration.

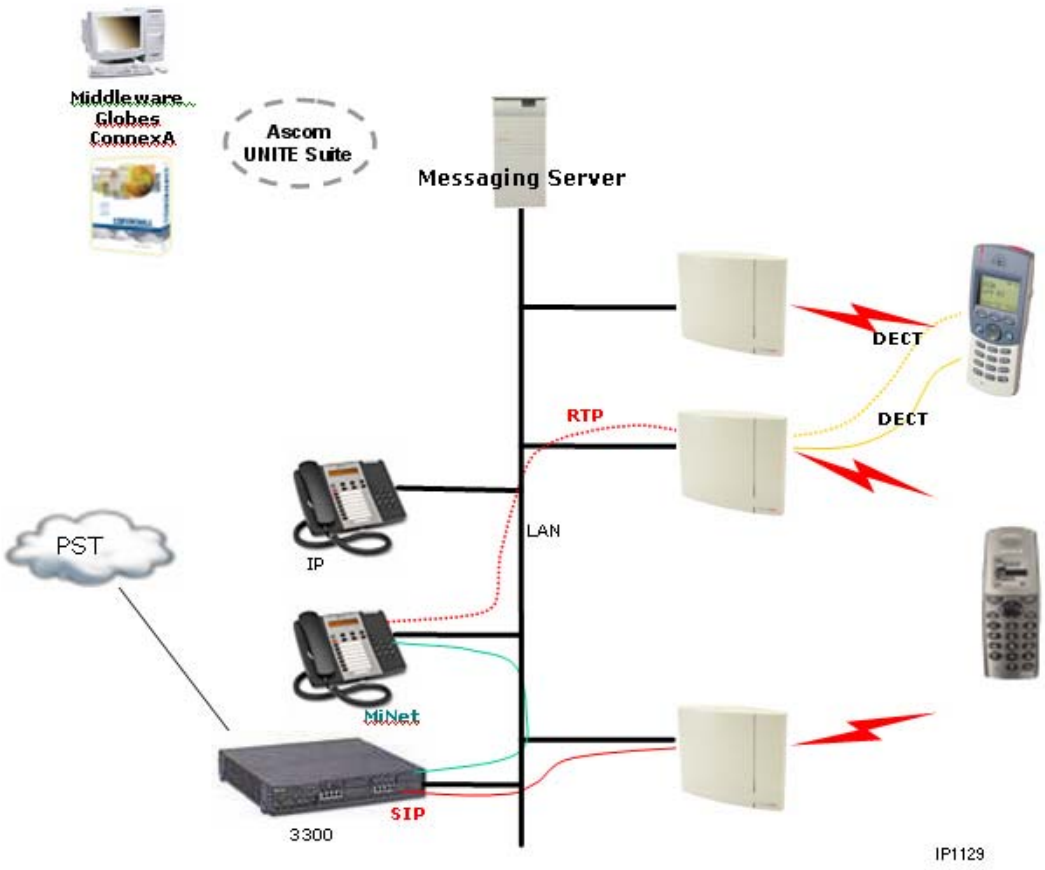


Figure 30: IP-DECT System (Global)

IP DECT Wireless Solution (EMEA)

The Mitel Internet Protocol Digital Enhanced Cordless Telecommunications (IP-DECT) wireless solution is available in the EMEA (Europe, Middle-East and Africa) and the Australia market. It consists of the following components:

- 3300 ICP controller
- Base stations for wireless phones
- OpenPhone 27 wireless phones
- Open Mobility Manager (IP-DECT wireless solution administration application)

The base stations connect to the 3300 ICP controller through the LAN. The wireless phones communicate with the base stations using standard Digital Enhanced Cordless Telecommunications (DECT) protocol. One of the base stations is designated as the Open Mobility Manager (OMM). Like the other base stations, the Open Mobility Manager transmits voice information to and from the wireless sets, but it also provides a management interface that enables you to configure the wireless system settings, base stations, and OpenPhones.

An SNMP agent configured in each base station conveys alarm information and facilitates overall SNMP management of large, wireless networks of base station and OpenPhone 27.

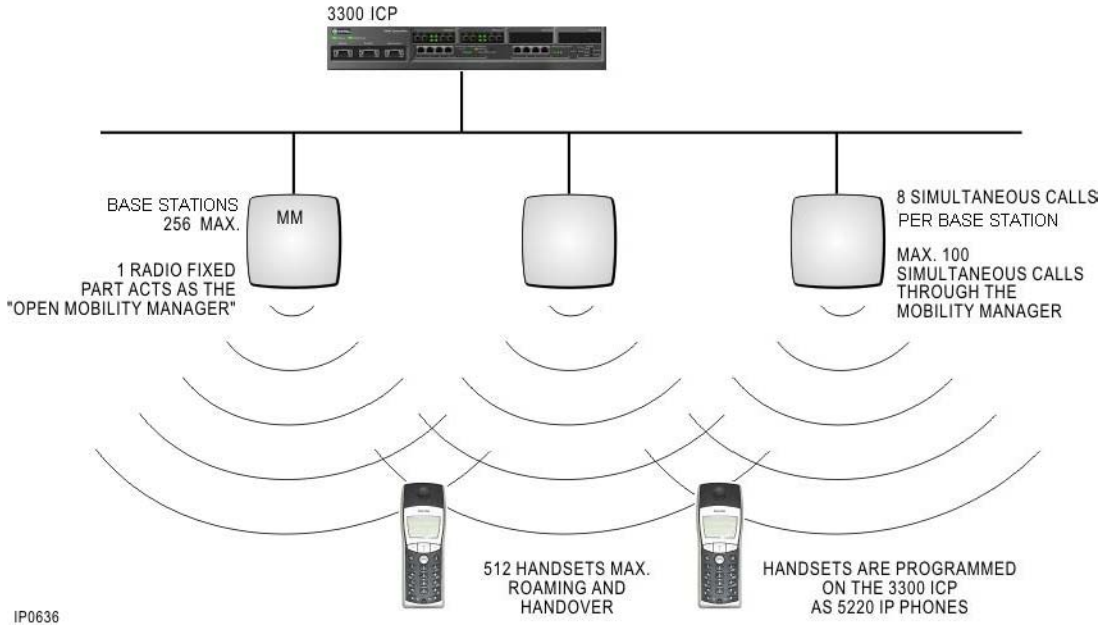


Figure 31: IP DECT Wireless Solution (EMEA version)

IP DECT Handset and DECT Stand

The Mitel 5610 Handset and DECT Stand are peripheral devices for 5300 Series IP Phones. Simply snap the DECT Stand onto a phone in order to support up to eight 5610 Handsets for localized mobile communication. The handsets have vibrant color screens and full dialpads, and can be programmed as unique SIP extensions or as members of a personal ring group.

SpectraLink IP Integrated Wireless Voice Solution

SpecktraLink wireless phones provide access to all the features and functionality available on a Mitel IP-desktop as well as the mobility of a compact 802.11 (Wi-Fi) wireless device. The SpectraLink Wireless Voice Solution is fully IP-integrated with the 3300 ICP to provide a single wireless data and voice infrastructure. SpectraLink Wireless Voice Solution provides investment protection by allowing customers to build wireless infrastructures using a choice of more than 27 different brands of Access Points. SpectraLink allows customers to consider out-of-building, campus-wide or even municipality-wide wireless voice networks. In addition, the SpectraLink Wireless Voice Solution allows the 3300 ICP to act as wireless gateway for third-party PBXs as well as Mitel's SX-200 and SX-2000 platforms. Communication between the 3300 ICP wireless gateway and the legacy PBXs is over standard trunk protocols such as MSDN/DPNSS, Q.Sig, or T1/D4.

SpectraLink Wireless Voice Solution provides:

- Multi-line support similar to the 5324 IP Phone
- Push-to-talk Walkie-talkie style communication

- Choice of two phone styles (Industrial and Enterprise)
- Choice of Access Points
- Support for text alerts
- Extensive integration support for applications such as Nurse-Call, security and emergency response.

In order to implement the SpectraLink Wireless Telephone Solution, you must also order the SpectraLink SVP Server (see SpectraLink Voice Priority Server on page 71). In addition, Mitel provides the optional SpectraLink Open Application Interface for two-way messaging.

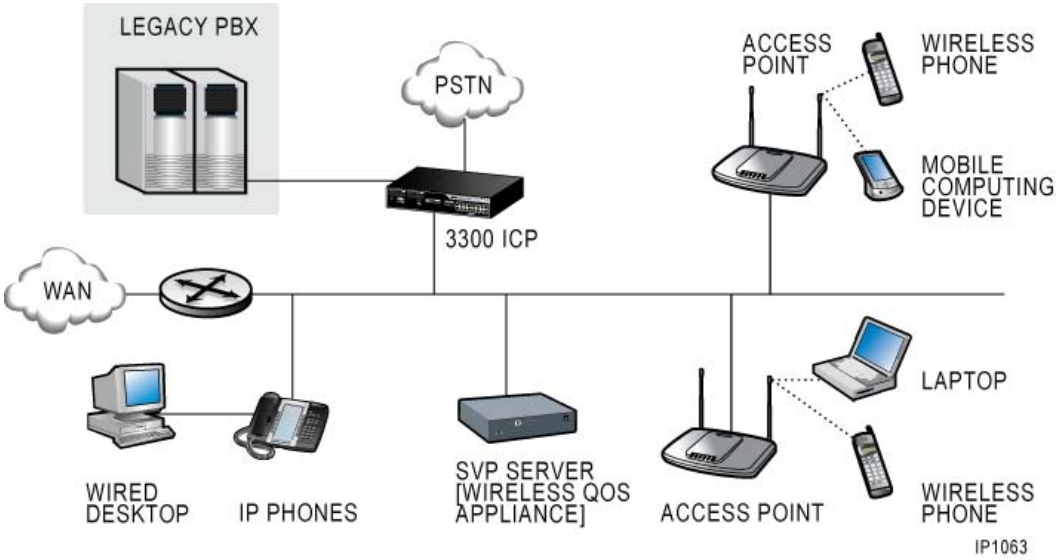


Figure 32: SpectraLink IP Integrated Wireless Voice Solution

SpectraLink Voice Priority Server

The SpectraLink Voice Priority (SVP) Server allows converged voice and data traffic over a common wireless network and reduces packet queuing delays for voice traffic. The SVP Server:

- ensures excellent voice quality on converged wireless networks
- is fully compatible with 802.11b wireless LANs
- handles 80 simultaneous calls per SVP Server (300-600 users).

SpectraLink Open Application Interface

The SpectraLink Open Application Interface (OAI) allows the handsets to function as two-way messaging devices to provide integration with other enterprise systems allowing mobile workers access to critical information. The NetLink OAI Gateway enables third-party applications to communicate directly with up to 10,000 NetLink Wireless telephones.

Messaging

Mitel's Messaging solutions include:

- "Embedded Voice Mail" on page 72
- "Mitel NuPoint Unified Messaging" on page 73

Embedded Voice Mail

MCD includes an integrated fully-featured voice mail system. Up to 30 ports are available for voice mail calls with support for a maximum of 750 mailboxes and 450 hours of storage time.

Features provided by the voice mail system include:

- Standard Unified Messaging allows users to forward voice messages, including Record-a-Call messages, to an e-mail address. Users can choose to manually forward individual voice messages, or automatically forward all voice messages.
- An automated attendant that plays different greetings during open and closed business hours, provides a company directory that uses extension numbers or names as the dialing method, and allows single-digit option selection
- Multi-level auto attendant (MLAA) allows a hierarchical menu to be programmed on the auto attendant. This provides callers with self-service options (for example, "Press 1 for Sales") to reach individuals, departments, pre-recorded information, or to leave voice messages
- Personal Contacts allows users to create a customized voice menu, allowing callers to reach them on cellular phone, fax etc.
- User mailboxes that are password-protected
- A tutorial that assists new subscribers with mailbox setup
- Simple message retrieval
- Easy-to-use menus that allows users to send urgent, private, or certified messages
- Notification of waiting messages
- Record a call that allows users to record a conversation and save it in their voice mailbox.

Voice Profile for Internet Mail

Voice Profile for Internet Mail (VPIM) allows voice mail users to send and receive messages between all the voice mail servers on a network that are VPIM2 compliant, whether they are Mitel systems or third-party products.

Note: VPIM on embedded voice mail does not support G.721 compression; it supports only G.721 without compression. The other sites in the VPIM must also support G.721.

VPIM is supported between embedded messaging and NuPoint Unified Messaging, and is compatible with Hot Desking.

Mitel NuPoint Unified Messaging

Mitel NuPoint Unified Messaging™ (UM) is a highly reliable, scalable, integrated voice and fax unified messaging system that can be accessed anywhere, anytime. NuPoint UM provides users access to a host of highly flexible and customizable applications including call director, speech auto-attendant and Microsoft LCS 2005 integration. These applications allow users the flexibility of having their calls routed to them while they are on the road, or having access to their voice or fax messages from their PC. Simple and cost-effective system configuration, implementation, administration, and management help streamline system management and deliver lower total cost of ownership.

Using the power of Unified Messaging allows users to receive all their voice mails and emails through one interface. When end users are at their desk, they can use their Outlook client to manage their messages and when they are on the road, they can phone into the Mitel NuPoint UM server to listen to their voice mails and emails. Unified Messaging simplifies the end user experience and leads to increased productivity.

NuPoint UM provides the mailbox user with an interface to these capabilities through the NuPoint UM Telephone User Interface (TUI) as well as through Outlook (using the NuPoint UM Outlook Client Plug-in). When using the Outlook Client Plug-in, users are able to record, playback, forward and reply to their voice mail messages as well as view their fax messages. The Message Waiting Indication (MWI) is synchronized on the phone with the reading of messages through the Outlook client, and vice versa to mark Microsoft Exchange voice mail messages as read when the message is played back through the NuPoint UM TUI. Text to Speech (TTS) allows users to listen to their email messages through their NuPoint UM Voice Mail (either unread messages only or all messages).

Additionally, the NuPoint UM Speech Auto Attendant (SAA) is a speech-enabled application that 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. An online tutorial introduces users to the system features, and voice-based help is available to answer questions.

NuPoint UM 640 is designed to scale up and provide voice applications to large enterprises who demand high capacity, high availability and resilient services with the Active/Active configuration. The NuPoint UM system allows the ability to network two active 640s to a single direct-connect storage array, thus allowing them to store all data, including voicemail accounts, messages, greetings and all matters relating to NuPoint users and NuPoint system data in one shared database. If a single 640 fails, the remaining one in the cluster will continue to operate. The overall system appears functional with reduced traffic capacity (that is, ports).

NuPoint UM 640 has been engineered so there can be no single point of failure, thus ensuring high availability and minimizing unplanned downtime, and it can be further integrated into an organization's data center infrastructure. The Active/Active configuration is also available as a four-node NuPoint UM system. All nodes are active and share the same database.

NuPoint UM supports Microsoft Live Communications Server (LCS) 2005. Support for OCS 2007 will be added in a future release.

Some examples of NuPoint UM applications include:

- Paging a mailbox owner when a new voice mail message arrives. NuPoint UM supports SMS notification to cellular phones. SMS notification text-messages users when they receive new voice messages.
- 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.
- Fax support -- you can configure up to six fax channels/ports for each NuPoint UM server. The Fax feature works in a network configuration where the NuPoint UM server is integrated directly with an MCD system or with another PBX.
- Mailbox Maintenance, System Maintenance, Report Generation, and Call Director management from a web-based console. Additionally, multiple Multiple networked NuPoint UMs can be managed from one NuPoint server using Cluster Configuration in the web-based console.
- Supports integration with up to four MCD systems
- Enhanced InBand integration permits users to interface to legacy PBXs, using enhanced inband with DTMF.
- Telephone softkeys that allow users to control voice mail functions through context-sensitive keys on the telephone.

For more information, refer to *Mitel NuPoint Unified Messaging General Information Guide* on Mitel On-line (MOL).

The following diagram illustrates a NuPoint UM system integrated with a 3300 ICP.

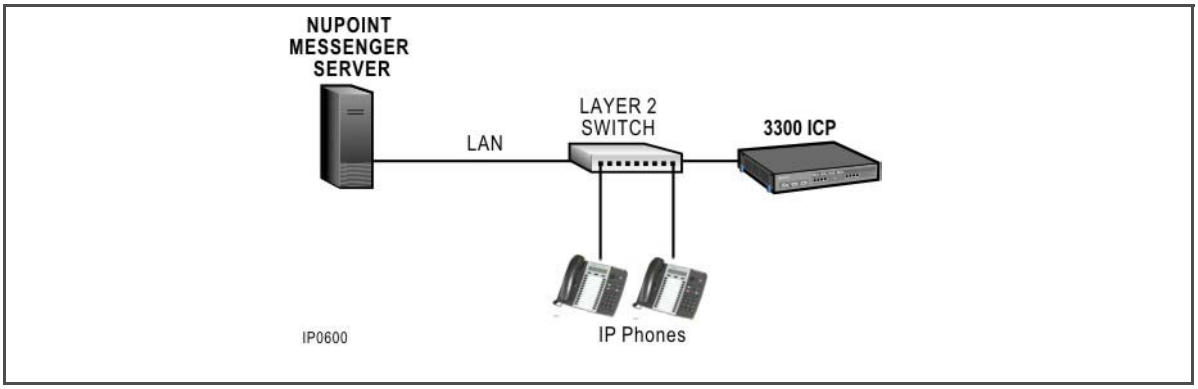


Figure 33: NuPoint Integration with 3300 ICP

Customer Interaction Solutions

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:

- “Automatic Call Distribution” on page 75
- “Applications for Formal Contact Centers” on page 76
  - “Mitel Contact Center Business Edition” on page 76
  - “Mitel Contact Center Enterprise Edition” on page 77
  - “Commander Contact Centre” on page 77
- “Applications for Informal Contact Centers” on page 78
  - “Mitel Customer Service Manager” on page 78

Automatic Call Distribution

MCD provides fully integrated Automatic Call Distribution (ACD) functionality through either the ACDII or ACD express call routing applications. Targeting formal and informal contact centers respectively, ACDII and ACD express provide call distribution, agent mobility, feature configuration, administration, and recorded announcements. The integrated ACD functionality of MCD is enhanced by Mitel’s Contact Center Solutions. These solutions are designed specifically for formal and informal contact centers, and they enable customers to maximize the efficiency of their contact center. Mitel’s Contact Center Solutions are described in more detail below, as well as in the *Customer Interactions Solutions General Information Guide*.

ACD applications benefit from Mitel’s comprehensive Hot Desking features. With Hot Desking, a pool of shared phones can be made available to all agents. Agents may login and logout of any phone with their unique Hot Desk ID and password. The system applies the agent’s personal profile to the set. Once logged in, agents may make themselves present in, or absent from, any one of their groups through a pre-programmed feature access key (FAK). Even when not present in any of their groups, agents still have access to their prime line key. Login and logout may be accomplished using feature access codes (FACs), ensuring the flexibility in choice of sets for the application. These capabilities are all controlled via Class of Service (COS) settings in MCD, ensuring that ACD administrators have full control over how permissions are assigned.

Agents may be active in 16 agent groups at any given time (in an ACD application employing the 3300 MXe II Controller) and 30 agent groups at any given time (in an ACD application employing the 3300 MXe Server).

ACD features dynamic license allocation enabling customers to purchase only the number of concurrent licenses needed for their operation.

For a description of the 3300’s Hot Desking capabilities see “Hot Desking” on page 63

Applications for Formal Contact Centers

Formal contact centers are typified by organizations whose call center is critical to their business. Be they large or small, these customers have advanced needs such as multi-channel, highly customized Interactive Voice Response (IVR), extensive reporting, and customized integrations with Customer Relationship Management (CRM) applications and other business processes. Mitel's solutions provide a formal way of dealing with incoming calls and support a range of basic to advanced functionality and price points. Mitel's Formal Contact Center Solutions are built upon the sophisticated call routing functionality provided by the ACDII application in MCD.

Mitel's Formal Contact Center Solutions include:

- "Mitel Contact Center Business Edition" on page 76
- "Mitel Contact Center Enterprise Edition" on page 77
- "Commander Contact Centre" on page 77

Mitel Contact Center Business Edition

Mitel's Contact Center Business Edition was developed for organizations who need a formal way of dealing with incoming calls, but who require a minimal amount of customization or very advanced capabilities. The solution is designed for individual/single site contact centers that have 25 or fewer agents and focused application needs. Mitel Contact Center Business Edition provides:

- Award-winning graphical agent desktop
- Core set of historical and real time reports
- Consolidated agent and queue management
- Rich voice automatic call distribution (ACD) functionality
- Mitel Contact Center Management tools for small to multi-site contact centers
- Mitel Contact Center Schedueling for automatic agent schedueling based on business rules and required skills
- Mitel Schedule Adherence: complements Contact Center Schedueling to allow administrators to see what agents are doing compared to their schedule
- Mitel Interactive Contact Center and Visual Queue for dynamic telephone system control in real-time
- Mitel Call Accounting
- Automatic Call Distribution
- Mitel Intelligent Queue: a browser-based IVR solution providing advanced routing and self service
- Mitel Multimedia Contact Center: ACD for e-mail, web chat, fax SMS and walk-in
- Mitel Contact Center PhoneSet Manager and Contact Center Softphone
- Mitel Contact Center Screen Pop: CRM screen-pop

- Outbound Dialing: automated dialing
- Remote agents via MBG teleworker service

Note: Contact Center Business Edition is limited to 25 agents and 8 ports.

Mitel Contact Center Enterprise Edition

Mitel's Contact Center Enterprise Edition is a scalable, resilient, and virtual solution for highly sophisticated contact centers of all sizes across one or more locations. Enterprise Edition targets organizations whose call center is fundamentally critical to their business, or their call center *is* their business. These organizations require a highly available system, very advanced integration, extensive reporting and sophisticated routing. All of these complex capabilities must be made available in a distributed (and/or virtual multi-site) environment. The Mitel Contact Center Enterprise Edition meets these demanding call center implementations by providing the feature set of Contact Center Business Edition plus:

- Extensive custom reporting
- Sophisticated routing and highly customized interactive voice response (IVR)
- Customized integrations for customer relationship management (CRM) and workforce management (WFM)
- Resiliency and high availability
- Support for distributed, multi-site, virtual deployments.

Contact center solutions are described in detail in the *Mitel Contact Center Solutions General Information Guide*.

Commander Contact Centre

Note: This solution is available in the UK only.

The Mitel Commander Contact Centre is an innovative advancement in contact center communications and control, extending the boundaries of the customer-agent interaction to support a wide range of contact types in a completely integrated environment. Commander's patented solution provides Multimedia Interaction Management™ through the most comprehensive set of tools on the market — routing, queuing, tracking, and reporting on inbound and outbound calls, e-mail, Web Chats, Web Requests, faxes, voice mail, and blended calls (preview dialing).

Commander handles customer requests from the arrival of an interaction to the final wrap-up. Fully integrated features such as Interactive Voice Response (IVR) and e-mail parsing identify customers and their needs. Commander applications query third-party applications or mainframe databases to lookup customer data or information about a call (that is, who the caller is, caller ID and type of support contract the caller has). By linking to CRM data, Commander retrieves details about the customer. Intelligent queuing and data-directed routing ensure an optimal path for every interaction. By using Web-based administration, real-time monitoring,

and a comprehensive decision management system, organizations can create a complete, customizable picture of the contact center operations for all levels of management.

Commander Contact Centre features are described in detail in the *Commander Contact Center product documentation*.

Applications for Informal Contact Centers

To address the needs of informal contact centers, Mitel Customer Service Manager provides an informal way of sharing incoming calls amongst a team of individuals that support each other but have other primary jobs. This solution is founded in hunt/ring groups and offers basic cross team visibility, reporting and Personal Information Management (PIM) integration. The solution is built upon the call routing functionality provided by the ACD Express application in MCD.

Mitel Customer Service Manager

Mitel Customer Service Manager enables informal call centers or workgroups to efficiently monitor, manage and route calls. It provides real-time business intelligence and measures departmental efficiency. It is deployed using the Mitel Applications Suite (MAS) (see “Mitel Applications Suite” on page 79). Customer Service Manager provides simplicity and flexibility at a low total cost of ownership. It features:

- Mitel Customer Service Manager (CSM) Server – bridges the gap between your phone system and your computer network with computer telephony integration (CTI). The CSM Server monitors the entire phone system, including all internal and external calls, and provides data to other CSM modules for business reporting and team productivity.
- Mitel Intelligent Router – intelligently routes both internal and external calls based on information in the database, with easily customized rules. Rules are created with a powerful GUI and can be applied to both internal and external calls.
- Business Reporting Tools – delivers both historical and real time management reporting as well as call recording capability.
- Team Productivity Tools – tools to make teams more productive including call management, screen pops with CRM integration and call control.
- Single site MCD system support.

Mitel Call Accounting

Mitel Call Accounting is a comprehensive call costing solution that is available as either a single site or multi-site solution, and can optionally be integrated with Mitel Contact Center Management. Mitel Call Accounting enables organizations to monitor and control telecommunication costs and clearly show how much money is being spent and who is spending it. Some features of call accounting include:

- Monitor usage and establish call patterns for departments and work groups
- Track, report and control telecommunication costs
- Track account codes in SMDR reports
- Perform cost recovery and carrier bill reconciliation

- Know if costs are excessive because employees are sharing toll free lines, calling restricted numbers, or calling their friends long distance
- Mitel Subscriber Services (optional module) – enables the administrator to charge back departments, employees and customers using markup or discount pricing
- Mitel Traffic Analysis (optional module) – helps the administrator to determine if the organization is using its incoming, outgoing, and bi-directional trunks efficiently.

Mitel Applications Suite

Mitel Applications Suite (MAS) is a combined application offering made available on a single server, including a management interface that allows you to install and configure multiple Mitel applications. In the past, deploying multiple applications required you to manage separate servers and their associated users. The Mitel Applications Suite reduces cost and time requirements by consolidating the installation and management of multiple applications in one easy-to-use server. Co-residency is provided for multiple applications, such as:

- NuPoint Unified Messaging
- Unified Communicator (UC) Mobile
- Mitel Border Gateway with Teleworker, SIP Trunk Proxy and Web Proxy services
- Audio and Web Conferencing
- Mitel Customer Service Manager

Refer to the latest *Mitel Applications Suite General Information Guide* on the Mitel Customer Documentation web site for a list of the currently supported applications.

Initial provisioning of the Applications Suite is simplified with the use of the Mitel Configuration Wizard. This integrated software application configures the 3300 ICP, MCS, embedded or NuPoint UM voice mail, UC Mobile, and the MBG. Standard or custom settings can be pre-programmed and uploaded to the MCD system. After installation, system configuration and administration of all applications is consolidated in a single web-based console. Common data elements are shared among the applications, reducing both the need for duplicate entry and the possibility for error.

For Applications Suite users, the End User Portal provides a common web interface where users can change their own settings and passwords, search a company directory, and manage their voice mail. Cost reduction while improving efficiency is a requirement and a challenge for most businesses. Mitel's Applications Suite solution is designed to help you meet that challenge.

Hospitality

The following applications comprise the Mitel Hospitality portfolio:

- “Hotel/Motel” on page 80
- “Property Management System” on page 81

Hotel/Motel

This application provides useful features commonly used by hotels, motels, and hospitals.The system can work independently, or in conjunction with a Property Management System (PMS).

An attendant can access hotel/motel management functions from the Guest Services Application (GSA) on the 5540 or 5550 IP Console. There are six modes of operation:

- Guest Service Mode: access to all other modes of operation
- Find Room Mode: search for rooms on the basis of condition and/or occupancy status
- Room Monitor Mode: listen to an activated room monitor extension
- Print Mode: print room status, message registration, or wake-up reports
- Hotel Program Mode: turn system-wide features, such as Call Block on or off
- Guest Room Mode: check guests in and out of the hotel, set automatic wake-up calls, activate call block, and program call restrictions.

Using the Hotel/Motel feature, an attendant can:

- Display information about the guest room, the guest, and the room extension. Existing information can be changed or new information added to a room.
- Check-in and check-out guests, thus keeping track of arriving and departing guests.
- Set the condition and occupancy status of a room.
- Search for rooms by using the room condition and occupancy status as search parameters.
- Listen to a room monitor extension.
- Set an automatic wake-up call for a room extension.
- Enable Call Blocking to prevent calls from being made between guest rooms.
- Restrict the type of calls that a guest can make from a room extension.
- Use Message Registration to calculate the total cost of calls made from a room extension.
- Print Automatic Wake-up, Room Status, and Message Registration reports.
- Access logs generated by the system during operation of the Hotel/Motel feature.

The Hotel/Motel feature set is supported on standalone systems and across a cluster of 3300 ICP systems.

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 3300 ICP via a Telnet connection. Applications that require a serial interface to connect to the network can use the serial port on the 5550 IP Console or a third-party Serial-to-IP port converter

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

Clustered Hospitality Description

Clustered Hospitality provides hotel/motel feature functionality across a cluster of 3300 ICPs. The cluster is comprised of a single Hospitality Gateway ICP and one or more Hospitality ICPs.The Hospitality Gateway ICP interfaces to the Property Management System (PMS) and the Guest Services Application (GSA) on the 5550 IP Console, and it can also host guest rooms and their extensions.

The following diagram depicts a typical clustered hospitality environment:

Note: Enterprise Manager and OPS Manager are not required to manage the telephone directories of the cluster elements if Remote Directory Number (RDN) Synchronization is enabled for the cluster.

Figure 34: Hospitality Deployment Example

Clustered Hospitality supports:

- Hotel logs and reports via a networked printer

- Linked suite membership by suites on any node in the cluster. Support for Shared Telephone Service (STS) is available if all members reside on the same 3300 ICP as the linked suite.)
- Configuration of room extensions and suites from any 3300 ICP in the cluster
- Resilient hotel room extensions
- PMS GRS General Reset/Get Reservation Status. (A Property Management System (PMS) function that compares its check-in/check-out data with the 3300 ICP to ensure the data on both systems match on a cluster-wide basis).

General Business Solutions

The Mitel Business solutions include:

- “Tenanting” on page 82
- “Emergency Services Support” on page 83
- “Emergency Response Advisor” on page 84
- “Multi-Level Precedence and Preemption (MLPP)” on page 84

Tenanting

Tenanting allows a system to be partitioned among a number of tenants and then be configured to appear as a separate system to each tenant. Up to 64 small businesses or departments of a larger business can share system features and capabilities.

Consoles, CO trunks, and dial-in trunks can be allocated individually to each tenant or shared between tenants. Switching to night service can be done centrally, or on a tenant-to-tenant basis. Calls through the system can be blocked, so tenants can only call each other on CO trunks.

Some system features are enabled for each tenant individually while other features are shared by all tenants. You can define groups, such as Attendant Groups, Trunk Groups, and Multiline Appearances with devices belonging to different tenants. The tenanting option must be purchased.

The following conditions apply to Tenanting:

- You can have up to 64 tenants, including the landlord (tenant 1).
- Each tenant can have its own Music on Hold source.
- The following devices and resources can be members of a tenant:
  - IP phones and consoles
  - DNI phones and consoles
  - wireless phones
  - analog trunks
  - digital trunks
- Unless otherwise programmed, all phones, consoles and trunks are in the landlord group.

- IP trunks are not tenantable resources.
- Tenanting is not supported with the following features:
  - Hot Desking
  - Resiliency
- Tenanting is a local system feature only and is not supported in networked or clustered configurations.

Emergency Services Support

With Emergency Services support, when an emergency number is dialed (for example 911, 112), a Caller's Emergency Services ID (CESID) is sent from the MCD system to the Public Safety Answering Point (PSAP). Note PSAP is only relevant in North America. The CESID is used as a key for the PSAP to determine the precise location of the caller. For this reason, it is critical that the CESID database within MCD be kept up to date.

CESID is supported on analog sets, Mitel's digital and IP sets, and generic SIP devices.

When users with digital or analog phones change offices or relocate within a building, a manual update is required to the MCD database and the phone move is typically managed by the Telecoms/IT team. The CESID database must be updated at this time to ensure that the user's new location is accurately reflected in the database.

IP phones, however, can be moved from one location to another, by the user, without the need for a manual update to the CESID database as automatic CESID updating is supported by MCD.

To utilize automatic CESID updating, the network environment must have all Layer 2 (L2) switches that are configured for Cisco Discovery Protocol (CDP), Spanning Tree Protocol (STP), or both. The system performs automatic CESID updating for IP devices that are moved to a known location. CESID Logs and CESID Alarms record all CESID-related activity on the system. By automatically updating this information, businesses save the cost of manual updates and, more importantly, ensure the safety of their employees.

Automatic CESID updates are not supported for teleworker IP phones, Mitel Your Assistant Softphone, or wireless IP phones.

Emergency services are supported for teleworker phones through the use of the Mitel Line Interface Module module. Emergency call routing for teleworkers is provided as follows:

- If Emergency Call Routing is not configured in MCD, the user picks up the teleworker phone and presses the Line Interface Module configured key and dials the emergency number. For more information, refer to Line Interface Module on page 114.
- If Emergency Call Routing is configured in MCD, the user picks up the teleworker phone and simply dials the emergency number.

Emergency Response Advisor

Mitel Emergency Response Adviser is an application that runs on a Microsoft Windows-compatible server and one or more remote terminals. It provides local security personnel with an emergency call display and response console that

- alerts them to new emergency calls
- identifies the exact location of the phone that was used to dial the emergency number
- lists any helpful extra information
- waits for call acknowledgement
- logs the call and time of acknowledgement.

This functionality is added to the existing Emergency Services feature offered by MCD. The switch performs the actual routing of emergency calls to the Public Safety Answering Point (PSAP) where emergency personnel (such as fire or ambulance) are dispatched by PSAP call-takers.

Emergency Response Adviser has options for alerting mobile personnel via their phones or their pagers. It can also simplify the generation of data files necessary for keeping the PSAP up to date with physical plant changes (an essential part of emergencies services management).

Multi-Level Precedence and Preemption (MLPP)

The Multi-Level Precedence and Preemption (MLPP) feature supports emergency communications for the military as part of the Defense Switched Network (DSN). MLPP allows authorized users to

- specify a precedence level when they make a call
- preempt calls that have a lower precedence level.

Preemption allows important calls to take precedence over less important calls. Important calls that need immediate attention are identified by a continuous preemption warning tone. While on a call, if a caller hears a continuous preemption warning tone, the caller must hang up immediately, wait for an MLPP ring, and then answer the telephone.

This functionality is supported for incoming and outgoing trunk calls on T1 ISDN PRI circuits, as well as for internal calls (calls between stations on the same switch).

About Precedence Levels and Service Domains

The precedence level of a call determines whether or not it can be preempted by another call. The following precedence levels are supported:

- FLASH OVERRIDE (highest precedence)
- FLASH
- IMMEDIATE
- PRIORITY

- ROUTINE (lowest precedence).

Calls that have a higher precedence level preempt calls that have a lower precedence level. So, for example, a call with a precedence level of FLASH can preempt a call with a precedence level of IMMEDIATE. In addition, you can designate users as either preemptable or non-preemptable. Non-preemptable users can still assign precedence levels to calls.

Trunk Support

MLPP trunk calls are restricted to TI ISDN PRI circuits on NSUs and embedded T1 cards that are programmed to support T1-619a signaling.

If the precedence level of an outgoing trunk call is above ROUTINE, and there are no idle trunks currently in the selected MLPP trunk route, the system will attempt to find a circuit that has a lower precedence level call. If the system finds a circuit that has a lower precedence, it will attempt to preempt that call and use the circuit for the new call.

If the remote switch (the switch to which the system is connected) needs to make a call above ROUTINE to the system, it may preempt a call on a busy trunk and use that circuit for the new call. After the system seizes an incoming trunk, the far end communicates the precedence level of the call to the system via out-of-band signaling.

Making a Call at a Precedence Level above Routine

To make a call at a precedence level above ROUTINE, a user dials a two-digit access code comprised of an MLPP access digit (range of 2 to 9) followed by a precedence level digit (range of 0 to 4) or a service digit (range of 5 to 9). The MLPP access digit identifies the call as a precedence call; the precedence level digit identifies the actual precedence level (for example FLASH); the service digit allows users to dial out trunks that are not part of the DSN. If the chosen precedence level is less than or equal to the user's programmed maximum, the system allows the call to proceed. Otherwise, the system provides an error tone or announcement.

Third-Party Developer Support

The Mitel Solutions Alliance (MSA) Developers Program offers third-party vendor partners and end customers access to software development tool kits and support services for integration to our award-winning range of IP communication devices.

MSA Universal SDK Development Kit

The MSA Universal Software Development Kit (SDK) is a set of software, testing tools, and documentation that provides developers what they need to effectively develop applications for MCD.

The SDK application contains the following software options and troubleshooting tools:

- MiTAI: Used to enable switch-to-application server communication for multiple switches.
- MiAUDIO: Used to enable an application to process voice on multiple Mitel phones.

- MiTAI Browser Tool: Used to ensure the connection is functioning properly, to make function calls and to view events from the API.
- MiTAI Server Logger Tool: Used to connect to the MCD host platform and download log files, and to capture all MiTAI server incoming and outgoing messages for debugging purposes.
- MiTAI Client Logger Tool: Used to access MiTAI application information, to collect MiTAI API information in a log file, and to capture MiTAI client data about incoming and outgoing messages for debugging.
- MiAUDIO Test Tool: Used to verify that MiAUDIO has been correctly installed and that all connections allow proper communication between the MiAUDIO application and the MCD host platform.

MiTAI

Mitel Telephony Application Interface (MiTAI) is a powerful telephony API designed for applications requiring sophisticated call- and PBX-control functionality. MiTAI offers a full suite of capabilities from simple third-party call control to contact center monitoring and control.

MiTAI follows the client-server model. The server component resides in MCD. The client component is co-resident with the application. A MiTAI application accesses the MCD host platform via a LAN connection.

MiTAI and MiAUDIO

MiTAI can integrate with MiAUDIO, enabling the development of applications that require more sophisticated capabilities beyond the standard servicing of calls. Examples of such applications include voice mail or automated call routing systems requiring DTMF detection.

MiAUDIO

MiAUDIO enables developers to include the processing of telephone audio streams in their applications for MCD. Examples of MiAUDIO applications include a voice mail system, or an automated recorded message delivery system.

MiAUDIO is used to control the calls of a physical phone or a workstation softphone. MiAUDIO can receive and interpret Dual Tone Modular Frequency (DTMF) signals while also handling multiple phones, trunk devices, and routing queues. Applications written for MiAUDIO allow for third-party call control (outside of the "conversation"). MiAUDIO targets server applications controlling multiple devices and handling such things as corporate voice mail, where speech recognition and DTMF detection are required.

Emulating the Mitel 5020 IP Phone controlled by MCD, MiAUDIO provides voice port capabilities to server-based applications. MiAUDIO offers the following:

- Up to 60 ports (softphones) for voice applications
- Voice stream record and playback
- Phone and line device interface for monitoring and controlling the softphone

- DTMF generation
- DTMF detection events for IP- and TDM-sourced calls
- Call control via MiTAI.

Secure Recording Connector

Mitel Secure Recording Connector (SRC) is a call recording solution that allows third-party recording equipment to record Mitel encrypted voice streams. SRC is placed on the LAN and accepts requests from properly authorized Call Recording Equipment (CRE) to establish taps in the voice stream.

SRC is a Mitel Standard Linux-based (MSL) software blade, similar to the MBG teleworker service. Phones that are enabled for call recording register with the ICP via the SRC. SRC then taps (mirrors) the voice streams of any enabled phone, or group of phones, to third-party call recording equipment. Developers can use the SRC-CRE interface to add, remove, and query recording taps.

HTML Toolkit for 5320, 5330, 5340, and 5360 IP Phones

The Mitel 5320, 5330, 5340, and 5360 desktop application phones feature a large graphics display and a built-in HTML player. The Mitel HTML Desktop Toolkit provides developers with the ability to easily develop graphical applications for the phones using a standard web authoring tool.

The toolkit helps organizations tightly integrate the 5320, 5330, 5340, and 5360 into their business processes and deliver tailored functionality for a wide range of business applications targeting horizontal and vertical market sectors. Custom applications result in easy navigation, enhanced usability of the display phone and the attainment of organizational objectives (sales, branding, process improvement, etc.).

Some examples of applications that may be developed with the HTML Toolkit are:

- Hospitality: Room phones can deliver unique guest services
- Education: Classroom phones can be used to take attendance, store student information
- Financial: Latest stock market information can be displayed
- Retail: Inventory checker, inter-store communications
- Healthcare: Medication profiles can be displayed, pharmaceutical prescriptions can be ordered
- General Business: Weather, Photo album, Screen Saver with company logo, Calculator

Tools

This section describes the tools that simplify programming, administration, management, and maintenance tasks:

End User Tools

Desktop Tool

The Desktop tool is a web-based interface that allows users of IP phones to:

- assign features to personal keys
- manage personal contact lists
- add and delete internet bookmarks.

The following figure illustrates the Desktop Tool.

Figure 35: Desktop Tool Interface

Administrator Tools

Group Administration Tool

This web-based interface allows administrators to configure basic IP phone settings for members of their group. Through this tool, a group administrator can:

- set basic system parameters
- create the system telephone directory
- manage extension and group parameters
- add, edit, or delete users from the system directory
- configure voice mailboxes
- program a user's personal keys with features.

The following figure illustrates the Group Administration Tool.

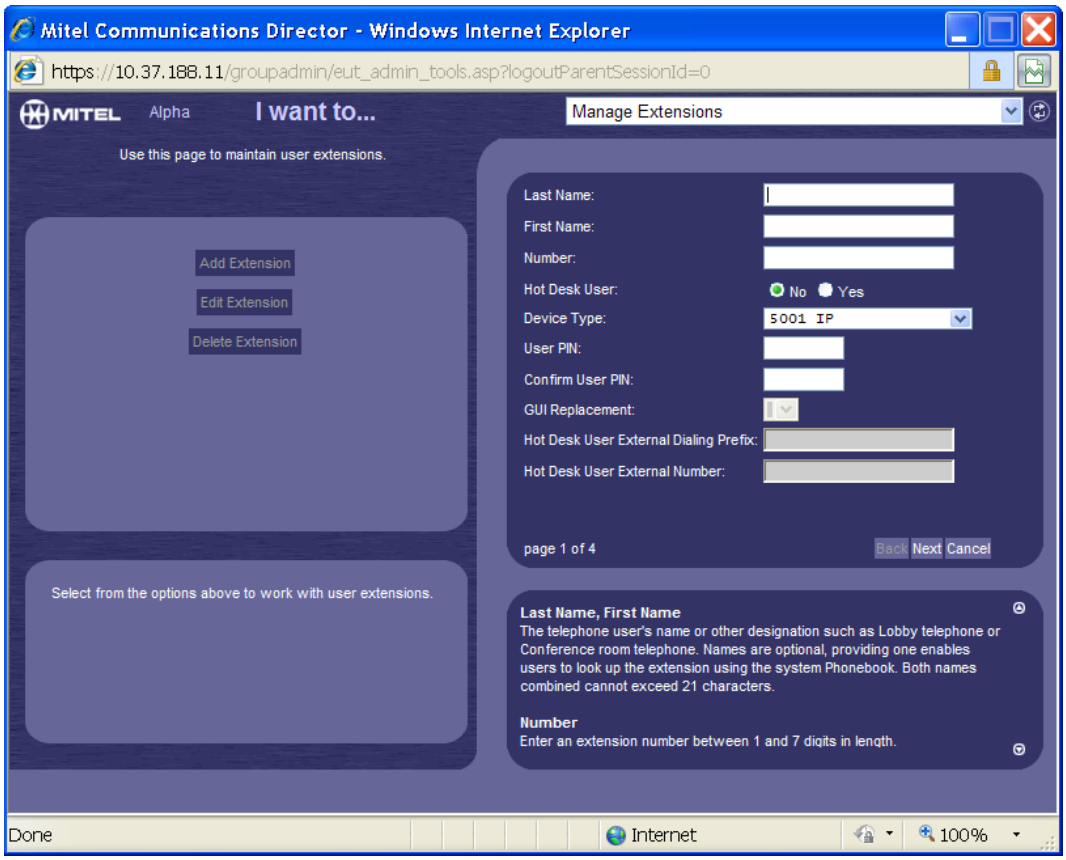


Figure 36: Group Administration Tool Interface

System Administration Tool

The System Administration Tool allows trained technicians and system administrators to program system-wide parameters, voice parameters (line, extensions, management parameters, system directory, and voice mail) and IP networking parameters. This tool also provides access to Maintenance Logs, Software Logs, and Login and Logout Audit Logs.

The System Administration Tool also includes the User and Device Configuration form which provides administrators with the following capabilities:

- Consolidated view of user or device information - simplifies the add, modify and delete functions for most users and devices. It reduces the number of times the same data is entered into the system which reduces the entry errors and time spent on these types of tasks.
- Copy user functionality - allows administrators to quickly create new entries using other existing user or device settings or configuration.
- Import capability - allows administrators to quickly collect and import user and device data via MS-Excel spreadsheets. These spreadsheets contain built in validation similar to ESM data entry rules which helps reduce errors.

The System Administration Tool supports Range programming. Range programming speeds up the programming and configuration of MCD by allowing the administrator to program repetitive data using a single command. The administrator can also print forms and form data.

The System Administration Tool also includes data import functionality which allows the administrator to quickly import large numbers of new users and devices via a .CSV format file. The administrator can collect a large amount of configuration data into the spreadsheet file, and then import it directly into the MCD database. The import functionality eliminates the requirement to manually enter configuration data for each user or device. It saves a considerable amount of configuration time, and it reduces the likelihood of data-entry errors.

Technicians can import new user data when setting up a new system, or administrators can import large numbers of users or devices whenever they need to be added to the system.

In addition, the System Administration Tool can be used to maintain a small group of network elements (10 or fewer) effectively and conveniently without a management tool such as Enterprise Manager. This capability enables an administrator to "reach through" to the System Administration tool of any network element to perform programming, and it also enables the backup of all databases from a single session on a network element. For additional details please see the *Mitel Communications Director System Administration Tool Help* available on Mitel's edocs website.

Note: The use of this feature requires that the network elements be grouped together within an SDS Administrative group. See "System Data Synchronization" on page 93. to learn more.

Note: The System Administration Tool is available in English only.

The following figure illustrates the System Administration Tool.

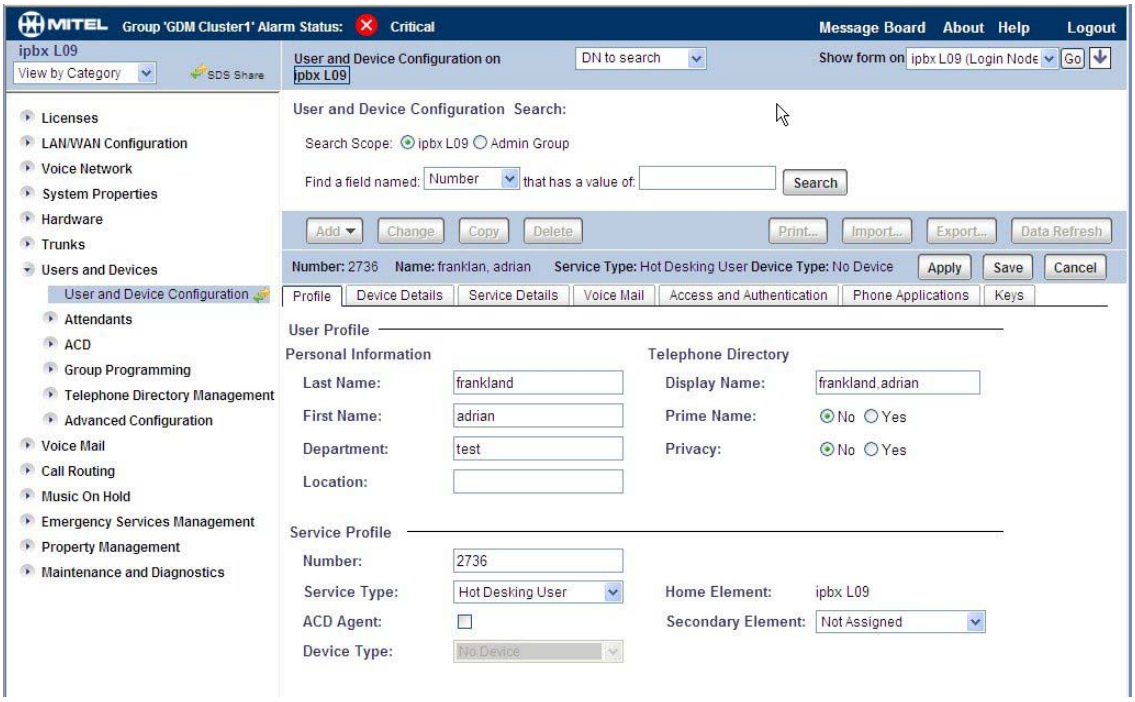


Figure 37: System Administration Tool Interface

Management Tools

Enterprise Manager

Enterprise Manager provides consolidated administration of Mitel's product portfolio. It provides a management desktop, inventory management, configuration, network monitoring, maintenance and diagnostics, and system administration.

Enterprise Manager includes a number of applications that provide:

- Support for up to 1,000 managed Mitel systems and up to 1,000 non-Mitel nodes.
- Network Inventory and Health Monitoring via Enterprise Manager.
- Software Management via MCD Software Installer.
- Support for Management Access Point (MAP).
- Product Management via Embedded System Management (ESM) tools. For more information, refer to product documentation for the 3300 ICP and Customer Interaction Solutions.
- Download of audio files for Music on Hold to multiple MCD nodes via Audio File Manager.
- Report generation using Crystal Reports.

- Collection of passive voice quality statistics from IP sets and consoles in the network via the Voice Quality Manager application. Enterprise Manager polls the MCD host platform for voice quality statistics and exports the data to Viola Networks NetAlly RealTime via XML.
- Discovery of IP sets and certain configured UPS (compliant with SNMP).
- Discovery of applications (such as Unified Communicator Mobile).
- Alarm monitoring on managed networks via the Mitel Alarm Monitor, without having to start the Enterprise Manager client.
- Export capability that allows you to export alarm, event, and inventory data in .csv format to the Enterprise Manager server.
- Alarm history which includes the ability to archive alarms on the server.
- Administrator defined user groups.
- Auto discovery of MCD host platforms, SX-2000, Unpinned Messenger IP, teleworker IP Phones, Wireless Access Points and data network devices.
- Single sign-on authentication based on security group settings

For more information, refer to *Enterprise Manager General Information Guide*.

System Data Synchronization

System Data Synchronization:

- Reduces the time to provision multiple MCD nodes.
- Ensures that changes to network data are performed consistently and accurately across the network, improving change management costs.
- Simplifies network deployment and reduces initial deployment costs by synchronizing the newly deployed MCD nodes with the existing network.
- Enhances security management across the network by allowing accounts and passwords to be managed centrally.

The System Data Synchronization application allows an administrator to synchronize database information among a network or cluster of MCD systems. Any database changes made to any one of the platforms in the network or cluster are applied to the databases of the other platforms.

In a network or cluster of MCD systems, certain programming data, such as Interconnect Handling Restrictions, Feature Access Codes, and Class of Service Options, should be identical for each platform. The System Data Synchronization (SDS) feature eliminates the need for administrators to log into each platform and manually program the data to be the same. It also simplifies all future modifications of system data because it keeps the network and/or cluster element databases in sync. After a network or cluster has been set up with System Data Synchronization, all adds, modifications, and deletions to the system data that have been designated as shared are automatically distributed to the other MCD systems in the network or cluster.

The System Data Synchronization (SDS) feature also synchronizes the data of resilient users and devices between primary and secondary controllers. After data sharing between the primary and secondary controllers has been set up, the application automatically maintains the synchronization of user and device data. This feature keeps user and device data, such as the DND feature key and personal speed call keys, synchronized between the primary and secondary controller regardless of whether users modify their personal settings while they are connected to their primary or secondary controller.

SDS supports Remote Directory Number (RDN) synchronization to distribute the telephone directory entries to all the element databases in a network or cluster. With RDN synchronization, all telephone directory entries added, deleted or modified though the System Administration Tool are automatically distributed to the other elements via SDS.

Management Access Point

The Management Access Point provides secure remote management access and optional alarm monitoring of an MCD system over public network connections. It offers access security and customer protection using dial-up or Internet VPN connectivity.

The Management Access Point requires no additional client software. It uses a 10BaseT Ethernet connection to the customer network, and the dialup versions of the product support connectivity for up to five IP-based systems over standard dialup PSTN V.90 modems. The broadband VPN version, the Management Access Point E/E, provides support for up to 16 IP systems for Internet VPN or dialup modem access.

Remote service engineers connect to the Management Access Point via a PPP or PPTP connection using standard Windows Network Connectivity capabilities and MS-CHAPv2 authentication. However, in order to connect to the Management Access Point console, the customer must enable a privileged remote session for the engineer. The remote engineer can establish an IP connection only to local IP destinations that are configured in the Management Access Point. The engineer must know the destination IP address in order to reach a managed session. At that point, the engineer must have a valid system login account to access the remote system. Once the engineer is logged into the remote system, any administrative, maintenance or diagnostic task can be performed.

The Management Access Point:

- Supports CLI-based call screening so calls are screened based on the origination of phone numbers. (See the latest Management Access Point FCI for supported countries.)
- Uses Point-to-Point (PPP) and Challenge Handshake Authentication (CHAP) protocols to prevent passwords from being sniffed or captured and replayed.
- Provides an embedded firewall so access is restricted to select devices, and the customer LAN is fully protected.
- Prevents remote Management Access Point configuration or firewall changes without the customer granting privilege.
- Tracks configuration changes during privileged connections.

- Supports source IP address screening for VPN connections with the Management Access Point E/E unit.

Maintenance Tools

AMC Licensing

The Applications Management Center (AMC) is accessible through Mitel Online. AMC licensing streamlines the Mitel product licensing process and reduces the interaction between Mitel's distributors, solution providers, and customers. Each group is able to satisfy their licensing needs online.

Each group is set up with an AMC account. Licenses are purchased by the Solution Provider who provides the licenses to the customer. The Solution Provider uses the AMC to create a list of customers and to associate a number of application records with each customer. Application records are simply license profiles of a customer's Mitel products. The Solution Provider can then allocate licenses from their own AMC account to the customer's application records.

All installations require a new Application Record ID and license key (password) that are requested from the AMC via Mitel Online. Upgrades require only a new license key. Technicians can then perform upgrades and updates by:

- synchronizing the system software with the AMC via the internet, or
- installing the software offline (no internet access required) using the Software Installer Tool.

The AMC allows customers to transfer licences between their MCD systems simply and quickly. The only restrictions are that all of the platforms must all belong to the same end customer and that the licences must be automatically activated. Manually activated licenses cannot be transferred to other systems. For more details on AMC licensing, refer to the *AMC Online Help.*

The licensing model, structure and packaging is undergoing significant updates with MCD software release 4.1. In order to simplify and rationalize the process, the number of base packages and user license types will be reduced, and a distinction will be made between "standalone" and "enterprise" systems. For detailed information concerning these updates, please refer to the *3300 ICP Licensing Primer*.

Solution Providers

When the Solution Provider's stock of licenses begins to run low, the Solution Provider orders new licenses from their Distributor or directly from Mitel, depending on their location. The Distributor or Mitel accepts the order and moves the licenses from their AMC account into the Solution Provider's AMC account. The Solution Provider now has the licenses available to sell to end users. At this point, license activation can occur. The end user has no involvement in the process. AMC Licensing allows a license to be moved with a user to different MCD system subject to certain restrictions. For additional details on these restrictions, refer to *AMC Online Help*.

Customer

When a customer wants a new license, they place an order with their Solution Provider. Using the AMC, the Solution Provider assigns a license to the relevant application record. A password is automatically activated and made available to the Solution Provider. The Solution Provider installs the license and the customer can now use the license without having any interaction with Mitel or the Distributor.

MCD Software Installer

The MCD Software Installer is a Windows-based software tool that enables technicians to automatically upgrade software on one MCD system or on multiple MCD systems simultaneously. The MCD Software Installer tool provides a single interface that allows technicians to define which installation steps to perform during the upgrade process. The MCD Software Installer Tool re-synchronizes with the AMC during software upgrades. For more information on AMC, refer to “AMC Licensing” on page 95.

The MCD Software Installer Tool is a standalone tool that simplifies the installation of MCD software by limiting the number of interactive steps that are required during installation. Technicians can use this tool to:

- backup, upgrade, install, change options, and restore
- set up a scheduled time to upgrade an MCD system
- send e-mails to a list of recipients to inform that installation is complete
- run a command line after MCD Software Installer is finished
- show start times and elapsed times when MCD Software Installer is finished
- save settings to .config files for later use
- support more than one instance of MCD Software Installer on one PC.

The Software Installer consists of one main screen that allows technicians to define the IP Address and account properties of the MCD system being upgraded, and a wizard screen that allows technicians to collect the necessary upgrade information.

In 3300 ICP Release 8.0 and later, the MCD Software Installer can be used to upgrade controller software while the system is running. This process is referred to as Online Upgrade. When the installation or upgrade is complete, the SI tool will switch from the old software to the new, and the only downtime will be the time it takes to reboot.

Mitel Integrated Configuration Wizard

The Mitel Integrated Configuration Wizard is an independent software application that simplifies initial programming and allows system databases to be set up quickly. The application is installed onto a maintenance PC and then run while the PC is either connected or disconnected from the MCD system. Technicians can create and save database templates that can be used for new installations. The technician connects the Configuration Wizard with the MCD system through the network and then applies the database.

The Configuration Wizard can also be used to commission NuPoint Unified Messaging, Unified Communicator Mobile, and teleworker users on a Mitel Applications Server. Refer to the *Mitel Communications Director System Administration Tool* online help for more information on the Mitel Integrated Configuration Wizard.

Line Measure Tool

The Line Measure Tool (LMT) allows technicians to determine the line settings for Loop Start (LS) trunks that are connected to the AX Controller Card Chassis, Analog Main Board, Analog Option Board, or ASU II by running the following tests:

- Individual or Batch Line Quality Test: Reports the loss level, Echo Return Loss (ERL), and line quality for a specified trunk(s). A recommended setting is provided based on the test results.
- Individual or Batch Distortion/Echo Test: Measures the non-linear distortion effects (for example, from clipping) of a specified LS trunk(s) for each candidate balance circuit setting.

The Line Quality test allows technicians to obtain the optimum Balance Network Setting and Trunk Category for each LS trunk, based on the signals received from the CO. These settings are then programmed into the Analog Trunks form of the LS trunk to reduce the possibility of echo and audio level issues between the trunks and IP phones.

Technicians can run the Loop Start (LS) Measure tests using calls between a 3300 ICP and CO, or by looping calls through one LS trunk on the 3300 ICP back through other LS trunk on the same 3300 ICP.

Alarms Management

The 3300 ICP system raises an alarm when an anomaly is detected and corrective action is required. The system continuously provides all attendants who use the Mitel consoles with alarm status information. Alarm threshold levels are programmable. There are three classes of alarms:

- Critical: indicates a loss of service that demands immediate attention. This alarm invokes System Fail Transfer.
- Major: indicates a fault that affects service to many users. This alarm usually results in a major degradation in service and requires attention to minimize customer complaints.
- Minor: indicates any fault that does not fall into either of the above two classes. When the system is not 100% operational, a minor alarm is raised. It may require the attention of a technician, but it is not urgent. Examples of a minor alarm include the loss of a single line or trunk circuit.

The system clears an alarm condition when the fault is corrected.

Remote Alarms Notification

MCD can e-mail notification of alarm conditions to up to 10 addresses should a critical, major, or minor alarm occur. Prompt notification of alarms ensures that issues can be resolved quickly.

The technician or administrator can set up this feature to send notifications based on the severity of the alarm.

MCD also supports Simple Network Management Protocol (SNMP). SNMP traps can be used to monitor the system devices and functions. The SNMP agent in MCD communicates with SNMP-compatible Network Management Stations and supports industry-standard MIB-II definitions as well as proprietary SNMP extensions.

SNMP defines asynchronous messages called "traps". SNMP traps are generated to alert the administrator when significant events take place such as alarms are triggered or cleared. Traps are sent by the SNMP agent in MCD to an SNMP manager usually to report error conditions, but other types of messages can also be transmitted.

Controlled System Access

System Administrator Policies allow the customer to control a technician's or administrator's access to programming forms in the System Administration Tool. When you create a policy, you set permissions that grant Read or Read/Write access to forms. You can also allow remote access or deny access to forms. Denying access to a form hides the form from the technician or administrator.

System Data Synchronization can distribute the policies to all platforms in a cluster of MCD systems.

Mitel also offers the Management Access Point (MAP) to provide secure controlled access to the system from a remote location. The MAP allows the customer to control the technician's remote access to the system and system tools.

IP Phone Analyzer

The IP Phone Analyzer is a Windows application that collects performance information from IP Phones within the network. It allows the technician to use one PC to monitor the status of all IP phones on the system. IP Phones within the network send debug, status, and statistical information to the IP Phone Analyzer. The IP Phone Analyzer allows the technician to direct phones to new IP Phone Analyzer addresses via an MCD Maintenance task. This eliminates the requirement to reset the phones manually.

The application provides the following information in four views:

- **Status View** - Displays status of each phone registered with the analyzer application, MAC Address, IP Address, Directory Number, State, Link Lost, Set Type, Absolute Time, Load Revisions, Current ICP, and the CODEC type being used by each set on the network.
- **Packet View** - Displays trace messages sent from each set for analyzing purposes.
- **Packet History View** - Sorts messages received by the IP Phone Analyzer.
- **Call Statistics View** - Displays call statistics, including RTP statistics, collected from IP sets.

ISDN Maintenance and Administration Tool

The ISDN Maintenance and Administration Tool (IMAT) provides the programming interface for Network Service Units (NSUs) that are connected to the system. IMAT requires Microsoft Windows 95, Microsoft Windows 98, Microsoft Windows 2000 Professional or Windows XP Professional.

Note: Embedded PRI programming for the T1/E1 module is supported through the System Administration Tool instead of IMAT.

Desktop Devices

Mitel offers a broad range of desktop phones, wireless phones, phone accessories, conference units, and consoles to meet user needs — from basic service to advanced feature and display capabilities.

Feature Support Matrix

The following table summarizes the features provided by Mitel IP Phones:

Physical	5302	5304	5312	5324	5320/5330	5340	5360
Desk/Wall Mountable	Yes	Yes	Yes	Yes	Yes	Yes	Yes
Desk/Wall Mount Stand	Not Included (ordered separately)	Not Included (ordered separately)	Included	Included	Included	Included	Included
5300 Series Handset Version 4	Yes	Yes	Yes	Yes	Yes (Wideband on 5320, narrow-band on 5330)	Yes (Wideband)	Yes (Wideband)
Length of handset cord	3 meters / 10 feet	3 meters / 10 feet	3 meters / 10 feet	3 meters / 10 feet	3 meters / 10 feet	3 meters / 10 feet	3 meters / 10 feet
LAN Ports	2-Port	2-Port	2-Port	2-Port	2-Port	2-Port	2-Port
Ethernet Cable (2 meters / 7 feet)	Included	Included	Included	Included	Included	Included	Included
Compression Support	G.711, G.729a	G.711, G.729a	G.711, G.729a	G.711, G.729a	G.711, G.729a	G.711, G.729a	G.711, G.729a
Voice QoS (802.1p/q)	Yes	Yes	Yes	Yes	Yes	Yes	Yes
Encryption	SIP Authentication	128 bit AES*	128 bit AES*	128 bit AES*	128 bit AES*	128 bit AES*	128 bit AES*
802.1x Support	No	Yes	Yes	Yes	Yes	Yes	Yes
CLASS B Support	Yes	Yes	Yes	Yes	Yes	Yes	Yes
Headset Jack	No	No	Yes	Yes	Yes	Yes	Yes
Peripherals (Modules) Support	No	No	No	Yes	5330 only (Does not include PKM support)	Yes (Does not include PKM support)	Yes (Does not include PKM support)
WLAN Stand/GigE Stand Support	No	No	Yes	Yes	Yes	Yes	5360 has embedded GigE
*Advanced Encryption Standard							
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Powering Options	5302	5304	5312	5324	5320/5330	5340	5360
Ethernet / AC Power Adapter Support (48 VDC LAN Power)	Yes	Yes	Yes	Yes	Yes	Yes	Yes
802.3af Power over Ethernet Compliant	Yes	Yes	Yes	Yes	Yes	Yes	Yes
Power Consumption (Idle)	Not available	2.03W	2.40W	2.40W	3.2W	3.2W (5340) 4.2W (5360)*	3.2W (5340) 4.2W (5360)*
Power Consumption (Typical)	3.8W	2.88W	3.32W	3.23W	4.8W (5330) 4.3W (5320)	4.8W (5340) 7.4W (5360)*	4.8W (5340) 7.4W (5360)*
Power Consumption (Maximum)	4.7W	3.45W	3.87W	3.87W	5.8W (5330) 5.3W (5320)	5.80W (5340) 7.9W (5360)*	5.80W (5340) 7.9W (5360)*
* 10/100 MB Mode values. GB Mode values: Idle - 4.8W; Typical - 8.6W; Maximum - 9.2W							
Display	**5302**	**5304**	**5312**	**5324**	**5320/5330**	**5340**	**5360**
Color	N/A	White	White	White	White	White	16-bit Color, Touch Sensitive
Size (pixels)	N/A	2 lines X 20 characters	2 lines X 20 characters	2 lines X 20 characters	320 x 240 (1/4 VGA)	320 x 240 (1/4 VGA)	800 x 480 (7in.)
Number of Pixels	N/A	160 x 28 (w x h)	160 x 28 (w x h)	160 x 28 (w x h)	160 x 320 (w x h)	160 x 320	800x480 (w x h)
Pixel Size	N/A	0.43 x 0.43mm	0.43 x 0.43mm	0.43 x 0.43mm	0.37 x 0.40 mm	0.37 x 0.40 mm	0.37 x 0.40 mm
Illumination	N/A	Reflective Backlit	Reflective Backlit	Reflective Backlit	Transmissive FSTN with White LED Backlight (5330 only)	Transmissive FSTN with White LED Backlight	TFT Color with LED Backlight
Contrast Adjust	N/A	Yes	Yes	Yes	Yes	Yes	Yes
Display (soft) Keys	N/A	No	No	Yes	Yes	Yes	Yes
Auto Dimming	N/A	Yes	Yes	Yes	Yes (Programmable) (5330 only)	Yes (Programmable)	Yes (Programmable)
Backlight Off Capability	N/A	No	No	No	Yes (5330 only)	Yes	Yes
Chinese Character Support	N/A	Yes	Yes	Yes	Yes	Yes	Yes
							Page 2 of 4

Function Keys	5302	5304	5312	5324	5320/5330	5340	5360
Number of Programmable Feature/Line Appearance Keys	6	9	12	24	8 on 5320 24 on 5330 (Self-labelling)	48 (Self-labelling)	48 (Self-labelling)
Fixed Feature Keys	5	2	10	10	10	10	10 plus 9 quick launch icons on Gadget Sidebar
Softkeys	0	0	0	3	3	6	6
Multi-line	Yes (the lines have to be configured as multi-call line appearances)	Yes	Yes	Yes	Yes	Yes	Yes
Hold	Yes	No (Definable)	Yes	Yes	Yes	Yes	Yes
Redial	Yes	No (Definable)	Yes	Yes	Yes	Yes	Yes
Cancel	No	No (Definable)	Yes	Yes	Yes	Yes	Yes
Volume Up/Down Keys	Yes	Yes	Yes	Yes	Yes	Yes	Yes
Ringer Up/Down Keys	Yes	Yes	Yes	Yes	Yes	Yes	Yes
Message Key	No (Definable)	No (Definable)	Yes	Yes	Yes	Yes	Yes
Transfer/Con-fence Key	Yes	No (Definable)	Yes	Yes	Yes	Yes	Yes
Call Forward (On/Off) Key	No	No (Definable)	Yes (Definable)	Yes (Definable)	Yes (Softkey)	Yes (Softkey)	Yes (Softkey)
Call Me Back Key	No	No (Definable)	Yes (Definable)	Yes (Definable)	Yes (Softkey)	Yes (Softkey)	Yes (Softkey)
Phonebook/ Directory Key	No	No (Definable)	Yes (Definable)	Yes (Definable)	Yes (Via Menu Key)	Yes (Via Menu Key)	Yes (Via Menu Key)
Microphone Key	No	No	No	No	No	No	No
Mute Key	No	No	Yes	Yes	Yes	Yes	Yes
Speakerphone	No	No	Yes	Yes	Yes	Yes	Yes
Program/ Superkey	No	No (Definable)	Yes	Yes	Yes (Via Menu Key)	Yes (Via Menu Key)	Yes (Via Menu Key)
Desktop User Tool	No	Yes	Yes	Yes	Yes	Yes	Yes
Indicators	**5302**	**5304**	**5312**	**5324**	**5320/5330**	**5340**	**5360**
Feature/Line Appearance LEDs	2	2	12	24	8 on 5320 24 on 5330	48	48
Message Waiting LED	Orange	Orange	Orange	Orange	Orange	Orange	Orange
Hold	Yes (Flashes Orange)	N/A	Yes (Flashes Orange)	Yes (Flashes Orange)	Yes (Flashes Orange)	Yes (Flashes Orange)	Yes (Flashes Orange)
Hold Button	Black	N/A	Red	Red	Red	Red	Red
Line LED Color	Orange	Orange	Orange/Green	Orange/Green	Orange/Green	Orange/Green	Orange/Green
Ringer LED	Orange	Orange	Orange	Orange	Orange	Orange	Orange
Microphone/ Mute LED	No	No	Orange	Orange	Orange	Orange	Orange

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Acoustic Functions	5302	5304	5312	5324	5320/5330	5340	5360
Ringing Volume Adjust	Yes	Yes	Yes	Yes	Yes	Yes	Yes
Handset Volume Adjust	Yes	Yes	Yes	Yes	Yes	Yes	Yes
Handsfree Speakerphone	No	No	Yes	Yes	Yes	Yes	Yes
Handsfree: Half Duplex Full Duplex	None	None	Full Duplex	Full Duplex	Full Duplex	Full Duplex	Full Duplex
Wideband Audio Hardware	No	No	No	No	Yes (5330 only)	Yes	Yes
On-Hook Dialing	No	No	Yes	Yes	Yes	Yes	Yes
On-Hook Call Announce (Paging Receive Capability)	Yes	Yes	Yes	Yes	Yes	Yes	Yes
Off-Hook Call Announce	No	No	Yes	Yes	Yes	Yes	Yes
Multicasting Capable	Yes	Yes	Yes	Yes	Yes	Yes	Yes
Amplified Receive >12 dB	Yes	Yes	Yes	Yes	Yes	Yes	Yes
Hearing Aid Compatible (HAC) Handset	Yes	Yes	Yes	Yes	Yes	Yes	Yes
Ring Warble / Pitch Adjust	No	No	Yes	Yes	Yes	Yes	Yes
System Software Requirements	**5302**	**5304**	**5312**	**5324**	**5320/5330**	**5340**	**5360**
3300 ICP	Release 8.0 (UR2) or later	Release 9.0 (UR1) or later	Release 9.0 (UR1) or later	Release 9.0 (UR1) or later	5320: MCD Release 4.1 (SP1) or later 5330: Release 8.0 (UR3) or later	Release 9.0 (UR1) or later	Release MCD R4.0 (SP1) or later
Mitel Border Gateway telworker service	N/A	Release 5.2 (SP1) or later	Release 5.0 or later	Release 5.0 or later	5320: Release 5.2 (SP1) or later 5330: Release 4.5 or later	Release 4.1 or later	Release 5.2 (SP1) or later
SIP Software Platform	N/A	SIP Release 7.2 or later	SIP Release 7.2 or later	SIP Release 7.2 or later	5320 - SIP Release 8.0 or later 5330: SIP Release 7.2 or later	SIP Release 7.2 or later	SIP not supported
							Page 4 of 4

Basic IP Phone

Mitel's basic IP phone is an economical, entry-level IP desktop device ideal for educational, health care, hospitality and general business markets.

- **5302 IP Phone** - a dual port IP telephone designed for users who require basic telephone and messaging services. A computer or other network device can be connected to the network through the second port on the 5302 thus reducing cabling costs.

Figure 38: Basic IP Phone

Display Phones

Mitel's display phones provide intuitive user access to more sophisticated call handling and converged applications supported by MCD. These phones support dual ports, MiNet and SIP protocols, and feature a 2 x 20 backlit character display. The MBG teleworker service allows the 5304, 5312 and 5324 phones to be located off-site and still be supported by MCD.

- **5304 IP Phone** - entry-level display phone supports 2 line keys with LED indication, 7 programmable multi-function keys in a small footprint appealing to the hospitality, education, retail, healthcare and general business market segments.
- **5312 IP Phone** - supports full duplex handsfree operation, 11 programmable multi-function keys, and a prime line key.
- **5324 IP Phone** - supports full duplex handsfree, 23 programmable multi-function keys, three intuitive call state sensitive softkeys, and a prime line key.

5304 IP Phone

5312 IP Phone

5324 IP Phone

Figure 39: Display Phones

Desktop Application Phones

These phones provide enhanced user value, innovative features and applications. Users can access features quickly using the many programmable self-labeling keys. A large graphics display and intuitive softkey interface provide easy-to-use applications such as PhoneBook and Call History. These phones are ideal for enterprise executives, managers, ACD agents, ACD supervisors, or teleworkers:

- **5320 IP Phone** - provides 7 programmable multi-function keys and three intuitive call state sensitive softkeys.
- **5330 IP Phone** - provides 24 programmable multi-function keys and three intuitive call state sensitive softkeys.
- **5340 IP Phone** - provides 48 programmable multi-function keys and six intuitive call state sensitive softkeys.
- **5360 IP Phone** - provides 48 programmable multi-function keys on a large color touch-sensitive display.

These phones provide a built-in HTML toolkit for desktop applications development (See “HTML Toolkit for 5320, 5330, 5340, and 5360 IP Phones” on page 87 for additional details). The following Mitel phone applications are currently available:

- 5300 Intelligent Directory Application: provides access to Active Directory and Outlook contacts on the 5320, 5330, 5340 and 5360 IP Phones.
- 5300 Intelligent Directory with Presence Option: provides presence information on the 5320, 5330, 5340 and 5360 IP Phones for contacts via Microsoft Office Live Communications Server.

For more information on other desktop applications, see “Applications” on page 51.

5320 IP Phone

5330 IP Phone

5340 IP Phone

5360 IP Phone

Figure 40: Desktop Application Phones

Mitel 5560 IPT

The 5560 IPT is a dual display / dual handset, multi-line trading appliance that is made especially rugged for operation within the high activity environment typified by trading floors. It is designed for traders who require access to many lines and who handle a high volume of calls on the stock trading floor. The 5560 IPT combines the speed and performance that split-second trading demands, at a fraction of the total cost of ownership of other turret solutions.

Note: You must obtain channel designation to sell the 5560. Contact your Mitel AE for information.

The 5560 IPT allows traders to:

- Accelerate multi-tasking with dual handsets and displays
- Prioritize calls using the multi-line display and float keys
- Cover other traders' calls within the team
- Access other trading partners with one touch dialing
- Handle two active calls at the same time
- Access embedded phone applications.

Figure 41: 5560 IPT

Wireless IP Phones (3300 ICP deployments only)

The 3300 ICP supports the following wireless phones:

- **SpectraLink NetLink i640** - 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 NetLink i640 Wireless Telephone 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.
- **SpectraLink NetLink e340/h340** - supports a broad range of enterprise applications and is 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.
- **OpenPhone 27** - an IP DECT phone for the EMEA market only. It provides convenient softkey control of the 3300 ICP features as well as a variety of other features that allow users to make calls quickly and easily.
- **5602, 5603, 5604, 5606 (Standard, Alarm, and Services) Wireless Handsets** - IP DECT phones for the IP-DECT Wireless System (Global). These handsets provide voice communication, text messaging, alarm handling plus a core of basic telephony features based on SIP integration with the 3300 ICP.
- **5610 DECT Handset and IP DECT Stand** - IP DECT phone and stand for Mitel 5300 Series IP Phones. The stand connects to the PC port on the phone and supports up to eight handsets. The handsets can be programmed as unique SIP extensions or as members of a personal ring group associated with the phone.

NetLink i640

NetLink e340/h340

OpenPhone 27

Mitel 5610

Mitel 5602

Mitel 5603

Mitel 5604

Mitel 5606

Figure 42: Wireless Phones

IP Phone Accessories

The following table lists Mitel's IP Phone accessories and identifies the supported sets:

Accessory	5302	5304	5312	5324	5320	5330	5340	5360
IP Programmable Key Module (12 or 48 keys)	No	No	No	Yes	No	No	No	No
5310 IP Conference Unit	No	No	No	Yes	No	Yes	Yes	Yes
Line Interface Module	No	No	No	Yes	No	Yes	Yes	Yes
Cordless Module	No	No	No	No	No	Yes	Yes	Yes
Wireless LAN (WLAN) Stand	No	No	Yes	Yes	Yes	Yes	Yes	Yes
Gigabit Ethernet Stand	No	No	Yes	Yes	Yes	Yes	Yes	Yes
IP DECT Stand	No	No	Yes	Yes	Yes	Yes	Yes	Yes

In addition to these IP Phone accessories, the IP Paging Unit is available for the system.

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

Figure 43: Programmable Key Modules

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, 5340, or 5360 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, 5340, and 5360 IP Phones.

Figure 44: 5310 IP Conference Unit

Line Interface Module

Mitel's Line Interface Module (LIM):

- Enables incoming and outgoing analog PSTN calls directly from an IP phone
- Supports failover to an analog line in the event an IP connection is lost
- Provides emergency dialing support for IP phones such that emergency calls connect through the analog PSTN connection.

The Line Interface Module has two different modes of operation: LIM Mode and Failover Only Mode. The System Administrator sets the operation mode through system programming.

- LIM Mode (recommended for teleworker/remote configurations) allows the user to select an external analog line via a line key programmed on the 5324, 5330, 5340, or 5360 IP Phones. The analog line can be used at any time.
- In Failover Mode the Line Interface Module line can only be used when the IP connection has failed (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).

Cordless Handset and Headset

The Cordless Handset and Cordless Headset offer corridor mobility for Mitel 5330, 5340, and 5360 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.

Figure 45: Cordless Handset and Headset

Mitel Wireless LAN (WLAN) Stand

The Mitel Wireless LAN (WLAN) Stand enables IP Phones to operate in a wireless environment. The stand provides the following interfaces:

- 10/100 Ethernet interface that is used to attach the WLAN Stand to the 5300 series phones
- 802.11b/g wireless interface that connects the IP Phones with the wireless network.

The WLAN stand and its attached phone can act either as a Wireless Client terminal (station) or an Access Point:

- As a Wireless Client terminal, it address market opportunities where only WLAN connectivity is available, such as new or temporary installations where it is more cost-effective to implement a WLAN rather than a wired LAN.
- As an Access Point, it is an attractive solution for small installations such as retail outlets or service depots where power is more accessible than LAN connectivity and where an existing WLAN infrastructure is unlikely.

Mitel Gigabit Ethernet Stand

The Gigabit Ethernet (GigE) Stand enables the phones to operate in a 10/100/1000 Mbit/s Ethernet (GigE) LAN environment and allows unconstrained Gigabit Ethernet bandwidth from the network to desktops. The GigE Stand supports the IEEE 802.3af Power over Ethernet standard, eliminating the need for a separate power supply to power the IP phone.

The GigE Stand attaches to the base of the IP phone and replaces the existing stand. It has three ports:

- GigE LAN port (to connect to the Gigabit Switch)
- GigE PC Port that allows a GigE-equipped PC to connect to the LAN via the stand
- 10/100 Mb Ethernet connection to the attached phone.

Wireless LAN Stand

Gigabit Ethernet Stand

Figure 46: Accessories - Stands

Mitel IP Paging Unit

The Mitel IP Paging Unit is an optional module that provides overhead or loudspeaker paging functionality.

The IP Paging Unit is installed as a standalone 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.

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

Figure 47: IP Paging Unit

Mitel 5550 IP Console

The Mitel 5550 IP Console is an advanced PC-based console and administration application. It has a highly intuitive Graphical User Interface (GUI) including screen based call status and call handling prompts. A telephony keypad and dual handset/headset jack provide fast, efficient attendant call handling.

The 5550 IP Console is ideal for both departmental and enterprise attendants requiring fast and easy access to call-processing functionality, and the ability to use other applications on the same PC.

In the low/medium traffic areas, the console operator can use the same PC for processing calls and for day-to-day office tasks (such as e-mail and word processing). By eliminating the need for a separate PC at the attendant station, this solution becomes more economical for the company. The 5550 IP Console provides easier access to future software upgrades, without the requirement to replace the hardware.

The 5550 IP Console solution consists of the telephony hardware and the console application software that enables it to run on a customer-supplied PC.

Figure 48: 5550 IP Console

Mitel 5540 IP Console

The 5540 IP Console performs call handling functions as well as some maintenance and administrative functions (such as moves and changes). The four-line by 80-character alphanumeric display shows source and destination, time and date, call waiting, and station information.

Features of the 5540 IP Console include:

- Two Ethernet ports
- Power over Ethernet (PoE)
- Support for the MiNet protocol (single mode).

Because the 5540 IP Console lacks a printer port, it cannot be used for printing. In addition, it does not support macro keys, maintenance commands, or the application menu.

Figure 49: 5540 IP Console

Features

Features of Mitel Communications Director

The following table describes features of MCD features and indicates whether the feature is supported by Resiliency. N/A indicates that the feature is not specifically related to resiliency or a resilient device, but the feature will function on a secondary controller in a resilient configuration.

Feature Name:	Description:	Support while the set is on the secondary controller
911/Lockout Notification to ONS/CLASS Sets	Allows an ONS CLASS extension to be programmed for 911 notifications. The 911 caller's name and number is identified on the display. This application is ideal for after-hours operation, when the attendant or sub-attendant is not at the desk. For example, in hotels for security guards, or in hospital applications when the on-duty personnel is away from their desks, they can still be notified of lockout alarms and/or 911 calls, with the use of an ONS/CLASS portable display telephone.	N/A
911 Console overflow	911-call info is split over to the console.	Yes
E-911 Support	Displays indicate the extension and the location of the person who dialled 911. Notifications of 911 calls are audible, continuous, and distinct from regular ringing patterns when the set is idle and on hook. If the user is already on a call, a new call tone alerts the user to the alarm condition. The alarm overrides sets having DND enabled.	Yes
Account Codes -Default	Default Account Codes are entered automatically by the system each time a user dials an external number. They may be used to segregate groups in SMDR for billing.	Yes
Account Codes -Verified and Non-Verified	Allows you to access features that are not normally available at a station. These account codes can be used to change the COS and COR at any station. Non-Verified Account Codes allows you to enter codes on the SMDR record for billing and/or call management.	Yes
Account Code Reporting for Internal SMDR	During a two-party call, Verified and/or Non-verified Account Codes can be reported in Internal SMDR logs. Each time an Account Code is entered during the call, a new SMDR log is generated. The first Verified/Non-verified Account Code entered during a call is the active Account Code. When subsequent Account Codes are entered during the call, a new SMDR log is generated. The SMDR log reports the previously active Account Code in the Call Completion field of the SMDR log.	Yes
Account Codes -System	System Account Codes are automatically outpulsed by the system when outgoing calls are made on a specialized carrier trunk circuit.	N/A
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Feature Name:	Description:	Support while the set is on the secondary controller
ACD Agent Hot Desking	Allows an agent to log into any ACD set and have the system apply the agent's personal phone profile to that ACD set.	Yes
ACD Dial out of Queue	Allows user to exit the ACD queue to perform another action. For example, you can exit the ACD queue to leave a voice mail for callback.	Yes
ACD Hold Retrieve/Abandon Event	Previously ACD Real Time Events did not report when a Non-ACD call was answered on an Agent phone and then placed on hold to be retrieved at another set. Currently, enabling Feature Level 3 and ACD Real Time Events modifies the reporting of the Hold Retrieve and Hold Abandon events. Requires: ACD Real Time Events (MSA-A-54) and Feature Level 3 (PN 54000510)	N/A
ACD 2000® Extended Agent Skill Groups	When this option is enabled, the maximum number of agent skill groups increases to 256 for the MXe Server and 128 for all other controllers. Each group can support up to 500 agents.	N/A
ACD 2000® Skill-based Routing	Each agent in an agent group is assigned a skill level. Calls to the group are routed to the most skilled available agent. If agents of equal skill are available, the call is routed to the longest-idle agent. To facilitate skill-based routing, agent IDs can appear in more than one agent group.	N/A
ACD Make Busy Reason Codes	ACD agents enter a reason code when phones are put into a Make Busy state.	No
ACD Real Time Event	Real time event records are used to monitor and record the activity of the ACD operation. Events are divided into two groups: call events and group statistics events. Call events report on individual ACD agent activity. Group statistics report on ACD group activity such as number of calls queued, longest waiting call, and number of active agents.	N/A
ACD Silent Monitor	Allows a supervisor to listen to an agent's telephone conversation, with or without the agent's knowledge. The supervisor can monitor an individual agent or a group of agents (hunt group). This feature uses a conference circuit, providing the supervisor with a one-way audio path into the conversation. The monitor acts like any normal conference except the supervisor's transmit path is not connected, thus preventing the agent or the customer from hearing the supervisor. A Silent Monitor can be performed on two-party conversations or conferences. Supervisors may also tape a particular agent's conversations. This feature can also be used to monitor non-ACD sets, including ONS, SIP, and external hot desk user sets.	Yes
Alpha Tagging	Associates names with external numbers entered in the system telephone directory. Alpha Tagging is intended for (but not restricted to) jurisdictions that do not provide calling party name in incoming signaling from the PSTN.	No
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Feature Name:	Description:	Support while the set is on the secondary controller
ANI Display on Non-prime Lines	Displays ANI information on Non-prime lines for 5 seconds. If the number is not seen, it can be redisplayed by pressing the Superkey and then the line key that is ringing.	Yes
Add Held	Allows you to move a call on Hold to another line, form a conference with a call on hold, or add a call on hold to an existing conference.	Yes
Advanced Analog Networking	Provides calling line identification and travelling class marks across T1/D4 trunks.	Yes
Advanced ARS	Allows you to program day and time zones, route plans, and ARS assignment.	N/A
Advice of Charge	Allows the caller to determine the cost of a toll call.	Yes
ANI/DNIS/ISDN Number Delivery	Automatic Number Identification and Dialed Number Identification Service identify numbers that are transmitted on an incoming trunk.	N/A
ANSWER PLUS® Automatic Attendant	Allows an external caller to dial through to an extension without going through an attendant. See also Multi-level Auto Attendant.	N/A
ANSWER PLUS Automatic Call Distribution II (ACD 2000)	Consists of four main components: call distribution, agent mobility, management and reporting, and feature configuration and administration.	N/A
ANSWER PLUS - Mitel Call Distribution	Permits the use of Recorded Announcement Devices (RADs) and a uniform call distribution to hunt groups.	N/A
Attendant Bulletin Board	Posts information for other attendants (for example, speed dial numbers). All 5550 IP Consoles on the system, that have a network connection, share bulletin board.	Yes
Attendant Busy-Out (Console)	Places your attendant console in a busy-out condition (absent status) under certain circumstances. In the busy-out condition, incoming calls are automatically rerouted.	Yes
Attendant Busy-Out (Station)	Allows you to busy-out a specific station by using the attendant console.	N/A
Attendant CAS Interface	Centralized Attendant Service interface allows an MCD system to be a remote node for a CAS site. CAS is an attendant call-handling service provided at a central office switch for calls from both public and private networks.	N/A
Attendant Call Answering Priority	Allows you to assign priority to calls based on origin when multiple calls are waiting; the call with the highest priority is answered first.	Yes
Attendant Call Information Display	Provides the attendant with information about called and calling parties.	Yes
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.	Yes
Attendant Conference	Allows the attendant to set up one or more conference connections between central office trunks and internal stations.	Yes
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Feature Name:	Description:	Support while the set is on the secondary controller
Attendant Consoles (Multiple)	Provides support for Multiple Attendant Consoles.	Yes
Attendant Console Firmkeys	Allows firmkeys to be programmed as one of the following feature keys: Phonebook, Guest Services (Hotel/Motel), Trunk Status, Alarm, SMDA, Select Option, Bulletin Board, Emergency Call Log, Guest Services, Help, Message Waiting, Operator Mode, Pager, Phone Book, Scratch Pad, Third Party application, Tones, TrkGrp Status, Voice Mail or blank (no application).	Yes
Attendant Console Status Display	Displays various parameters such as Day/Night Service, Attendant Status, and Alarm Status.	Yes
Attendant Directory Number	Allows you to dial a number (typically "0") to reach the attendant. Separate directory numbers can be programmed for each attendant console.	Yes
Attendant Help	Provides online assistance.	Yes
Attendant Hold	Allows you to temporarily place a call on hold so you can use other phone features.	Yes
Attendant Identity Information Display	Allows you to view the console's prime directory number, the Phone Book software version, and the console's hold slot number. This feature applies to the SC1000 only. From the 5550 IP Console, you can view the system software version.	N/A
Attendant Language Selection	Enables attendant to choose the language of operation for the attendant console. The 5550 IP Console supports the following languages: • English • French • EU Spanish (Europe) • LA Spanish (Latin America) • Dutch • Italian • German • PT Portuguese (Europe) • Romanian • Swedish • Polish. Note that an attendant's language selection is preserved when the MCD system undergoes an update or restore.	Yes
Attendant Messaging	Allows you to activate a message-waiting condition on a station from the attendant console. The condition can be queried or cancelled by the attendant or by a station user with the appropriate Class of Service.	Yes
Attendant Metered Calls	Allows you to use the attendant console to track the cost of outgoing trunk calls.	Yes
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Feature Name:	Description:	Support while the set is on the secondary controller
Attendant New Call Tone	Provides audio notification of new calls to the attendant console.	Yes
Attendant Position Busy-Out	See Attendant Busy-Out (Console).	Yes
Attendant Recall	Automatically alerts the attendant when either a trunk call to an idle station or a call on hold at the console has not been answered within a specified time period.	Yes
Attendant Ringer Control	Allows you to mute the attendant console ringer. When the ringer is muted, the Call Waiting indicator at the top of the display alerts you to incoming calls.	Yes
Attendant Scratch Pad	Functions as your personal telephone directory and speed dial list. You use it to save telephone numbers for faster dialling or to store the names and numbers of callers for future reference.	Yes
Attendant Serial Call	Automatically returns a call to the attendant console when the call ends.	Yes
Attendant Setup and Cancellation of Station Features	Allows the attendant to set up and cancel certain station features such as Call Forward, Do Not Disturb, Callback, and Reminder.	No
Attendant System Login	Requires the attendant to log on to the system to access certain programming functions from the attendant console.	N/A
Attendant Tone Signaling	Allows the attendant to send tones over the circuit once a call has been established.	Yes
Attendant Trunk Group Busy Status	Allows the attendant to display and/or print the busy status of the system trunk groups from the attendant console.	Yes
Audio Files Update	You can upload audio files to the MCD system and use them for embedded Music on Hold, all Auto Attendant greetings, set greetings, and RAD greetings. You can upload an audio file to a single MCD system by using the System Audio Files Update form, or to multiple MCD systems by using Enterprise Manager.	No
Auto-Answer	Automatically answers calls that ring your Prime line. This is typically used in an ACD environment.	No
Auto-Hold	Automatically places an active call on hold when you press a line key to originate or receive another call.	Yes
Automatic Route Selection (ARS)	Simplifies local and long distance dialling by automatically selecting the most convenient and cost-effective route for the call and by inserting and/or deleting the proper routing digits.	Yes
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Feature Name:	Description:	Support while the set is on the secondary controller
Bandwidth Management	Measures and manages bandwidth consumption by the VoIP media stream. This feature allows you to perform the following functions for the voice data packets at predetermined bottleneck points in the network: • Measure and report consumed and available bandwidth • Establish maintenance alarms when bandwidth consumption exceeds configured threshold levels • Provide Call Admission Control, that is, the rejection of new calls through a specific bottleneck point when consumed bandwidth exceeds maximum configured levels.	No
BRI (Basic Rate Interface)	A basic ISDN service consisting of two 64Kbps channels and one 16Kbps channel. Basic Rate Interface is supported on a 3300 ICP by the Quad BRI Framer Module.	N/A
Broadcast Groups	See Groups-Key System and Multicall.	Yes
Broker's Call	Allows you to temporarily suspend a telephone call while you originate a new one. Once the new call has been established, you can alternate between the two calls.	Yes
Busy Dial Through	Allows you to dial a feature access code sequence when a busy condition is encountered. See Callback and Camp-on.	Camp on – Yes Callback when on secondary or callback destination on secondary - No.
Calculator	Allows you to use your telephone as a basic four-function calculator by using the telephone keypad, display and softkeys.	No
Callback	Allows you to request that the system notify you when a busy line becomes idle or when an unanswered station goes off-hook and on-hook.	No
Callback for EHDU	Eliminates or reduces tariffs that External Hot Desk Users are charged for calls to system. Callback works by disconnecting the user's call, and then calling the user back within a few seconds. On answering, the user is presented with dial tone and can then dial the required number.	No
Callback – System Programmable	Allows you to program the destination of a matured callback set against a key line or multi call line group.	N/A
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Feature Name:	Description:	Support while the set is on the secondary controller
Call-by-call Service	With Call-by-Call Service, access channels do not have to be dedicated to specific services such as OUTWATS or 800 services. This enables the customer to reduce facilities and integrate dedicated and switched, inbound and outbound, voice and data traffic on a single facility. It also allows a business with calling peaks to dynamically allocate coverage across channels so that access lines are optimized. This implementation ensures that incoming calls are not turned away because all incoming channels are busy while adjacent outgoing channels are idle.	Yes
Call By Name	See Phonebook.	Yes
Call Coverage	Provided through a combination of features: Call Rerouting, Call Forward, Do Not Disturb, and Answer Plus-Mitel Call Distribution.	Yes for all features except DND
Call Duration Display	Displays the call duration for incoming and outgoing calls, in one minute increments (starting at 0:00).	Yes
Call Forward	Allows you to redirect incoming calls to an alternate number.	Yes (features and access keys)
Call Forward -Cancel All	Allows you to cancel all types of Call Forward.	No
Call Forward Delay	When the Call Forward - Busy feature is activated on a phone, a call to that phone can be delayed at a busy extension. A person on the phone receives a warning that there is another call waiting. The telephone can either be set to display the name of the waiting caller, or provide interrupted dial tone.	No
Call Forward -Follow Me-End Chaining	Ensures that calls are not further redirected.	Yes
Call Forward -Follow Me-Reroute When Busy	Forwards the call to the original set's First Alternative Rerouting if the call forward destination is busy.	No
Call Forward -Forced	Allows you to manually redirect an incoming call on your prime or private line to another number.	No
Call Forward Group	Allows you to forward group and prime lines to different locations.	No
Call Forward Out of Service	This feature behaves like Call Forward No Answer. If no destination is programmed, calls are handled as if the phone is not installed.	No
Call Forward - Override	Allows you to bypass or override any Call Forward condition that is set at the station that you are calling.	Yes
Call Hold	See Hold.	Yes
Call History	Call History keeps track of the names (if available) and telephone numbers of missed calls, unanswered outgoing calls or external answered incoming or outgoing calls. It allows the user to view and quickly place a callback. This feature is supported on the 5330/5340 IP Phone and the Unified Communicator Advanced Softphone.	Yes
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Feature Name:	Description:	Support while the set is on the secondary controller
Calling Line Identification	The telephone number of the calling party is transmitted to the Mitel PBX and can be sent to devices within the system.	Yes
Caller Line Identification Presentation (CLIP)	Allows ONS CLIP sets using CLIP protocol to receive Caller Line Identification Delivery (CLID) information and the time and date of a call. There is no CLIP support for the ASU (UK).	N/A
Call Park	Allows extension users and attendants to park calls and automatically initiate a page to announce the call to the requested party. Formerly, only the attendant could park calls (with no automatic paging) for extensions to retrieve. See also Group Park.	Yes
Call Pickup	Allows you to answer an incoming call that is ringing at another station.	Yes
Call Pickup - Clustered	Provides Dialed Call Pickup functionality across a cluster.	Yes
Call Privacy	Protects a call from audible Call Waiting tones, as the result of a camp-on, and prevents intrusion of any kind (for example Busy Override).	Yes
Call Recognition Service for EHDU	Simplifies or eliminates log-ins for External Hot Desking Users by authenticating them based on their calling line ID.	No
Call Release	See Release.	Yes
Call Rerouting	Redirects calls to alternate answering points or devices under specified conditions. May be used to redirect calls always (in Day, Night 1, and/or Night 2 mode) or under busy, no answer, or Do Not Disturb conditions.	Yes
Call Split	See Conference Split.	Yes
Call Swap	See Swap.	Yes
Call Transfer	See Transfer.	Yes
Call Waiting Swap	Allows you to use the switch hook to alternate between two calls when parties are in Call Waiting for your station or when you have a call on Consultation Hold.	Yes
Called Party Features Override	Allows calls from an extension to override any call redirection features, such as call forwarding, that are enabled on the destination extension. If this feature is activated before a call is made to an extension and the call is unanswered, the call remains ringing on the extension.	Yes
Camp-on (Call Waiting)	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.	Yes
Camp-on Tone Security	Prevents you from hearing Camp-on tone. If any party in a call has this option enabled, no Camp-on tone is sent to anyone in the call.	Yes
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Feature Name:	Description:	Support while the set is on the secondary controller
Centralized Attendant Service (CAS) interface	See Attendant CAS Interface.	N/A
Centrex (Flash and Double Flash over Trunk)	Provides the ability to send a double switchhook flash out over a trunk. Flashing over a trunk enables a telephone on the PBX to use CENTREX features.	Yes
CLASS (Customer Line Access Subscriber Services)	Allows the system to receive Calling Line ID digits or CLASS name on CLASS sets.	N/A
CLASS Station Side Software Support.	Enables ONS CLASS sets using the CLASS protocol to receive caller line identification delivery (CLID) information.	N/A
Class of Restriction	Limits a station's access to specified numbers. A station may have three CORs (Day/Night1/Night2 service). The COR may also be changed by using a Verified Account Code.	Yes
Class of Service	Defines a station or trunk's feature and timer options. A station or trunk may have three COSs (Day/Night1/Night2 service). The COS may also be changed by using a Verified Account Code.	Yes
Clear All Features	Allows you to cancel the features that are activated on your extension or another user’s extension.	Yes (also for Remote Clear All Features)
CLI Substitution	Allows the PBX/BRI extension number to be appended to the outgoing CLI.	Yes
Clustered Hospitality	Provides hotel/motel feature functionality across a cluster of 3300 ICPs. The cluster comprises a single Hospitality Gateway ICP and one or more Hospitality ICPs.	Resiliency support in a hospitality application is limited to devices only; guest services (wake-up calls, room status information, suite services etc.) are not resilient.
Compression	Allows IP calls to utilize less bandwidth than an uncompressed call.	Yes
Conference	Allows you to connect three or more calls into a single telephone conversation. While you are in a Conference, you can use any of the features that would normally be available during a two-party call.	Yes
Conference Split	Allows you to separate a 3-party conference so that two of the parties can speak privately, while the other is placed on Consultation Hold.	Yes
		Page 9 of 25

Feature Name:	Description:	Support while the set is on the secondary controller
CPN Substitution	Allows you to send a substitute directory number for the calling party's DID number to the network (rather than sending the actual DID). You can define CPN substitution for individual DID numbers or ranges of DID numbers.	Yes
Credit Limit Support	The PMS uses a Credit Limit message to inform the PBX of a specific room or suite's telephone credit limit. The PBX uses an Alert message to notify the PMS when the established telephone credit limit has been reached. The PMS may then send a Station Restriction message to the PBX to apply previously programmed Class of Restriction parameters (calls in progress are not affected when a credit limit is reached). The PBX does not make any call restriction decisions; the PMS is solely responsible for informing the PBX of any action to take in regards to credit limit exhaustion. Emergency Services (911/999) and internal calls are never restricted.	Yes
DASS II Voice I	Allows basic calls to be made from the system to a DASS II protocol Central Office, using CEPT Digital Trunks and DASS II signaling.	Yes
Date and Time	Set through the System Administration Tool. This data appears on all Station Message Detail Recording (SMDR), traffic measurements, data dumps, display telephones, and attendant consoles.	Yes
Day/Night Service Control	Allows you to redirect calls to alternate answer points for individual trunks. Answer points can vary according to the selected mode of operation (Day, Night 1, or Night 2).	Yes on Consoles, No on sets
Destination-based Call Display	Displays the name of the destination hunt group. When individuals are assigned to different hunt groups, they can still answer calls appropriately, based on the display.	No
Dial Tone	Users normally hear continuous dial tone when they lift the handset. They hear discriminating (also called interrupted), or transfer dial tone under certain conditions.	Yes
Dial Tone -Outgoing Calls	The system can provide a pseudo-CO dial tone to prevent possible confusion to station users.	Yes
Dialed Number Editing	Allows you to edit numbers during dialing.	Yes
Dialing -Conflicting Numbers	The system can differentiate between conflicting numbers such as 1-0-0-0-0 and 1-0-0-0. In this example, if the fifth digit is not dialed within a time-out period, the system assumes that the dialed sequence is complete and makes the call.	Yes
DID Single Ring Cadence	Gives single ring back to outside callers.	N/A
Direct-In Lines (DIL)	Allows incoming trunks to be assigned to a specific station or hunt group so that calls from the trunk ring the station or hunt group directly.	Yes
Direct Inward Dialing (DID)	Permits incoming calls on designated trunks to directly access predefined stations (or other answering points) on the system.	Yes
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Feature Name:	Description:	Support while the set is on the secondary controller
Direct Inward System Access (DISA)	Allows external callers to access the system by using a special trunk. The system sees the DISA trunk as a station with its own Class of Service and Class of Restriction. Calls that enter the system on DISA trunks have access to a variety of system features. In all cases, the DISA trunk can be assigned account codes to provide a high degree of security or additional options.	Yes
Direct Outward Dialing (DOD)	Allows you to make external calls without attendant assistance.	Yes
Direct Page	Allows you to page another telephone over its built-in speaker. See Off-Hook Voice Announce.	Yes
Direct Station Select/Busy Lamp Field (DSS/BLF)	A Busy Lamp Field (BLF) allows the status of a directory number to appear on the line status indicator of a telephone or Programmable Key Module. The monitored device may be on the same system or another system within the same cluster. The key associated with the busy lamp acts as a Direct Station Selection (DSS) key.	Yes
Direct Voice Call	Allows you to establish a two-way handsfree call at the called party set whether or not Handsfree Answerback or Auto-Answer is enabled.	Yes
Disable Send Message	Allows you to disable the send message key function on certain sets, through class of service.	Yes
Display Caller ID on all Lines	Provides Caller ID on other lines when idle (shows any ringing lines), and when the user is talking (priority based on key position).	Yes
Display Contrast Control	Allows you to adjust the contrast of the alphanumeric display on your phone.	Yes
Display Identity of Ringing Non-Prime Line Keys	Allows users of SUPERSET display telephones to display the calling line identifier of ringing non-prime keys on their sets.	Yes
Display of Name and Number	Displays name and number and offers the ability to switch between displays.	Yes
DNI	Allows the programming of Mitel digital devices.	N/A
DNIC as a RAD	DNIC ports may be programmed as Recorded Announcement Devices (RADs). When a DNIC port is programmed as a RAD, the device capabilities are limited to those of a RAD.	Yes
Do Not Disturb	Allows you to place your set in an apparent busy condition without affecting the outgoing functionality. If people call your set while DND is activated, they hear a special busy tone.	No
DTMF Keypad Support	Allows ONS/OPS extensions to use all 16 keys on a 4x4 DTMF keypad. The additional row of four keys (ABCD) is used to access features in the system.	N/A
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Feature Name:	Description:	Support while the set is on the secondary controller
Dual PKM 48 Support	The Programmable Key Module 48 (PKM48) provides 48 additional feature keys for telephones. Each feature key has a Line Status Indicator that behaves the same way as those on a phone. A second PKM48 can connect to the first to provide for a total of 96 additional feature keys.	Yes
Emergency Services	Allows an Emergency Services number to be dialed, which sends a Customer Emergency Services ID (CESID) from the system to the Public Safety Answering Point (PSAP). The CESID is used as a key in the Automatic Location Information (ALI) database to retrieve a database record indicating the precise location of the caller.	Yes
Feature Keys	Allows you to activate features without dialing feature access codes.	Yes. See 3300 ICP Resiliency guide
Flash -Calibrated	Allows you to generate a Switchhook Flash with a precise time interval.	No
Flash -Switchhook	Allows you to place a call on Consultation Hold and return to dial tone so that you can invoke station features.	No
Flash -Trunk	Allows you to single- or double-flash a trunk in order to access Centrex™ features.	No
Flexible Answer Point	Allows station and console users to program a night answer point for their incoming trunk calls.	No
Flexible Dimensioning	Allocates database memory to each feature resource. The amount of memory determines the maximum size of the feature resource; the system borrows memory from other resources that are not in use. This feature allows individual systems to be tailored to individual business needs, resulting in optimal performance for a particular system.	N/A
Forced Non-Verified Account Codes	Customers such as law firms require ways of tracking calls for billing purposes and need the ability to enter a number (account code) as a record for a call. These numbers do not have to be "verified", as the number might only be valid for the duration of a case. But they must be "forced" in order to ensure that an Account Code can be used as a billing tracking mechanism (tracked in SMDR record). The solution is to have the ability to use a Forced Non-Verified Account Code.	Yes
Ground Button	Allows you to place a call on Consultation Hold and return to dial tone to invoke station features. The Ground Button provides an alternate method of producing a Switchhook Flash.	N/A
Group Listen	Allows you to carry on a conversation using the handset or headset while allowing others nearby to listen to the person at the far end over the handsfree speaker.	No
Group Page	Allows you to page a group of phones over their built-in speakers.	Yes
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Feature Name:	Description:	Support while the set is on the secondary controller
Group Park	Group Park is a variant of Call Park that uses a single feature key to both park and retrieve calls. Call indication is provided to all members in the group.	Yes
Groups - Key System and Multicall	Allows multiple telephones to share the same extension number. Incoming calls ring at all of the idle stations, and the stations stop ringing when one group member answers the call.	Yes
Group -Presence	Allows group members and answer points in groups (Voice hunt groups, Name Tag hunt groups, Ring Groups, Personal Ring Groups, and ACD agent groups) to be easily made “present” (i.e. included) or absent from the group. Only members who are present in a group are offered calls directed to that group. Group Presence employs COS so that administrators or end users can be granted control depending on the specific application. For example, in the case of a Personal Ring Group, a user would likely be granted the ability to opt an answer point in or out of his/her group. However, in the case of an ACD agent group, the control to make agents present may be given to supervisors or agents depending on the application. Feature access keys can be programmed to enable simple toggling between present and absent. Presence can also be controlled through FACs, the 3300 Desktop Tool and MiTAI.	Yes
Group Silent Monitor	See ACD Silent Monitor.	Yes
Handset Receiver Volume Control	Allows you to adjust the volume of the handset receiver.	Yes
Handsfree Operation	Allows you to use your telephone without lifting the handset.	Yes
Headset Operation	Allows you to use a Headset to make and receive telephone calls.	Yes
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.	Yes
Hold on Hold	Allows both parties of a two-party call to put the call on hold.	Yes
Hot Desking	Hot Desking allows a number of users to share one or more Hot Desk-enabled IP sets. To use a Hot Desk set, the user logs in using a Hot Desk DN and PIN. Once logged in, the user can: • Receive incoming calls at the set • Place outgoing calls • Retrieve voice messages • Program and use feature keys. Hot Desking is ideal for telecommuters, sales agents, and other employees who spend only part of their time in the office. With Hot Desking, a company does not have to provide a dedicated phone for each of these employees. Instead, the company can make a pool of shared phones available for users.	Yes
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Feature Name:	Description:	Support while the set is on the secondary controller
Hot Desking -External	Allows users to configure any external telephone number (e.g. mobile phone, home phone) as a Hot Desk. When the Hot Desk user is not logged into one of the system's Hot Desk sets, the system automatically routes the call to the external telephone number. As a system extension, the external device user has access to extension dialing along with other system resources such as voicemail. Coupled with "Presence" it enables the presence of the external number to be treated the same as an internal number. Support for External Hot Desking continues while the set is on the secondary controller.	Yes
Hotel/Motel	Provides a property-management interface and features commonly used by hotels, motels, and hospitals.	No
Hotline	Automatically dials a designated answer point when you go off-hook. The answer point can be another extension, an attendant, a trunk, or a hunt group.	Yes
Hunt Groups	Allows you to define a group of extensions under a pilot number; calls to this number ring the first idle extension in the group. You can directly access any phone within a hunt group by dialing its unique extension number.	Yes
Hunt Groups - Networked	Provides hunt group functionality across a network or cluster. See 3300 ICP Resiliency Guidelines for more details.	Yes
Intercept Handling	Allows the system to control what happens to a call when it cannot be completed as dialed. Such a call may be routed to a tone or to a directory number; two destinations can be programmed for either condition.	Yes
Interconnect Restrictions	Restricts access to certain trunks, stations and equipment (such as data communications equipment). Interconnect restrictions are a function of the direction of the call. Every peripheral device is assigned an Interconnect Number that prevents it from connecting with another.	Yes
Interconnect Restriction Override	Allows 911-access to telephones in a hotel environment that must be restricted from dialing various internal numbers.	Yes
IP Networking	Allows calls to be placed or received over an IP trunk.	Yes
ISDN PRI	The Universal NSU (dual link) provides an interface between users (voice or data) and the ISDN Primary Rate Interface (PRI) services offered by the Network Service Providers.	N/A
Keep TelDir Entry on Check Out	Ensures that the telephone directory entry associated with a particular room or suite extension is unchanged upon check out.	Yes
Key System Groups	See Groups-Key System and Multicall.	Yes
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Feature Name:	Description:	Support while the set is on the secondary controller
LLDP-MED	Link Layer Discovery Protocol-Media Endpoint Discovery (LLDP-MED) is an open standard extension of the LLDP core standard used by endpoint devices to discover each other on the same network link (segment). Certain Mitel IP Phones can use LLDP-MED to obtain the VoIP-specific configuration information that they require to operate in a converged network—information such as VLAN ID, COS Priority, and DSCP values.	N/A
Language Change	Provided they are made available by the system administrator, this feature allows the user to change the language of their set's telephone prompts and softkeys to any one of the following languages: • English • French • EU Spanish (Europe) • LA Spanish (Latin America) • Dutch • Italian • German • PT Portuguese (Europe) • Romanian • Swedish • Polish • Chinese (5312, 5324, 5330 and 5340 IP Phones only) Note that a user's language selection is preserved when the MCD system undergoes an update or restore.	No
Line Types and Appearances	Allows an administrator to program any of the programmable keys on a phone as line appearance keys for single or shared lines (up to 32). There are three types of lines: Prime, Non-Prime, and No Where Prime.	Yes
Line Appearance Ring Types	Line appearances can be programmed to ring in a variety of ways.	Yes
Maintenance	The system provides extensive maintenance coverage periodically testing all types of peripheral hardware. Maintenance users may test individual circuits on demand.	N/A
Malicious Call Trace	The Malicious Call Trace feature provides network-wide tagging capability of malicious calls. The Malicious Call Trace feature provides a record of malicious calls in the SMDR record. Malicious calls can be recorded using the Record a Call feature (when available).	Yes
Meet Me Answer	Allows a paged party to respond to a Group Page without knowing the identity or location of the paging party.	Yes
Message Board	Provides a method for administrators to communicate with each other on the System Administration tool.	No
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Feature Name:	Description:	Support while the set is on the secondary controller
Messaging-Advisory	Displays a short advisory message to display-set users who call your telephone.	No
Messaging-Callback	Allows you to leave a callback message on a telephone when the called party is busy or does not answer. When you receive a callback message, you can review the message on the display (if applicable) and/or call the sender back.	Yes
Messaging-Dialed	Allows you to leave a message-waiting indication on a telephone. When you receive a message-waiting indication, you call your message taker to accept the message.	Yes
Mixed Station Dialing	Allows you to use DTMF telephones within the system and on the same line.	N/A
MNMS	Supports OPS Manager functions.	N/A
MSDN/DPNSS	A digital signaling system that provides many features and is used within a private network of PBXs.	N/A
MSDN Release Link Trunk	Allows the attendant to make an outgoing call on an incoming trunk. It provides centralized attendant service by allowing attendants on the attendant system to reroute calls without tying up additional trunk resources.	N/A
Multicall Groups	See Groups-Key System and Multicall.	Yes
Multiple Consoles	See Attendant Consoles (Multiple).	Yes
Multi-Level Auto Attendant	Allows a hierarchical menu to be programmed on the auto attendant. This provides callers with better self-service access to the person or department they are calling.	N/A
Multi-Level Precedence and Preemption (MLPP)	Supports emergency communications for the military as part of the Defense Switched Network (DSN). MLPP allows authorized users to • specify a precedence level when they make a call • preempt calls that have a lower precedence level.	Yes
Music	Allows you to listen to the Music On Hold music source through the speaker on the telephone.	Yes
Music On Hold	Music On Hold provides callers with music or information while they are waiting for a call to be completed. Music On Hold is provided when a call is on Hold, transferred to a busy party, or camped-on to a station. The music or information source is provided by the customer. There are three types of Music on Hold: • Analog Music on Hold • Digital Music on Hold • Embedded Music on Hold (allows systems to use embedded .wav files as music sources).	Yes
Music On Hold Transfer	Allows external callers who are transferred to a set to hear Music on Hold while waiting for an answer. For a transferred call, the caller hears Music On Hold until the call is answered at the destination.	No
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Feature Name:	Description:	Support while the set is on the secondary controller
Name Suppression on Outbound Calls	Allows callers to block the name of the caller from the ISDN network even if the name is programmed in the telephone directory.	Yes
Simple Network Time Protocol (SNTP)	The 3300 ICP supports a client for Coordinated Universal Time (UTC) distribution. Administrators benefit from automatic synchronized clocks for all 3300's in a system, automatic updates for daylight savings time, and descriptive timestamps and logs (NOTE: Clock synchronization on MCS systems is provided by the Mitel Standard Linux operating system.)	N/A
Networking	The system supports both analog and digital networking. See Node ID Recognition and Uniform Numbering Plan.	N/A
Networking using MSDN/MSAN	MSDN/DPNSS provides fast call setup capabilities and feature transparency across the network. No significant difference between making a local call and a network call is apparent to the user All of the MSDN networking packages require that each PBX has MSDN Voice I or MSAN installed.	N/A
Networked ACD	Supports ACD functions over a Mitel Switched Digital Network (MSDN). Agent skill groups at different locations (on different systems) may service calls on the network independently of where the call entered the network.	No
Networked Group Page	Group Paging can be completed across a network or network cluster, allowing, for example, a set on system A to page a specific group on system B.	Yes
Network Selectable Music Source	Each site can select their own music source or a networked source from the originating PBX.	N/A
Night Service	Allows you to redirect calls to alternate answer points for individual trunks. Answer points can vary, according to the selected mode of operation (Day, Night 1, or Night 2). A key appearance may be programmed to indicate if the MCD system is operating in Night Service mode.	Yes
Night Service - Scheduled	Allows administrators to schedule Night Service modes on the MCD system. This scheduling allows transitions between all the supported service modes (Day, Night 1 or Night 2) at independent times.	Yes
Night Service - Automatic	Automatically places the system into Night service if all attendant consoles are unable to receive calls or if all attendant consoles are inactive when the time-out period has expired.	Yes
Node ID Recognition	Enables a system in a network to determine whether an incoming call applies to it or to another system in the network.	N/A
Non-Busy Station	Allows you to program an extension to never return a busy tone. This feature is used for special situations such as emergencies. A non-busy extension can originate calls if it is also programmed as a Hotline extension.	No
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Feature Name:	Description:	Support while the set is on the secondary controller
Non-DID Extension	Allows the system to support phones that are not directly accessible to DID trunks. Calls to and from these phones are transferred to non-DID extensions by an intercept handling point (such as an attendant or a station).	Yes
Off-Hook Detection to Display sets	Used in hospitals and nursing applications. If someone fails to complete dialing, the alert is sent to a set.	Yes
Off-Hook Voice Announce	Allows you to receive a direct page during a handset or headset call. See Direct Page.	Yes
ONS Ports as Music Sources	Allows a music source, either the system's Music on Hold source, or an ACD alternate music source, to be an ONS port instead of a DMP module. The Alternative Recording Device (ARD) is an off-hook ONS port that connects to callers in a listen-only conference. The user decides what is supplied on the ONS port - silence, music, or endless loop recordings. **Note:** An ARD should not be used as a first-level announcement (Music On Hold, for example). Eliminating or reducing the number of DNIC circuits and DMP modules translates into cost savings for the organization.	N/A
Overlap Outpulsing	Reduces post-dialing delay when trunk calls are originated. Once ARS has determined a route, a trunk is seized and tones are outpulsed to the CO. These pulses are sent before the user has finished dialing to allow faster call setup on analog trunks.	N/A
Override	Allows you to enter a conversation at a busy station or ring a station with Do Not Disturb activated. Before you enter the conversation, all parties receive a warning tone.	Yes
Override Security	Prevents users from using Override on your station.	Yes
Paging	Allows you to connect to loudspeaker/paging equipment to access individual paging zones or all paging zones simultaneously. Before you are connected to the paging equipment, you hear a two-second burst of tone.	Yes
Permanent Do Not-Disturb	Allows an extension to be placed in a permanent busy state.	N/A
Phonebook	Allows you to locate and call a system user based by name, extension number, department, and/or location.	Yes
Phone Lock	Phone Lock locks a set preventing access to the majority of features, with the following exceptions: unlocking the set via a user PIN, Hot Desk Login and Logout support, and Emergency Call Notification support. Phone Lock has no effect on incoming calls but restricts outgoing calls, with the following exceptions: calls to emergency trunk routes and local operators.	Yes
PRI (Primary Rate ISDN)	Protocol supported by the T/E1 Modules. PRI supports features such as Min/Max, Automated Min/Max, NFAS (Non-Facilities Associated Signaling), D-channel Backup, and Remote LAN Access.	N/A
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Feature Name:	Description:	Support while the set is on the secondary controller
Printer Support	The system has complete RS-232 printer flexibility. Any printer port may be programmed for any application. The system supports system printers both for its own applications (such as SMDR and maintenance) and as dedicated data communications printers.	N/A
Priority Queuing	Handles calls in order of priority. When waiting for calls to be completed internal or external callers are placed in a queue and assigned an access priority.	N/A
Privacy Release	Call privacy between users who share line appearances in key systems groups is automatic. The privacy release feature allows users to release privacy during a call to include another member of the key system group in the call.	Yes
Private Line Automatic Ringdown	Provides rapid connections between devices, primarily 5560 IPTs used by securities and commodities traders.	Yes
Programmable Key Modules	Provide telephones with additional personal keys.	Yes
Property Management System (PMS)	A PBX feature that allows the hospitality industry to connect their Hotel PMS systems to the PBX via an IP interface or serial interface. This connection allows the PMS to notify the PBX when a user checks in or checks out.	Yes
Q.SIG	A protocol that allows you to connect a minimum of two systems together to form a virtual private network. Q.SIG is supported by the T1/E1 Modules for both incoming and outgoing calls. **Note:** Resiliency does not work over QSIG (NSIs not passed)	N/A
RAD Support	Recorded Announcement Devices (RAD) are supported in the system as recording hunt groups. These special hunt groups support features and restrictions that allow efficient use of the recording resources. Recording hunt groups are used in ACD, UCD, Hotel/Motel Wakeup, Automatic Attendant Overflow and Automated Attendant.	N/A
Range Program Trunks	Allows installers to select a consecutive range of trunk circuits. The system automatically assigns sequential trunk numbers to those circuits. Also copies parameters from the first programmed trunk including Class of Service, Day, Night1, Night2 and Circuit Descriptor Number. Trunk Name and Comments are left blank.	N/A
Recall	Allows an incoming caller, who has been transferred to an idle station and not answered within a specified time-out period, call back the last party who handled the call. Similar time-out recalls occur for parties who are transferred to busy stations or placed on hold.	Yes
Recall Button	See Ground Button.	N/A
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Feature Name:	Description:	Support while the set is on the secondary controller
Record-A-Call	Record-a-Call allows extension users to record a two-party call when one party is connected to a trunk. The recorded conversation is stored in the users' voice mail mailbox. You can configure this feature to automatically record incoming calls when the call is answered at the extension, record external outgoing calls that are made from a system extension to the PSTN, and record both incoming calls and external outgoing calls for the same extension.	Yes
Redial	Automatically dials the last manually dialed number.	Yes
Redial -Saved Number	Allows you to save a number for future dialing. The number remains saved until a replacement number is saved.	Yes
Release	Allows you to release from an attempted connection to an external party without going on-hook. Release is useful when you encounter a busy or unavailable external party that you are attempting to add to a conference.	Yes
Reminder	Allows you to program your set to ring and provide a message at a specified time within a 24-hour period.	No
Remote Wake-up Calls	Wake-up calls can be set or cancelled remotely from a telephone or attendant console using the Hotel/Motel Room Remote Wake-up Call feature access codes.	No
Reroute after Call Forward Follow Me to Busy Destination	This feature uses the class of service option Call Reroute after CFFM to busy destination. With this option set to YES, if the user programs call forward always and the call forward third party or group call forward destination is busy, the call follows the original called set’s programmed call reroute first alternative for busy. For example, a call arrives at station A that is call forwarded under one of the above stated conditions to station B. If station B is busy or does not answer, the call follows station A's First Alternative Rerouting. With the COS option set to NO, the call only follows set A's rerouting on a no answer condition. This functionality applies only to calls using call forward always; call forward third party or group call forward with the "forwarded to" destination being an internal party, another user across MSDN or calls forwarded externally via ISDN.	No
Resiliency (3300 ICP only)	Allows the IP Phones to re-home to a secondary controller if a 3300 ICP fails or is taken out of service. This ensures that there is no disruption in service. In addition, calls that are in progress when an outage occurs remain in progress and are not lost. Network administrators may configure IP Phone and IP Console resiliency from the System Administration Tool of the local element.	Yes
Ringer Control	Allows you to adjust the volume and pitch of the telephone ringer.	Yes
Ring Groups	Provides the ability to ring all members of a group simultaneously or sequentially.	Yes
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Feature Name:	Description:	Support while the set is on the secondary controller
Ring Groups -Personal	Provides the ability for a user to configure a collection of up to 8 answer points as a personal ring group. An incoming call to the Prime Number will simultaneously ring all devices in the group."One busy/All busy" may be configured for the group so that if one answer point is busy, they will all appear busy. Users also have the ability to "push" a call back to the ring group so that it may be "puled" (answered) by another device. "Push and pull" can be made quite simple for the user by pre-configuring a feature key for this purpose.	
Ringing -Discriminating	Allows you to distinguish between incoming internal calls, incoming trunk calls, tie line calls, and Callbacks by using different ringing patterns (cadences).	Yes
Ringing -Discriminating (Optional)	Allows you to change the Discriminating Ringing patterns on ONS/OPS lines so that you hear internal ringing (1 second on and 3 seconds off) for both internal and external calls.	N/A
Ringing Line Select	Allows you to answer any ringing line by going off-hook.	Yes
Scheduler	Allows you to schedule common events to run automatically. For example, you can create an event that switches the system to night service every weekday evening. The Scheduler includes a calendar that can be customized with holidays that are unique to your locality.	N/A
Silent Monitor	See ACD Silent Monitor.	Yes
SMDR -External	Collects data for outgoing and incoming trunk calls.	N/A
SMDR -Internal	Collects data for calls made between stations within the system.	N/A
SMDR Extended Reporting Level 1	Allows SMDR record format changes to accommodate: • International ANI digit strings • Attendant Line Appearances • Incomplete Internal calls (optional).	N/A
SNMP Agent	Simple Network Management Protocol (SNMP) governs the management and monitoring of network devices and their functions.	N/A
Speak@Ease™ Softkey Support	Provides quick and easy access to the Mitel Speech Server voice recognition system.	Yes
Speaker Volume Control	Allows you to adjust the volume of the phone speaker.	Yes
Speed Call -CDE	Allows users to speed dial telephone numbers that the administrator has programmed into the system. The administrator programs the number into a "CDE speedcall" key on a user's set through the Multiline Set Keys form. Users initiate the speed call by pressing the key.	Yes
Speed Call -Pause	When the system encounters a pause while dialing a speed call digit string, the system ceases dialing for the duration of the pause. Dialing resumes when the pause ends.	Yes
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Feature Name:	Description:	Support while the set is on the secondary controller
Speed Call -Personal	Allows you to store and dial frequently-used numbers using access codes and index numbers.	Yes
Speed Call -System	Allows you to dial stored system numbers.	Yes
Speed Call -User	Allows you to store external numbers under feature keys for faster dialing. You can press a Speed Call Key to dial a phone number or, during a call, to outpulse DTMF tones.	
Station Message Detailed Accounting (SMDA)	Allows the system to accumulate meter pulses (up to an assigned buffer size) that can be read, printed, and cleared from a console. You can collect meter pulses either with a device (device meter unit accumulation) or an account code (account code meter unit accumulation).	N/A
Station-To-Station Dialing	Allows you to dial any other station directly.	Yes
Suite Service	Allows you to group a number of telephone lines through interconnected hotel/motel rooms, or suites, for the purposes of billing and sharing telephone service. There are two kinds of suite services: • Single suite services • Linked suite services. Suites and linked suites allow you to specify a number of member extensions (1 to all) that ring simultaneously (up to 24 for linked suites). These extensions can be multi-member broadcast groups. Suites and linked suites require all member extensions to be defined on the same 3300 ICP.	No
Swap	Allows you to temporarily suspend a phone call to originate a new one. Once the new call has been established, you can alternate between the calls.	Yes
Switchhook Flash	See Flash-Switchhook.	No
System Access Authorization	Passwords control administrative access to the system. The installation technician assigns usernames and passwords for access to the different system tools.	N/A
System Alarm Indications	See Alarms and Attendant Console Status Display.	N/A
System Fail Transfer	Maintains telephone service in the event of system failure (such as during a power outage). When the system goes into SFT mode up to four POTS phones are connected directly to the Central Office via LS Trunks.	N/A
T1/D4	Provides support for T1 Channel Associated Signaling.	N/A
Tag Call	Provides a record of malicious calls in the SMDR record.	N/A
TAPI Support	Supports MiTAI and TALK TO® TAPI computer telephony interfaces.	No
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Feature Name:	Description:	Support while the set is on the secondary controller
Tandem Trunking	The system can transparently interconnect trunk circuits originating from one CO or PBX and terminating on another (tandem trunking), without attendant intervention.	N/A
Telephone Directory -Privacy Option	Any extension number in the system telephone directory can be designated as private. When an extension number is private, the number is not displayed on other users' phones.	Yes
Telephone Usage Restriction (Curfew Control)	Provides the ability to restrict calls based on the time of day. It is used in conjunction with existing Call Block (Hotel Motel functionality). When the curfew time is reached, users receive a warning tone indicating that calls in progress will be cleared down.	Yes
Tie Trunk Support	Tie trunks terminate at the attendant console, at station sets, in hunt groups, or on night bells. They may also be arranged as dial-in tie trunks or tandem trunks. Like CO trunks, tie trunks are arranged in groups.	N/A
Timed Reminder	See Reminder.	No
Toll Control	Allows or denies access to specified routes, CO exchanges, and directory numbers.	N/A
Tone Demonstration	Allows you to hear the tones provided by the system.	Yes
Tone Detection	The system can detect and analyze call progress tones that originate from the Central Office during the course of a trunk call.	N/A
Tone Plan Flexibility	Call progress and supervisory tones generated within the system are programmed to meet the requirements of the telephone authorities of the country in which the system is installed.	N/A
Traffic Reporting	Provides traffic reports of system usage to allow better system resource management.	N/A
Transfer	Allows you to move a call from one phone to another. Before completing a transfer, you can consult privately with the third party and swap between private conversations with each party.	Yes
Transmission Tests	Allows you to perform milliwatt, balance, and 100 tests on a trunk.	N/A
Travelling Class Marks	Travelling Class Marks (TCM) extend users access to features and services available to them on their host MCD system to other MCD systems in a cluster or network. TCM allow callers in a private network to access features and services based on the their Class Of Service (COS), Class of Restriction (COR), and Interconnect Number, rather than on those of the incoming trunk on the remote system.	Yes
Trunk Access	Allows you to directly access a specific trunk. No toll control or ARS checking is done when you use Trunk Access. This feature is used when a maintenance telephone is required.	Yes
Trunk Answer From Any Station (TAFAS)	Allows you to answer any call that rings a night bell.	Yes
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Feature Name:	Description:	Support while the set is on the secondary controller
Trunk Busy-Out	Allows you to busy-out a specific trunk. When you perform a Trunk Busy-Out, the trunk is busied out if it is idle; if the trunk is in use, it is busied out as soon as it becomes idle. When you busy-out the trunk, it cannot be accessed.	N/A
Trunk Group Busy Status	Enables attendants to query the status of trunk groups from the attendant console.	Yes
Trunk Group Hunting	Allows you to search for trunk groups in either a terminal or circular pattern. In a terminal trunk hunt group, trunks are selected in a predetermined order. In a circular hunt group, trunks are selected in a distributed manner (the first free trunk after the last one used becomes the new first choice).	N/A
Trunk Labels	May be assigned to individual trunks or groups of trunks. When a trunk call appears at an attendant console or set, the trunk label and trunk number is displayed.	Yes
Trunk Range Busy Out and Return to Service	Allows the installer/trouble-shooter to busy out and return to service an entire digital link. All trunks in the “Range Busy Out” must be on the same card. Trunk Range Busy Out and Return to Service is only available in maintenance mode. This reduces the amount of time required to troubleshoot programming or operation problems with digital trunks.	N/A
Trunk Select -Direct	Allows you to access an outside trunk for the purposes of originating and receiving external calls. Because the trunk is assigned to a line appearance, you can access the trunk to make or answer calls without trunk access codes.	Yes
Trunk Support	The system supports most public network trunk types (both analog and digital).	N/A
Two B-Channel Transfer (TBCT)	Allows you to transfer an external call to another external destination and have the two external parties connected through the trunks at the Central Office (CO).	No
Uniform Numbering Plan	The system supports the use of a network Uniform Numbering Plan that allows you to use the same digits to reach a station from any location in the network.	N/A
Voice Mail	The system has its own integral voice mail system.	Yes
Voice Mail Interfaces	Most voice processing systems work in conjunction with the system. The system provides the following voice processor interfaces: • Voice Mail - E&M Interface • Voice Mail - Digital E&M Interface • Voice Mail - Softkey support with Mitel’s NuPoint and Express Messenger™ • Voice Mail - ONS Interface.	N/A
Voice Mail Softkeys	Provides the user with a quick and convenient way to navigate voice mail. Access to the system is provided through context-sensitive softkeys on an IP telephone.	Yes
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Feature Name:	Description:	Support while the set is on the secondary controller
XNET	Proprietary switched MSDN/DPNSS networking over the PSTN. Also supported is a Hybrid XNET configuration. Hybrid signalling delivers voice over PRI channels, with MSDN call setup, feature invocation, and tear-down signalling over the IP network. Full XNET DPNSS feature transparency is maintained.	N/A
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Auto Attendant Features

Feature	Description
Open and Closed Greeting	A company greeting can be programmed to automatically change from open business hours to closed or after hours.
Expire at a preset Time Greeting	A Company Greeting can be programmed for use over holidays or shutdowns that automatically expires after a specified number of days.
Alternate Greetings	Each port can use one of eight alternate greeting sets (Open, Closed, or Temporary) to allow special greetings per port.
Play Greeting by Incoming Trunk Assignment	Each port can be assigned to answer calls on specific incoming trunks and play a greeting based on the destination dialed – for example, Sales, Shipping and Receiving, Customer Service.
Flexible Mailbox Numbering (Dial Plan)	In addition to supporting single-digit mailboxes (1 - 8), a mailbox dial plan of 2, 3, 4, or 5-digits can be selected.
Directory	Also known as Name Dialing. Callers may access a mailbox directory where they are able to reach a mailbox owner by dialing the person's first or last name rather than their mailbox number. The system can be configured for either first or last name dialing (but not both at the same time).
Caller Type-Ahead	Callers who are familiar with the system may enter their keypad selections without waiting for the system prompts.
Operator Revert	Callers may reach a live attendant at any time by dialing "0".
Fax Finder	Detects an incoming fax tone and directs it to the fax mailbox/extension.
Operator Transfer to a Mailbox	Allows an operator to transfer an outside caller to a specified mailbox where the caller immediately hears the subscriber's personal greeting and is prompted to leave a message. Callers press # to bypass or interrupt the greeting and begin recording a message.
Transfer to Any Extension	Allows the user to dial any internal extension defined in the system.
Quick Message Feature	Allows a caller reaching the auto-attendant to leave a message in a specific mailbox without transferring to the mailbox extension and possibly speaking live with the subscriber.
Multiple Message Capability	Allows an outside caller to leave more than one voice mail message per call, therefore saving on toll charges.
User Programmable Dial 0 Extension	Allows the user to program the dial 0 extension to any internal extension, for example, a personal or departmental secretary. The administrator can override the system default ("0" for the operator) with any valid phone number, including an external number or even a long distance number. The administrator can also override the system default on an extension by extension basis, with any valid phone number.
Park and Page	Auto Attendant Park and Page enables the Auto Attendant to park incoming calls and announce them to the requested party using paging. The requested party can then retrieve the call by using the "Call Park - Retrieve" feature.
Supervised/Unsupervised Transfer	The Auto Attendant can be programmed to perform either supervised or unsupervised transfers. The addition of supervised transfer capability allows calls that cannot be completed to return to the Auto Attendant for further processing.

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.
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.
	Page 1 of 4

Feature	Description
Forward Voice Mail to E-Mail	This feature 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.
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.
Outside Message Notification Calls	The administrator configures a trunk access code for use in all outside notification calls. The trunk access code controls the lines to be used for notification.
Distribution List, Broadcast Message	Allows four system-wide and five (per mailbox) personal distribution lists as well as a broadcast message facility to deliver a message to all mailboxes. Individual subscribers can belong to any number of distribution lists.
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.
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Feature	Description
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. **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.
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 an IP interface. This IP 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 creates or deletes a mailbox for the guest.
Record a Call	Using Voice Mail as a recorder, this feature allows a subscriber to record a live conversation between themselves and another party.
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.
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 VM Distribution Lists 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.
	Page 3 of 4

Feature	Description
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. Record a Call is supported through embedded voice mail functionality.
Voice Mail Hunt Group	MCD supports a single, large, voice mail hunt group with up to 240 members. This large hunt mail group can be resilient; however, you can only use it with NuPoint Messenger Release 10 or later voice mail systems.
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Features supported by protocols

The following tables summarize the features supported by QSIG and PRI protocols:

QSIG

The following table lists features supported by QSIG.

Feature	Description
QSIG Calling Name	Allows the system to send and receive the name of the caller; in turn, the called party will see the name of the caller on the telephone display screen if the appropriate Class of Service options are set.
QSIG Call Forwarding and Diversion	Incoming calls are diverted to another destination as defined by the user when the service is activated. This includes: • QSIG Call Forward Busy • QSIG Call Deflection • QSIG Call Forward No Reply • QSIG Call Forward Unconditional.
QSIG Message Waiting Indication	Users can set or cancel message waiting indications on the set of another party to indicate that they wish to be called back.
QSIG Call Transfer	A user can connect two other calls together, of the same basic service, as a new call (there must be three parties). This feature does not support placing a party on soft hold before making an enquiry call to another. This feature maps to the Transfer portion of the MSDN/DPNSS Call Hold and Three-Party Working Service.
QSIG Callback (Call Completion)	Users can request a Callback when they reach a busy or unanswered station. Callback with service retention is supported for the following Call Completion Supplementary Services: Completion of Calls to Busy Subscribers (SS-CCBS): users can set a Callback against a busy station. Completion of Calls on No Reply (SS-CCNR): users can set a Callback against a station that doesn't answer.
Call Offer	Users can offer calls to parties at a busy destination. The busy user receives indication of a call offer, while the calling party receives indication that a call offer has been invoked. The called user has the choice of clearing the current call and being re-rung, putting their current call on hold and accepting the offered call, or ignoring the offered call.
Path Replacement	Active calls, connected through the Q.SIG network, can be replaced with new connections which are more efficient or cost effective (when possible). The originating system requests the path replacement and the terminating system makes the optimized call. None of the existing call path is used when path replacement occurs. There must be an established call (a call that has been answered) before this feature is invoked.

The following table lists the supported QSIG ISO features that are supported by the Mitel 3300 Release 5.1 product. The X in the third column indicates that the feature is fully supported. For QSIG features not supported, the 3300 does not act as a transit switch.

Standard	Feature	Mitel 3300
ETS 300 012 (Ed 1)	Layer 1	X
ETS 300 402-1&2	Layer 2	X
ISO 11574, 11572	Audio Speech	X
ISO 11571	Numbering Plan	X
ISO 11582	Generic SS Platform (GF)	X
ISO 14136	Calling Line Identification Presentation (CLIP)	X
ISO 14136	Connected Line Identification Presentation (COLP)	X
ISO 14136	CLIP/COLP Restriction (CLIR)	X
ISO 13864, 13868	Calling Name Identification Presentation (CNIP)	X
ISO 13864, 13868	Connected Name Identification Presentation (CONP)	X
ISO 13864, 13868	CNIP/CONP Restriction (CNIR)	X
ISO 13872, 13873	Call Forwarding Unconditional (CFU)	X (note 1)
ISO 13872, 13873	Call Forwarding Busy (CFB)	X (note 1)
ISO 13872, 13873	Call Forwarding No Reply (CFNR)	X (note 1)
ISO 13865, 13869	Call Transfer (CT)	X (By join)
ISO 13863, 13874	Path Replacement (PR)	X (note 2)
ISO 13866, 13870	Call Completion to Busy Subscriber (CCBS)	X
ISO 13866, 13870	Call Completion on No Reply (CCNR)	X
ISO 14841, 14843	Call Offer (CO)	X (note 3)
ISO 15505, 15506	Message Waiting (MWI)	X (note 4)
ISO 15055, 15056	Transit Count (TC)	X
ISO 13866 13870	Call Completion Busy Subscriber (CCBS)	X (note 5)
ISO 13866 13870	Call Completion No Answer (CCNA)	X (note 5)

Notes:

1. Does not support Interrogation. It is a way to determine the call forwarding status of a remote phone.
2. Only supports Originator Requesting Path Replace. Either end may ask for the route optimization but Mitel only supports this for the originator. It is recommended that the route optimization timer on the Mitel switch be set to a shorter time than the other side so that the Mitel switch initiates the optimization request.
3. Only supported without path retention. Path retention retains the connection between two PBXs so that a supplementary service can be invoked without establishing a new connection. This method holds up a trunk resource and is not supported.
4. Does not support MWI interrogate function. It is a way to determine the message waiting lamp status of a remote phone.
5. Does not support connection retention. Connection retention holds up a virtual call between the two end-points. Mitel supports path reservation which ensures that resources are available when User B can accept User A's call and service retention in that the call is compelled to complete.

PRI

The following table lists features supported by PRI.

Feature	Description
ANI/DNIS/ISDN Number Delivery	Automatic Number Identification and Dialed Number Identification Service identify numbers that are transmitted on an incoming trunk.
Call-by-call Service	With Call-by-Call Service, access channels do not have to be dedicated to specific services such as OUTWATS or 800 services. This enables the customer to reduce facilities and integrate dedicated and switched, inbound and outbound, voice and data traffic on a single facility. It also allows a business with calling peaks to dynamically allocate coverage across channels so that access lines are optimized. This implementation ensures that incoming calls are not turned away because all incoming channels are busy while adjacent outgoing channels are idle.
Calling Line Identification	The telephone number of the calling party is transmitted to the Mitel PBX and can be sent to devices within the system.
E-911 Support	Displays indicate the extension and the location of the person who dialled 911. Notifications of 911 calls are audible, continuous and distinct from regular ringing patterns when the set is idle and on hook. If the user is already on a call, a new call tone alerts the user to the alarm condition. The alarm overrides sets having DND enabled.

MSDN/DPNSS

The following table lists features supported by MSDN/DPNSS.

Feature	Description
Callback	Allows you to request that the system notify you when a busy line becomes idle or when an unanswered station goes off-hook and on-hook.
Call Forward	Allows you to redirect incoming calls to an alternate number.
Calling Line Identification	The telephone number of the calling party is transmitted to the Mitel PBX and can be sent to devices within the system.
Camp-on (Call Waiting)	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.
Call Split	See Conference Split.
Conference	Allows you to connect three or more calls into a single telephone conversation. While you are in a Conference, you can use any of the features that would normally be available during a two-party call.
Conference Split	Allows you to separate a 3-party conference so that two of the parties can speak privately, while the other is placed on Consultation Hold.
Do Not Disturb	Allows you to place your set in an apparent busy condition without affecting the outgoing functionality. If someone calls your set while DND is activated, they hear a special busy tone.
SMDR -External	Collects data for outgoing and incoming trunk calls.
SMDR -Internal	Collects data for calls made between stations within the system.
Recall	Allows an incoming caller, who has been transferred to an idle station and not answered within a specified time-out period, call back the last party who handled the call. Similar time-out recalls occur for parties who are transferred to busy stations or placed on hold.
Tandem Trunking	The system can transparently interconnect trunk circuits originating from one CO or PBX and terminating on another (tandem trunking), without attendant intervention.
Trunk Select - Direct	Allows you to access an outside trunk for the purposes of originating and receiving external calls. Because the trunk is assigned to a line appearance, you can access the trunk to make or answer calls without trunk access codes.
Override	Allows you to enter a conversation at a busy station or ring a station with Do Not Disturb activated. Before you enter the conversation, all parties receive a warning tone.
Serial Call	Allows a centralized attendant to set up serial calls for users on remote PBXs.
Route Optimization	Replaces non-optimal call routing with routings that use the fewest number of network channels.
Hold on Hold	Allows a person on a two party call to temporarily suspend the telephone call. While the call is on hold, the person that placed the call is able to use other telephone features. The call can be retrieved from the telephone that placed the call or from another telephone.
Direct Page	Allows you to page another telephone over its built-in speaker. See Off-Hook Voice Announce.
Networked Group Page	Group Paging can be completed across a network or network cluster, allowing, for example, a set on system A to page a specific group on system B.

Security Features

Encrypted Media Path and Signaling Path

Encrypted media and signaling path is supported for all Mitel's IP phones. The media path encryption is accomplished with Secure RTP using 128-bit Advanced Encryption Standard (AES). Encryption is backwards compatible to support both currently shipping desktops as well as previously deployed Mitel IP desktops. Mitel provides encryption of the signaling path between multiple MCD systems using Secure Sockets Layer (SSL) protocol. This allows scalability of applications by configuring MCD systems into clusters or deploying them as part of a centrally managed but distributed architecture.

Phone and User Authentication

Mitel implements phone authentication that requires a unique association of MAC addresses and IP and user-entered PIN registration numbers. Additionally, desktop software downloads are encrypted. Mitel also provides 802.1X authentication for desktops (Release 6.0 and later) which offers support for the Extensible Authentication Protocol (EAP) using EAP-MD5 challenge authentication to a RADIUS Server.

Worm and Virus Protection

MCD uses an embedded real time operating system. This system is less susceptible to virus or worm attacks that target traditional applications and their OS services because it provides a very small base of common functionality with general purpose operating systems. This lack of common functionality means that VxWorks is not affected by the viruses and worms typically found on networks and the Internet. This also makes it difficult for an attacker to write a virus targeted at generic VxWorks implementations.

Application servers based on Windows NT/2000 must be properly maintained with regard to current operating system security updates. Mitel products based on Windows NT/2000 include the Contact Center Solutions, Speech Server and Messaging Server systems and Enterprise Manager. These key application servers must be maintained with the latest in Microsoft security updates and worm protection.

Prevention of Toll Abuse

Any communication system that has a combination of Direct Inward System Access (DISA) integrated auto attendant or RAD groups and peripheral interfaced auto attendant or voice mail can be susceptible to toll abuse. Therefore it is important to assign appropriate telephone privileges and restrictions to devices. In addition, public telephones should be denied toll access unless authorized through an attendant.

MCD has comprehensive toll control built in. It lets you restrict user access to trunk routes and/or specific external directory numbers. It also provides Class of Restriction (COR) and Class of Service (COS) features that can substantially reduce the risk of toll abuse.

As a deterrent to toll abuse by internal callers, Station Message Detail Recording (SMDR) can be used to track calls from within your company, providing detailed information such as the originating extension number, time, duration, and number dialed. SMDR record access should be restricted as with any other function.

Secure Management Interfaces

MCD includes a fully integrated set of management tools designed to install, manage, and administer MCD systems. Three levels of access are provided in order to meet the needs of system technicians, group administrators, and the desktop telephony users themselves. All of these integral management tools use Secure Socket Layer (SSL) security for data encryption.

User access to the management tools is controlled by a login and password. Once a user logs into an MCD, the system displays a menu of the specific tools to which they have been granted access.

Mitel also offers the Management Access Point to provide secure remote administration for VPN or dial-up access.

Secure Applications

Mitel addresses application security via:

- Your Assistant - Provides a softphone with encrypted call path and call signaling as well as secure instant messaging to keep IM traffic encrypted and inside the network.
- Wireless Solutions - Includes secure IP-DECT solution (EMEA) and encryption for 802.11b wireless telephony, support for encryption using Wi-Fi Protected Access (WPA) and authentication using WPA and WPA2.
- XML Implementation - Supports encryption of all traffic using standard SSL and provides strong certificate-based authentication for API use.

SIP Security

Mitel SIP desktops support secure RTP and also satisfy the PROTOS test suite for CERT advisory CA-2003-06. The SIP desktops also provide support for firewall traversal and SSL-encrypted SIP.

Product Availability by Region

North America

This table indicates the availability of products in Canada and the United States.

North American Region		
	Canada	**United States**
Mitel Communications Director	Y	Y
Mitel Communications Suite	Y	Y
3300 ICP Components	Y	Y
3300 Voice Mail	Y	Y
3300 Wireless	Y	Y
Peripheral Cabinet	Y	Y
SX-200 Bays	Y	Y
Applications		
OPS Manager	Y	Y
Contact Center Management	Y	Y
Interactive Contact Center	Y	Y
Contact Center Scheduling	Y	Y
Multimedia Contact Center	Y	Y
Intelligent Queue	Y	Y
Call Recording	Y	Y
Speech Server/Messaging Server	Y	Y
Unified Communicator	Y	Y
5300 Intelligent Directory Application*	Y	Y
*** Note:** This application is supported but the interface is in English only.		
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North American Region (continued)		
	Canada	**United States**
Phones		
5302 IP Phone	Y	Y
5304 IP Phone	Y	Y
5312 IP Phone	Y	Y
5320 IP Phone	Y	Y
5324 IP Phone	Y	Y
5330 IP Phone	Y	Y
5340 IP Phone	Y	Y
5360 IP Phone	Y	Y
OpenPhone 27 (DECT)	Y	Y
5602, 5606 and 5606 (Alarm) IP DECT Phones	Y	Y
SUPERSET 4025	Y	Y
Consoles		
5540 IP Console	Y	Y
5550 IP Console	Y	Y
Accessories		
5310 IP Conference Unit	Y	Y
IP PKM (12 and 48 Button units)	Y	Y
IP Paging Unit	Y	Y
Line Interface Module	Y	Y
Wireless LAN (WLAN) Stand	Y	Y
Gigabit Ethernet Stand	Y	Y
Cordless Module and Accessories	Y	Y
5610 DECT Handset and IP DECT Stand	Y	Y
		Page 2 of 2

Asia Pacific

This table indicates the availability of products in Australia, New Zealand and China.

Asia Pacific Region			
	Australia	**New Zealand**	**China**
Mitel Communications Director	Y	Y	Y
Mitel Communications Suite	Y	Y	Y
3300 ICP Components	Y	Y	Y
3300 Voice Mail	Y	Y	Y
3300 Wireless	Y	Y	N
Peripheral Cabinet	Y	Y	Y
SX-200 Bays	N	N	N
Applications			
OPS Manager	Y	Y	Y*
Contact Center Management	Y	Y	N
Interactive Contact Center	Y	Y	N
Contact Center Scheduling	Y	Y	N
Multimedia Contact Center	Y	Y	N
Intelligent Queue	Y	Y	N
Call Recording	Y	Y	N
Speech Server/Messaging Server	Y	Y	N
Unified Communicator	Y	Y	Y**
5300 Intelligent Directory Application*	Y	Y	Y
*** Note:** This application is supported but the interface is in English only. **** Note:** YA Collaboration Option is not translated to Chinese.			
			Page 1 of 2

Asia Pacific Region (continued)			
	Australia	**New Zealand**	**China**
Phones			
5302 IP Phone	Y	Y	Y
5304 IP Phone	Y	Y	Y
5312 IP Phone	Y	Y	Y
5320 IP Phone	Y	Y	Y*
5324 IP Phone	Y	Y	Y
5330 IP Phone	Y	Y	Y
5340 IP Phone	Y	Y	Y
5360 IP Phone	Y	Y	Y*
OpenPhone 27 (DECT)	Y	Y	N
5602, 5606 and 5606 (Alarm) IP DECT Phones	Y	Y	Y
SUPERSET 4025	Y	Y	N
Consoles			
5540 IP Console	Y	Y	Y
5550 IP Console	Y	Y	Y
Accessories			
5310 IP Conference Unit	Y	Y	Y
IP PKM (12 and 48 Button units)	Y	Y	Y
IP Paging Unit	Y	Y	Y
Line Interface Module	Y	Y	N
Wireless LAN (WLAN) Stand	Y	Y	Y
Gigabit Ethernet Stand	Y	Y	Y
Cordless Module and Accessories	Y	Y	N
5610 DECT Handset and IP DECT Stand	Y*	Y*	N
*** Note:** Approvals pending.			
			Page 2 of 2

EMEA Region

This table indicates the availability of products in the different countries of the EMEA region.

EMEA Region									
	UK	**Spain**	**Portugal**	**Netherlands**	**Italy**	**Germany**	**France**	**UAE**	**South Africa**
Mitel Communications Director	Y	Y	Y	Y	Y	Y	Y	Y	Y
Mitel Communications Suite	Y	Y	Y	Y	Y	Y	Y	Y	Y
3300 ICP Components	Y	Y	Y	Y	Y	Y	Y	Y	Y
3300 Voice Mail	Y	Y	Y	Y	Y	Y	Y	Y	Y
3300 Wireless	Y	Y	Y	Y	Y	Y	Y	Y	Y
Peripheral Cabinet	Y	N	N	Y	Y	Y	N	Y	Y
SX-200 Bay	N	N	N	N	N	N	N	N	N
Applications									
OPS Manager*	Y	Y	Y	Y	Y	Y	Y	Y	Y
Contact Center Management	Y	N	N	Y	Y	N	N	N	N
Interactive Contact Center	Y	N	N	N	N	N	N	N	N
Contact Center Scheduling	Y	N	N	N	N	N	N	N	N
Multimedia Contact Center	Y	N	N	N	N	N	N	N	N
Intelligent Queue	Y	N	N	N	N	N	N	N	N
Call Recording	N	N	N	N	N	N	N	N	N
Speech Server/ Messaging Server	Y	N	N	N	N	N	N	N	Y
Unified Communicator	Y	Y	Y	Y	Y	Y	Y	Y	Y
9100 Call Center Commander	Y	N	N	N	N	N	N	N	N
5300 Intelligent Directory Application*	Y	Y	Y	Y	Y	Y	Y	Y	Y
* **Note:** This application is supported but the interface is in English only.									
									Page 1 of 2

EMEA Region (continued)									
	UK	**Spain**	**Portugal**	**Netherlands**	**Italy**	**Germany**	**France**	**UAE**	**South Africa**
Phones									
5302 IP Phone	Y	Y	Y	Y	Y	Y	Y	Y	Y
5304 IP Phone	Y	Y	Y	Y	Y	Y	Y	Y	Y
5312 IP Phone	Y	Y	Y	Y	Y	Y	Y	Y	Y
5320 IP Phone	Y	Y	Y	Y	Y	Y	Y	Y	Y
5324 IP Phone	Y	Y	Y	Y	Y	Y	Y	Y	Y
5330 IP Phone	Y	Y	Y	Y	Y	Y	Y	Y	Y
5340 IP Phone	Y	Y	Y	Y	Y	Y	Y	Y	Y
5360 IP Phone	Y	Y	Y	Y	Y	Y	Y	Y	Y
OpenPhone 27	Y	Y	Y	Y	Y	Y	Y	Y	Y
5602, 5606 and 5606 (Alarm) IP DECT Phones	Y	Y	Y	Y	Y	Y	Y	Y	Y
SUPERSET 4025	Y	N	N	Y	Y	Y	Y	Y	Y
Consoles									
5540 IP Console	Y	Y	Y	Y	Y	Y	Y	Y	Y
5550 IP Console	Y	Y	Y	Y	Y	Y	Y	Y	Y
Accessories									
5310 IP Conference Unit	Y	Y	Y	Y	Y	Y	Y	Y	Y
IP PKM (12 and 48 Button units)	Y	Y	Y	Y	Y	Y	Y	Y	Y
IP Paging Unit	Y	Y	Y	Y	Y	Y	Y	Y	Y
Line Interface Module	Y	Y	Y	Y	Y	Y	Y	N	N
Wireless LAN (WLAN) Stand	Y	Y	Y	Y	Y	Y	Y	Y	Y
Gigabit Ethernet Stand	Y	Y	Y	Y	Y	Y	Y	Y	Y
Cordless Module and Accessories	Y	Y	Y	Y	Y	Y	Y	N	N
5610 DECT Handset and IP DECT Stand	Y	Y	Y	Y	Y	Y	Y	Y	Y
Page 2 of 2									

Latin America

This table is a list of products available in Latin America. These products may not have completed regulatory approvals.

Latin America				
	Argentina	**Brazil**	**Chile**	**Mexico**
Mitel Communications Director	Y	Y	Y	Y
Mitel Communications Suite	Y	Y	Y	Y
3300 ICP Components	Y	Y	Y	Y
3300 Voice Mail	Y	Y	Y	Y
3300 Wireless	Y	Y	Y	Y
Peripheral Cabinet	Y	Y	Y	Y
SX-200 Bay	Y	N	N	N
Applications				
OPS Manager*	Y	Y	Y	Y
Contact Center Management	Y	Y	Y	Y
Interactive Contact Center	Y	Y	Y	Y
Contact Center Scheduling	Y	Y	Y	Y
Multimedia Contact Center	Y	Y	Y	Y
Intelligent Queue	Y	Y	Y	Y
Call Recording	Y	Y	Y	Y
Mitel Speech Server (attendant only)	Y	Y	Y	Y
Unified Communicator	Y	Y	Y	Y
5300 Intelligent Directory Application*	Y	Y	Y	Y
*** Note:** This application is supported but the interface is in English only.				
				Page 1 of 2

Latin America (continued)				
	Argentina	**Brazil**	**Chile**	**Mexico**
Phones				
5302 IP Phone	Y	Y	Y	Y
5304 IP Phone	Y	Y	Y	Y
5312 IP Phone	Y	Y	Y	Y
5320 IP Phone	Y	N	Y	Y
5324 IP Phone	Y	Y	Y	Y
5330 IP Phone	Y	Y	Y	Y
5340 IP Phone	Y	Y	Y	Y
5360 IP Phone	Y	N	Y	Y
OpenPhone 27	N	N	N	N
5602, 5606 and 5606 (Alarm) IP DECT Phones	Y	Y	Y	Y
SUPERSET 4025	Y	Y	Y	Y
Consoles				
5540 IP Console	Y	Y	Y	Y
5550 IP Console	Y	Y	Y	Y
Accessories				
5310 IP Conference Unit	Y	Y	Y	Y
IP PKM (12 and 48 Button units)	Y	Y	Y	Y
IP Paging Unit	Y	Y	Y	Y
Line Interface Module	Y	Y	Y	Y
Wireless LAN (WLAN) Stand	Y	Y	Y	Y
Gigabit Ethernet Stand	Y	Y	Y	Y
Cordless Module and Accessories	Y	Y	Y	Y
5610 DECT Handset and IP DECT Stand	N	N	N	Y
				Page 2 of 2

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