MITEL

COMMUNICATIONS DIRECTOR

GENERAL INFORMATION GUIDE
MITEL COMMUNICATIONS DIRECTOR RELEASE 6.0 SP1

Mitel Communications Director Release 6.0 SP1
General Information Guide
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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), and Industry Standard Servers (ISSs). The topics covered in this guide include

- A description of the system architecture and components
- Migration strategies
- Supported applications

Audience

This guide is intended for

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

Related Documentation

You can access documentation on the Mitel Customer Documentation web site at http://edocs.mitel.com. 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 (phone) user guides.

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

- ***General Information Guide:*** an overview of the system
- ***Site Planning Guide:*** site planning and site preparation guidelines
- ***Technician's Handbook*****:** installation, upgrade, and maintenance instructions
- ***Hardware Technical Reference Manual:*** hardware specifications
- ***System Administration Tool Online Help:*** programming, maintenance, and troubleshooting procedures
- ***Troubleshooting Guide:*** information on diagnosing and resolving common problems with MCD

- ***Resiliency Guidelines:*** a comprehensive overview of the Mitel Resiliency solution and offer customers the tools to understand, plan, and implement a resilient network
- ***Engineering Guidelines*****:** information required to engineer a MCD system for a customer site. The guidelines are intended to highlight specific areas of the product that need to be considered before installation.

Overview

Mitel Communications Director (MCD) provides businesses of all sizes with a scalable, feature-rich communications system using a single stream of software. MCD is designed to meet the needs of businesses that have from 5 to 65,000 users, whether they are single-site deployments or multi-site networks that span many countries.

Platforms

MCD is a modular, scalable system that runs on the following hardware platforms:

- Mitel 3300 ICP controllers, including AX, CX II and CXi II
- Industry Standard Servers (ISS)

Modular Platform Design Provides Scalability and Flexibility

Deploying MCD on the above-mentioned platforms enables customers to meet current requirements and invest in a system that can grow with them 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 MCD

You can deploy MCD to support a broad spectrum of site configurations. For example, you could

- Implement a highly centralized solution at the head office with the call control and IP telephony services delivered over Wide Area Network (WAN) connections to small branch offices.
- Configure larger branch offices with main controllers on site to provide local support.
- Cluster an entire network of controllers to function as one large system.

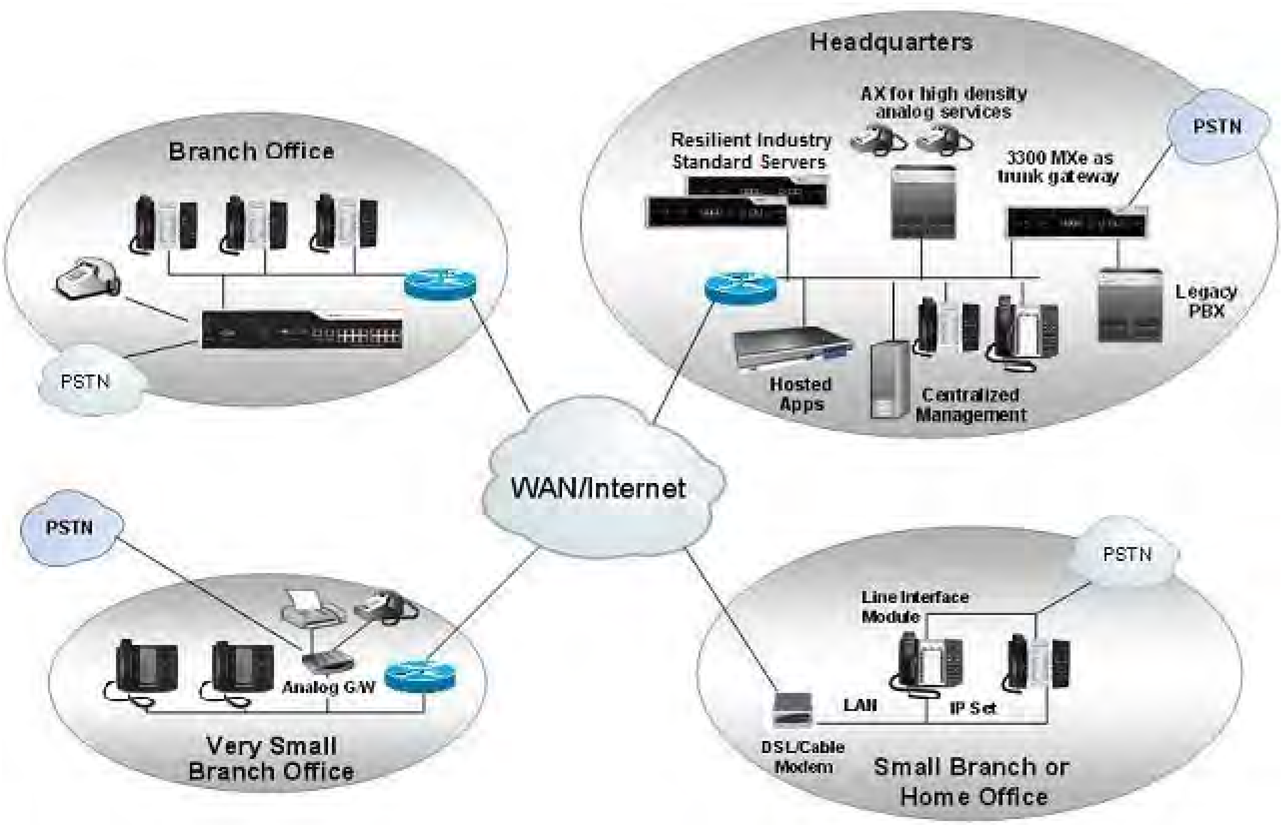


Figure 1: MCD Site Configuration

For smaller organizations, Mitel delivers a 3300 ICP system which incorporates a powered Ethernet switch and has a number of imbedded applications that can be complimented with Mitel Applications Suite (MAS).

For large organizations or multi-site deployments, you can deploy up to 999 controllers in a cluster to deliver extensive features, services, and applications. These controllers use a peer-to-peer communication protocol to share management and administration data between systems to ensure consistency of features and applications 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 the following setup:

- 5000 users on an industry standard server with a backup MXe III Controller to provide resilient support
- An MXe III Controller acting as a trunking gateway with connections to the traditional telephone network for outside access
- 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. Customers 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

Mitel supports VMware® to enable voice and business applications to run together in fault tolerant, highly available environments. Mitel's unified communications features run as virtual appliances on the VMware vSphere™ 4 virtualization platform.

Mission-critical environments can use industry standard servers that have the processing capacity to power redundancy and can provide 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 there is no single point-of-failure.

Virtual Mitel Communications Director offers the same MCD functions and capabilities, while being treated like any other virtualized application in the data center. With Virtual MCD, you can access the full range of standard MCD telephony features such as Dynamic Extension, clustering, resiliency support, SIP service provider interconnect, and multi-node management.

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.

Applications that Enhance Productivity

MCD includes an extensive number of applications that provide significant value to an organization and its employees. These applications enhance communication, productivity, accessibility, mobility, and 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). The program helps businesses to develop custom applications or features to achieve higher productivity.

Devices that Support Users

Mitel offers 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 powerful system features through programmable feature keys, softkeys, and menu-guided applications such as Call Forwarding and Call History. Mitel provides the following devices:

- Display Phones
- Desktop Application Phones
- Wireless Phones (DECT and WiFi)
- Session Initiation Protocol (SIP) Phones
- Consoles
- Conference Units
- Video Conferencing Devices
- Digital Phones
- Phone Accessories
- Softphones
- PC Clients

Tools That Minimize Configuration and Support

MCD includes tools that simplify Installation, configuration, administration, and the work of end-users, group administrators, system administrators, and installers. The tools and their functions are as follows:

- 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 use the default features and system settings to minimize configuration requirements, or can configure these settings for maximum flexibility. Administrators can enable or disable system settings across the entire system or network.

Migration Made Easy

Because of Mitel's long history in voice communications, Mitel continues 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 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.

MCD Software Overview

Mitel delivers sophisticated call management applications and desktop solutions on the 3300 ICP platform. Scalable, resilient, call control functionality is powered by IP and fully supports 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 phone devices:

- For IP telephony, it provides call setup, tear down, and signaling between Ethernet IP connected phones.
- 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 required only between IP and non-IP networks optimizing the use of system resources.
- Migration from traditional PBX to IP telephony is seamless and efficient.

MCD provides call control features and applications that enhance business communications.

Licensing

The MCD licence strategy delivers simplicity and flexibility while maintaining cost effectiveness.

Every MCD system, whether it is deployed on a Mitel 3300 controller, Industry Standard Server or a virtual appliance, requires a Core package that sets the MCD System Type and allocates specific licences for immediate customer use. Further licences can be purchased and allocated required. The core software package also defines any license limitations or restrictions.

System Type

MCD systems are activated either as Standalone or Enterprise Systems. The underlying software running the two system types is the same; however, the different system types allow Mitel to present the MCD solution to different markets and customer segments while using the same software stream.

As of MCD Release 5.0, virtual MCD systems can be activated as Standalone or Enterprise Systems.

The Enterprise System provides the inherent networking capabilities of MCD along with full User and Device resiliency.

Individual User Licences

Individual User Licences include

- **User Licence:** enables an IP device to be fully activated for all features or can be used for a Hotdesk User as the user logs on to a phone that does not have a licence
- **External Hotdesking licence:** enables an off PBX number to be added to the system and is typically used to add a mobile phone for twinning
- **ACD Active Agent licence:** permits concurrent usage license—one required for each concurrent agent log in
- **HTML licence:** required for certain HMTL applications
- **Analog line licence:** required for each ONS port enabled on the ASU II
- **Voice mail licences:** one licence is required per mailbox, also used for auto attendant applications
- **Multi-device User Licence:** intended for users who have a range of devices: desk phones, soft phones, mobile phones, in-building wireless phones
- **Suite Licence:** used in Hospitality solutions where up to six phones can be added to a single suite using only a single Suite licence

Individual user licences may vary depending on whether the customer's system type is Standalone or Enterprise: Enterprise Systems require Enterprise User licences and Enterprise ACD licences and Standalone Systems require Standard User licences and Standard ACD licences.

Trunking and Compression Licences

Individual Trunking and Compression Licences include

- SIP Trunk licence
- Digital Link Licence
- G729 Compression licences
- T38 Licences

With the introduction of the Standalone and Enterprise System types, the system wide licence options available are as follows:

- Integrated Directory Services Integration
- MLPP
- Voice mail - Hospitality
- Enterprise License Sharing

Enterprise License Sharing allows Enterprise customers to group all MCD systems together and move licences around their solution.

Compression

Bandwidth optimization is a key requirement of VoIP systems: MCD supports G.722.1 and compresses calls using G.729a. Compression reduces the bandwidth of a call from 64 kbps to 8 kbps plus packet overhead. By using voice compression across the LAN/WAN infrastructure, you can optimize bandwidth usage for voice calls. The mechanism for managing this feature is based on zones. You can place groups of MCS devices in zones to compress calls between zones (not within zones). You can define zones 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 voice compression: calls between IP Phones on the LAN/WAN infrastructure can be compressed 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.

You can purchase optional compression licenses and DSP modules to enable TDM-to-IP compression on the 3300 ICP.

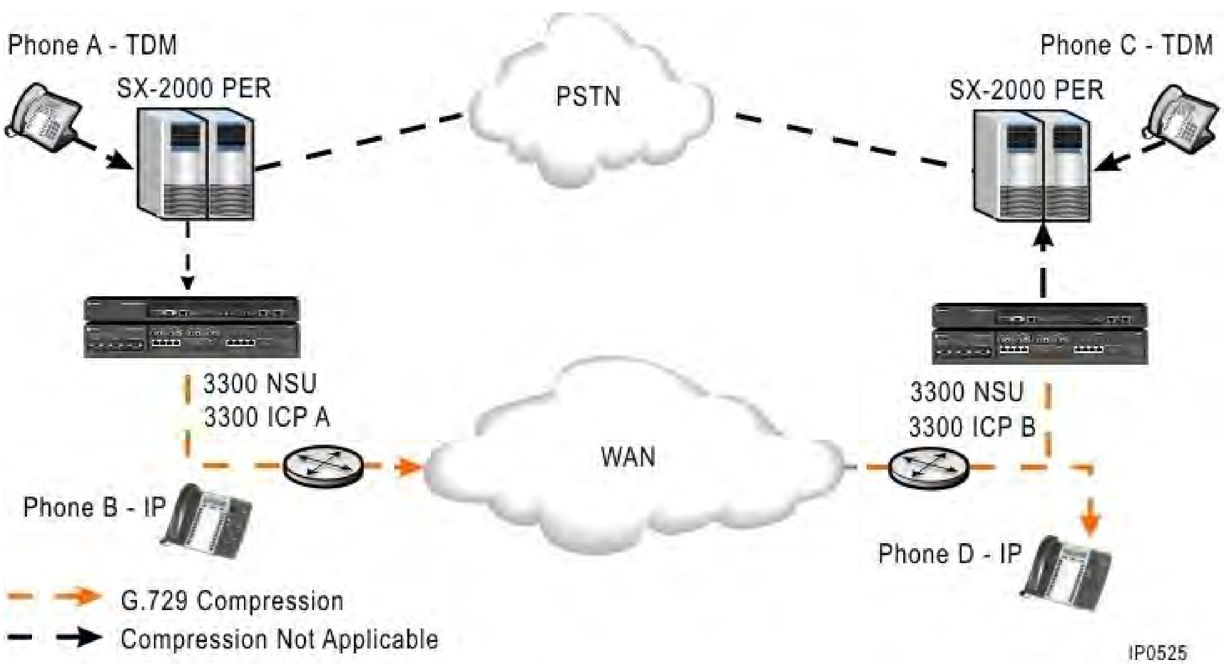


Figure 2: Voice Compression Between 3300 ICPs

IP Networking

IP Networking enables you to network systems together. Instead of leasing dedicated voice circuits, you can route voice traffic over the existing LAN/WAN infrastructure. The Mitel IP Networking implementation uses point-to-point topology to optimize network resources.

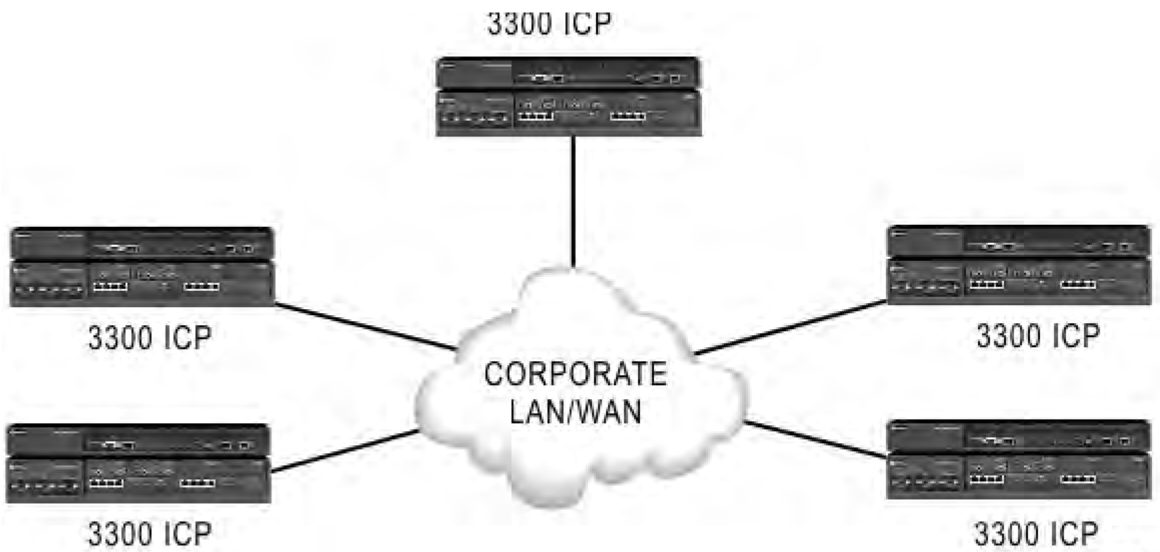


Figure 3: IP Networking - Point to Point Topology

IP Networking supports the MSDN/DPNSS protocols over the IP infrastructure. You can cluster controllers in a single location to provide greater resiliency than that of a single controller operating autonomously. You can seamlessly network geographically separated controllers 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.

You can configure Enterprise Networking Solutions to act as an IP Networking gateway for SX-2000 and SX-200® PBXs, or third-party PBXs.

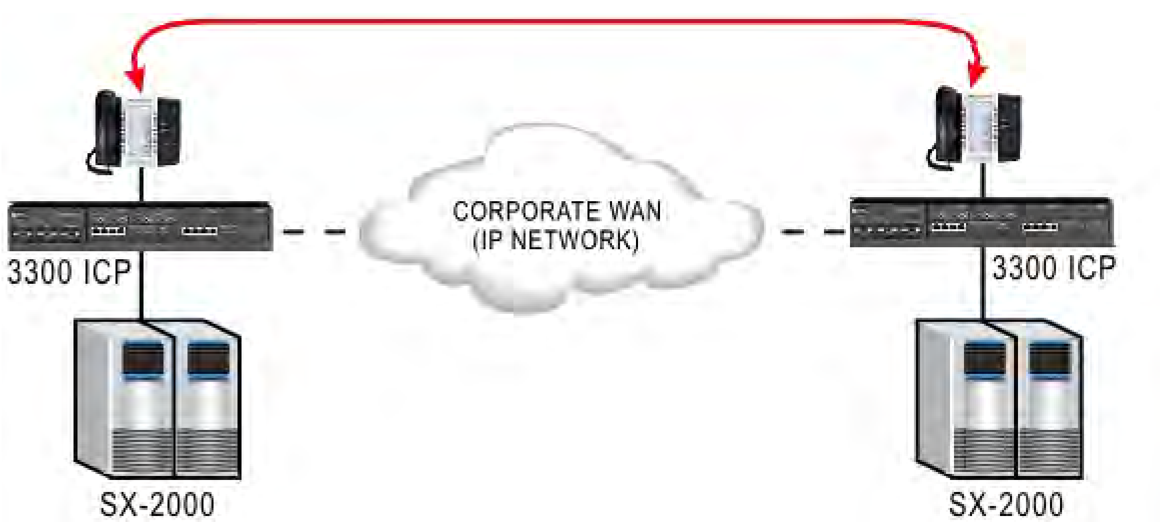


Figure 4: 3300 ICP as Trunking Gateway for PBXs

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.

SIP Trunking

To manage costs within their organizations, many companies opt to replace their traditional PSTN connections with new SIP services deployed by service providers.

MCD connects to service provider networks using the SIP protocol over the IP network. The SIP Trunking solution provides many features, such as basic calling features, billing capability, Emergency Services support, and FAX support.

Mitel operates a SIP center of excellence. The SIP team undertakes interop activity. They use the SIP protocol to certify integration with SIP service providers for applications that connect to MCD using SIP and third party SIP devices. Mitel OnLine publishes interop compliance policies on a monthly basis.

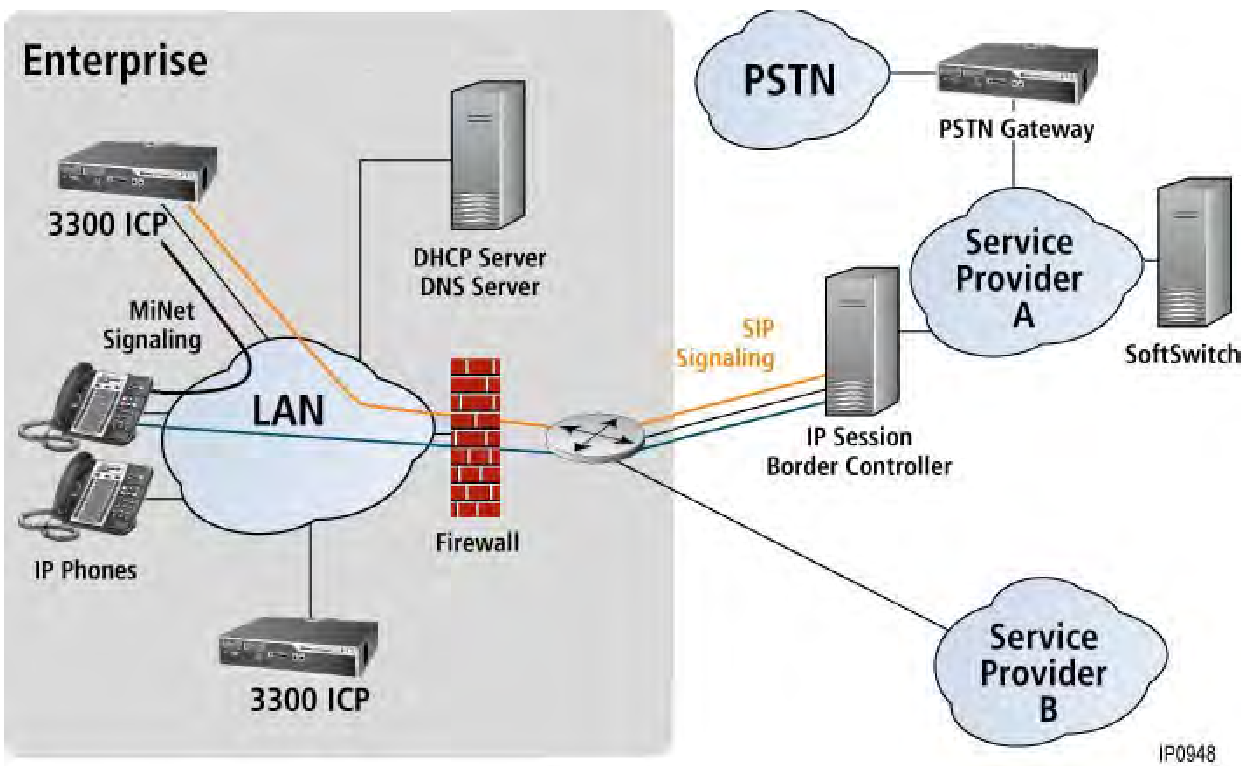


Figure 5: SIP Trunking

Configurable Real-time Transport Protocol (RTP) Packetization

MCD administrators can configure a voice stream packet rate for SIP trunks to their service providers between 10ms to 80ms (with 10ms increments).

Malicious Call Trace

For incoming SIP calls that are tagged as Malicious Calls, MCD records the Media IP address and port used remotely and captures SIP signalling information. This information cannot be sent to the SIP Service Provider but is recorded if required.

FAX Support

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

Real-time, Group 3 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.

Bandwidth Management

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 readily be deployed across the WAN (or Internet using the Mitel Border Gateway teleworker service) hosted by a centralized MCD controller with gateways for remote survivability. When you deploy remote sites, you must ensure non-voice data has adequate bandwidth and voice quality is preserved.

If the bandwidth between locations is restricted, you can reduce consumption by applying compression to voice traffic voice between IP Phones. Compression reduces the bandwidth demands of a standard voice call (G.711) by compressing the call using the G.729/G.722.1 codec. Compression is applied to calls between the zones of IP Phones (see Bandwidth Management on page 14 for details).

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

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

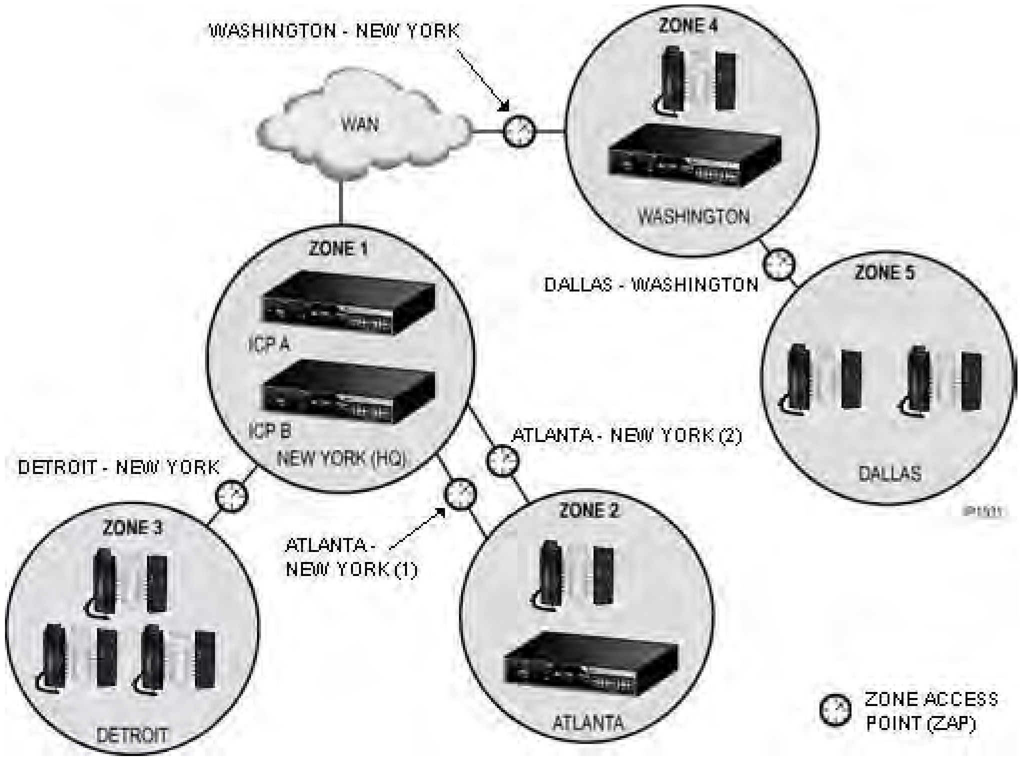


Figure 6: Bandwidth Management at Zone Access Points

Resiliency

Resiliency on MCD increases the reliability of communications by maintaining calls in progress, handling new incoming and outgoing calls, and continuing to provide voice mail services in the event of MCD or network failures. 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 failures.

By taking advantage of IP-network characteristics of location independence, resiliency provides a flexible solution to enhance system reliability. By using resources that are spread across the network, resiliency ensures there is no single point of failure and optimizes hardware use. Resiliency provides an advantage over many other competing alternatives where solutions involve costly hardware redundancy for each controller.

If the primary controller experiences a service outage, support for 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 following figure illustrates how you can configure a site with fully resilient devices. Node B is the secondary controller for the phones on Node A, and Node A is the secondary controller for the phones on Node B. If a controller experiences an outage, phone support is transferred to its secondary controller.

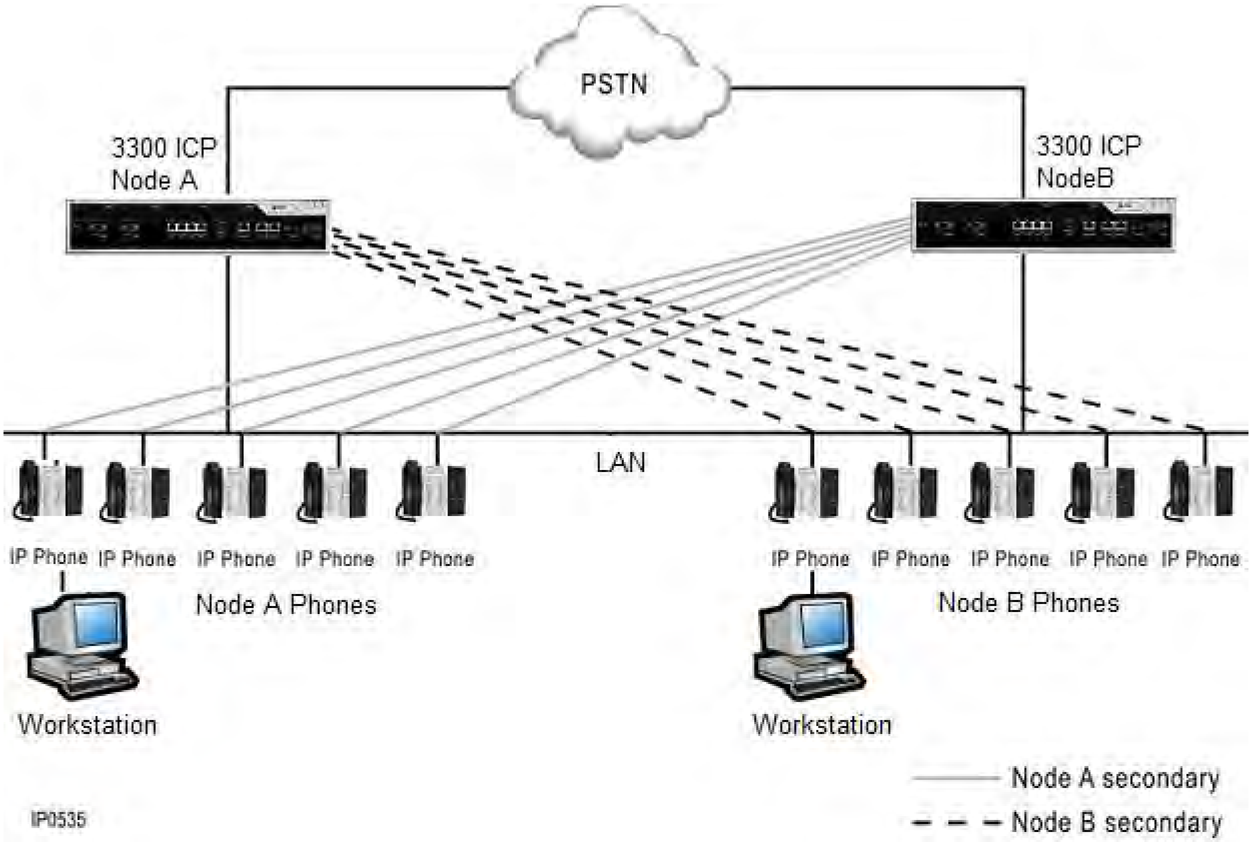


Figure 7: Resilient Configuration

Advantages Over Redundancy

Resilient solutions are less costly and more flexible than redundant solutions. While the redundancy model is highly effective and reliable, it is unnecessarily costly for some customers.

Distributed resilient networks enable you to route around failed or otherwise inaccessible portions of an IP network. Distributed resilient networks provide 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. The secondary controller can also function as one of the following devices:

- 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 III Controller, and AX Controller.

Hot Desking

Hot Desking creates a more flexible work environment by enabling users to share IP phones. This is ideal for businesses that employ telecommuters, sales agents, and other employees who spend much of their time out of the office.

Hot Desking enables a pool of shared phones to be made available to employees instead of assigning a dedicated phone to each employee. You can configure IP phones as hotdesk phones without requiring System User licences.

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

- 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: users can log on to any Hot Desk-enabled set in the cluster. After a user logs on, the set is redirected to the user's host ICP. Figure 8 provides an example of a hot desk user in a cluster.

External Hot Desking

Hot Desking is supported on external answering points such as cellular phones, home phones, and remote phones using a VoIP service. Mitel can also treat extensions on other manufacturers PBX's as external hotdesk devices. After a user's number is programmed to support External Hot Desking, calls to the user are routed to the user's External Hot Desk phone number.

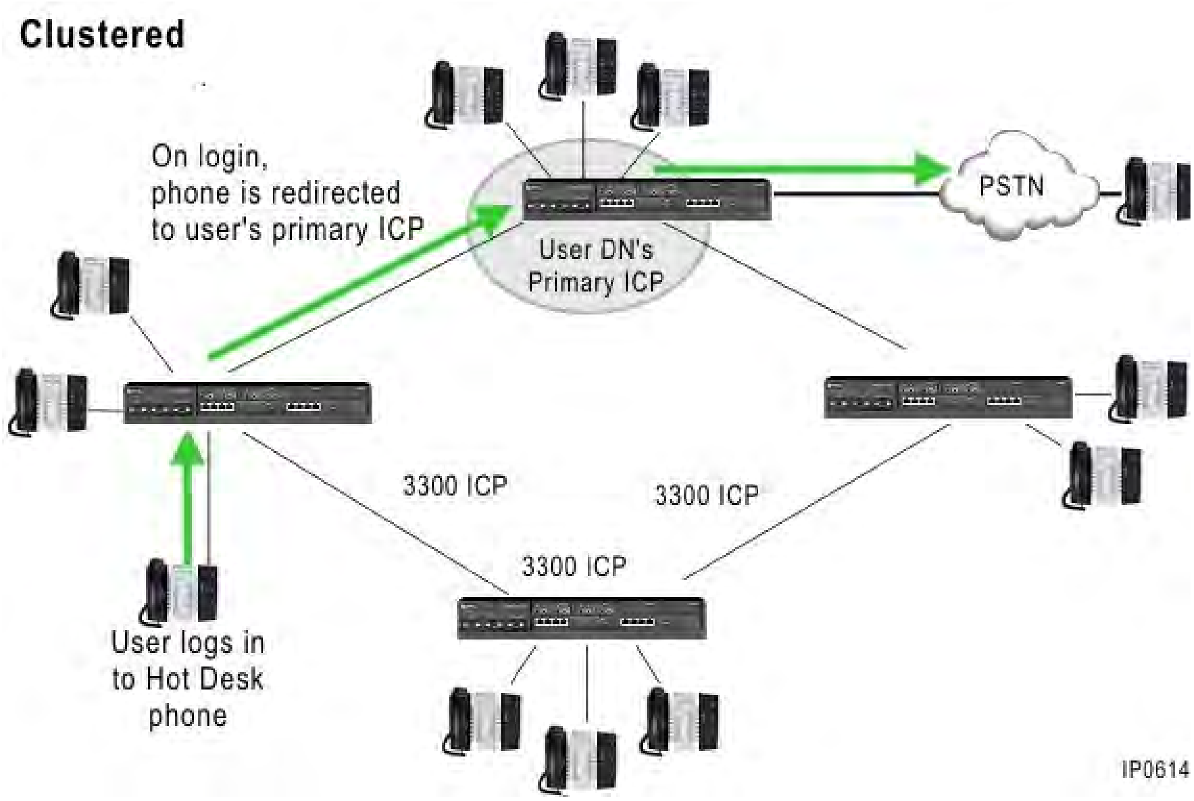


Figure 8: Clustered Hot Desking

Mitel supports resilient Hot Desking. If both the set and user are programmed for resiliency, then the Hot desk user will not lose service if the host 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.

Multi-device Capability

Mitel provides a multi-device capability: instead of one phone being used by many users, each user has multiple devices and requires a single User licence only.

External Twining offer a less costly Multi-device solution for users who require two devices, for example a desk phone and a cell phone, without consuming a Multi-device license.

Embedded Unified Messaging

Embedded Unified Messaging (UM) enables users to receive and manage voice messages through IMAP-enabled e-mail clients such as MS Outlook or Outlook Express. Message states are synchronized between the e-mail client (e-mail message store) and IP phone (voice message store). Secure connections to the IMAP server are made through TLS or STARTTLS.

Embedded UM is not compatible with Standard Unified Messaging. Only one or the other can be enabled for a mailbox.

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.

The voice mail system includes the following features:

- Standard Unified Messaging enables users to forward voice messages, including Record-a-Call messages, to e-mail addresses. Users can manually forward individual voice messages, or automatically forward all voice messages.
- An automated attendant plays different greetings during and following business hours, provides a company directory that uses extension numbers or names as the dialing method, and allows single-digit option selection.
- A Multi-level auto attendant (MLAA) enables 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, or pre-recorded information, or to leave voice messages.
- Personal Contacts enables users to create a customized voice menu so callers can reach users on their cellular phone, or by fax, etcetera.
- User mailboxes can be password-protected.
- A tutorial assists new subscribers with mailbox setup.
- Messages can be quickly retrieved.
- Easy-to-use menus enable users to send urgent, private, or certified messages.
- Users are notified of any messages customers have left them.
- Users can record conversations and save them to their voice mailboxes.

Voice Profile for Internet Mail

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

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 and is compatible with Hot Desking.

Embedded System Management

Embedded System Management (ESM) includes the following end user tools:

- “Desktop Tool” on page 21
- “Administration Tools” on page 21

Desktop Tool

The Desktop tool is a web-based interface that enables IP phones users to

- Assign features to personal keys
- Manage personal contact lists
- Add and delete internet bookmarks

The following figure illustrates the Desktop Tool.

Figure 9: Desktop Tool Interface

Administration Tools

Group Administration Tool

The Group Administration Tool is web-based interface that enables administrators to configure and mange the following basic IP phone settings for group members:

- Basic system parameters
- The system phone directory
- Extension and group parameters
- Voice mailboxes
- Group membership (add, edit, or delete users from the system directory)

- Users' personal keys

The following figure illustrates the Group Administration Tool.

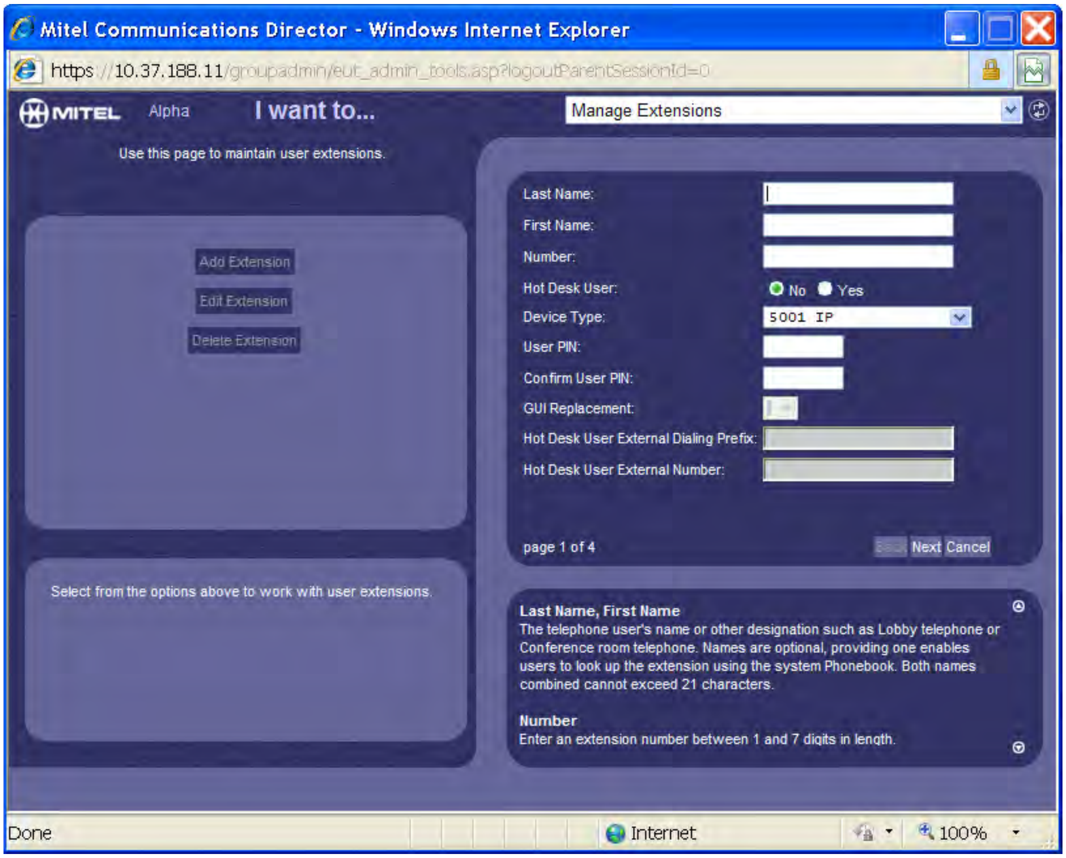


Figure 10: Group Administration Tool Interface

System Administration Tool

The System Administration Tool enables trained technicians and system administrators to program system-wide settings, voice settings (lines, extensions, management parameters, system directories, and voice mail) and IP network features. The System Administration Tool provides access to Maintenance Logs, Software Logs, and Login and Logout Audit Logs.

The System Administration Tool includes several forms.

The User and Device Configuration form provides administrators with the following capabilities:

- Consolidated view of user or device information: this simplifies the add, modify, and delete functions for users and devices and reduces the number of times the same data is entered into the system.
- Copy user functionality: administrators can quickly create new entries using existing user or device settings and configurations.

- Import capability: administrators can quickly collect and import user and device data using Microsoft Excel spreadsheets. These spreadsheets contain built in validation similar to ESM data entry rules which helps reduce errors.

The following figure illustrates the User and Device Configuration Form.

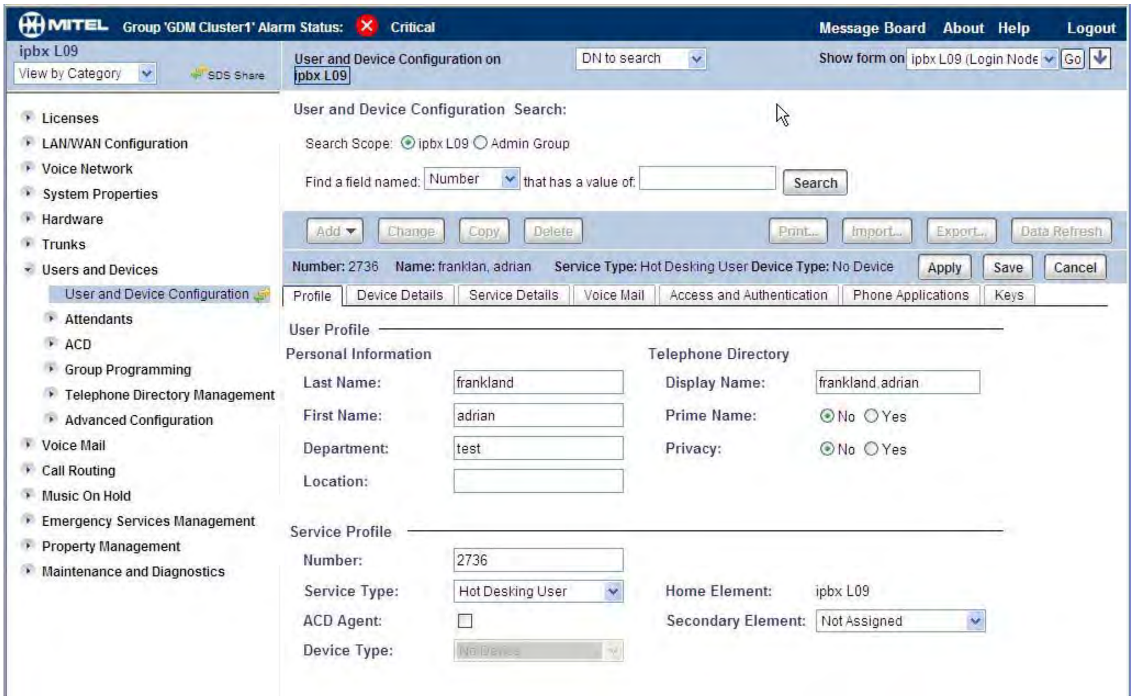


Figure 11: User and Device Configuration Form

The Scheduler form is used to schedule system events to run automatically. For example, you can create an event that causes the system to switch to night service every evening, and another event that causes it to switch to day service every morning.

To reduce management overhead and improve productivity you can schedule the following events to run automatically:

- Backups
- CSV File Import/Export
- File Transfers
- IDS Synchronization
- Night/Day Service

The Scheduler tool can also automatically log out Hotdesk Users at a set time.

The Scheduler tool includes a calendar that can be updated with holidays. When you add an event, you can specify a repetition interval, such as daily or weekly, and indicate whether the event should run on holidays or only on weekdays.

The following figure illustrates the Scheduler Form.

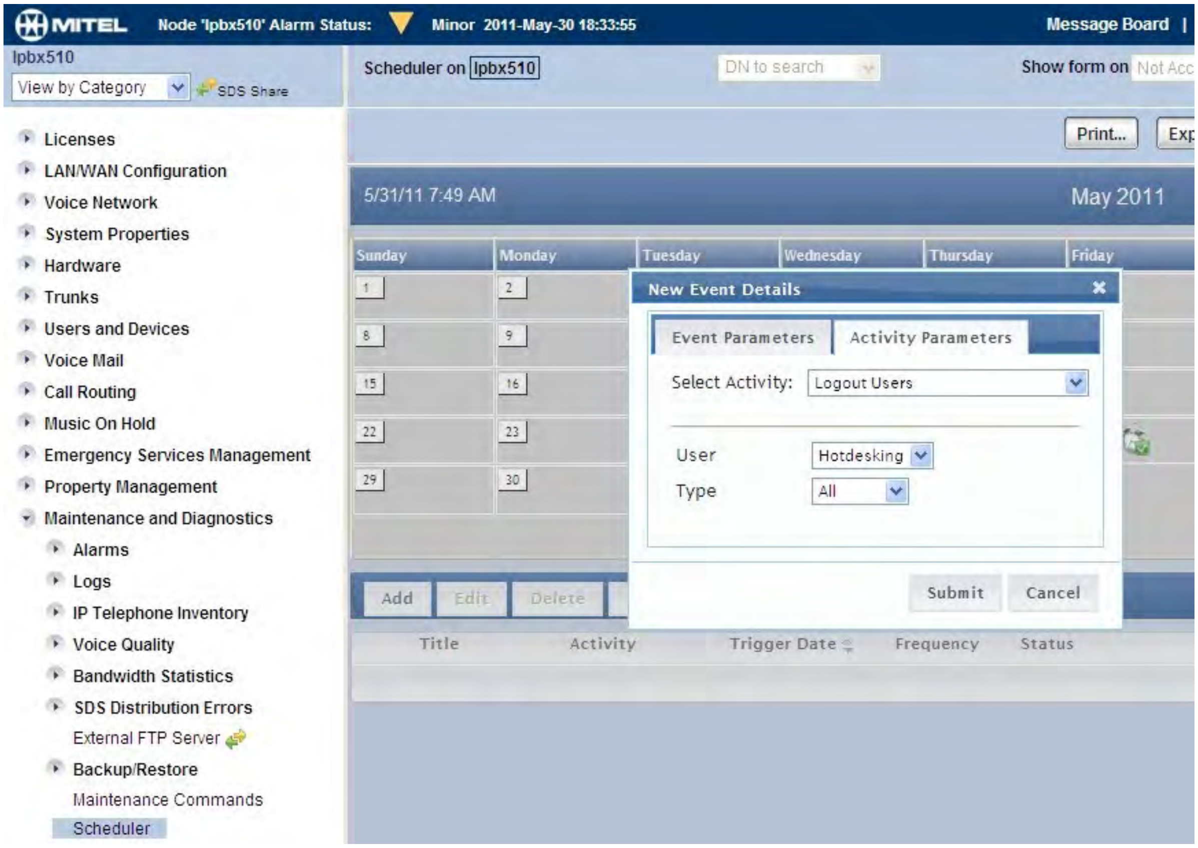


Figure 12: Scheduler Form

The System Administration Tool

- includes Audit Logs that provide a historical record of changes made to the system from the System Administration Tool and various other user interfaces and applications. This assists with troubleshooting problems that arise, enabling you to determine who, in a multi-administrator system, is responsible for a particular change.
- supports Range programming. Range programming speeds up MCD programming and configuration by enabling the administrator to program repetitive data using a single command. The administrator can also print forms and form data.
- includes data import functionality that enables administrators to quickly import large numbers of new users and devices via a .CSV format file. Administrators can collect a substantial configuration data in the spreadsheet file and then import it directly into the MCD database. The import functionality eliminates the need to manually enter configuration data for each user or device and reduces the likelihood of data-entry errors. Technicians can import new user data when setting up a new system and administrators can import large numbers of users or devices whenever they need to be added.
- enables you to maintain a small group of network elements (20 or fewer) effectively and conveniently without the need for a management tool such as Enterprise Manager. Administrators can "reach through" to the System Administration tool of any network element to

program it, and backup all databases from a single session on a network element. For additional details, refer to the *Mitel Communications Director System Administration Tool Help* available on Mitel's edocs website.

Note: Network elements must be grouped together within an SDS Administrative group to use this feature.

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

Alarms Management

The 3300 ICP system raises an alarm when an anomaly is detected and corrective action is required. The system continuously provides attendants who are using Mitel consoles with alarm status information. You can program alarm threshold levels. There are three classes of alarms:

- **Critical:** indicates a loss of service that demands immediate attention.
- **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

Administrators can set up remote alarms to notify technicians of critical, major, or minor alarms. MCD e-mails the notifications to up to 10 addresses. Prompt notification helps ensure issues are addressed quickly.

MCD supports Simple Network Management Protocol (SNMP). SNMP defines asynchronous messages called "traps". Administrators can set up SNMP traps to monitor system devices and functions. SNMP traps are generated to alert administrators to significant events (for example, when alarms are triggered or cleared).

Traps are sent by the SNMP agent in MCD to an SNMP manager to report error conditions and other types of messages. 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.

Controlled System Access

System Administrator Policies enable you to control access to System Administration Tool forms for individual users. When you create a policy, you set permissions that grant Read or Read/Write access to forms. Denying access to a form hides it from view.

You can enable remote access to forms and distribute policies to all platforms in a MCD cluster using System Data Synchronization.

Mitel offers Management Access Point (MAP) to provide secure, controlled access to systems and system tools from remote locations.

IP Phone Analyzer

IP Phone Analyzer is a Windows application that collects performance information from IP Phones on a network. Technicians can 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 IP Phone Analyzer. Technicians can direct phones to new IP Phone Analyzer addresses via a MCD Maintenance task. This eliminates the requirement to reset the phones manually.

IP Phone Analyzer provides information in four views:

- **Status View:** displays the status of each phone registered with IP Phone Analyzer, 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 analysis.
- **Packet History View:** sorts messages received by IP Phone Analyzer.
- **Call Statistics View:** displays call statistics, including RTP statistics, collected from IP sets.

System Data Synchronization

System Data Synchronization is an enabling technology that

- Reduces the time to provision and administer multiple MCD nodes by automatically updating common data changes around all of the relevant nodes without any administrator intervention
- 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 enables administrators to synchronize database information among a network or cluster of MCD systems. Database changes made to a platform in the network or cluster are applied to the other platforms.

Hospitality

Mitel is renowned for delivering comprehensive solutions for the hotel industry—from small hotels/motels through to large resorts and cruise ships.

The new centralized hospitality solution

- Meets the needs of international markets—including Europe, the Middle East, and Asia Pacific—and supports multi-national guests upon check-in
- Employs leading-edge technologies to deliver customized applications, such as large display touch screen phone sets for high-end boutiques and hotels
- Enables our partners to create custom HTML applications for large screen phone displays for resort amenities, restaurants, advertising, etcetera, to enhance service
- Provides full-service integration with wireless SIP devices with embedded resiliency support for third-party devices
- Provides enhanced analog scalability, centralized administration, and a redundant CPU IP platform to accommodate large hospitality deployments
- Provides greater capacity for devices and suites and comprehensive support for numerous PMS and Call Accounting solutions. See “Centralized Hospitality Deployment” on page 29.

MCD provides the features commonly used by hotels, motels, cruise ships, and hospitals. It also provides elite features required of full service hotels, such as VIP status, automatic personal wakeups, multiple languages, and maid identification.

Using the Hotel/Motel feature, receptionists, operators, and front desk personnel can

- View information on guest rooms, guests, and room extensions. Receptionists can add and edit guest room information.
- Check in and check out guests, thus keeping track of arrivals and departures.
- Set the condition and occupancy status of rooms. Receptionists can set the condition and occupancy status of rooms or maids can do this by entering personal ID codes—clean, not clean, maid present, out of service, to be inspected. Requiring maids to identify themselves when indicating room status changes enforces accountability and enhances quality.
- Search for rooms by using room condition and occupancy status as search parameters.
- Assign VIP status labels to room extensions so hotel employees respond accordingly when VIP guests call the front desk or other hotel extensions. Assigning VIP status labels to rooms optionally triggers the system to provide personal wake ups when VIP guests set the wake-up feature. In addition, hotel employees can associate labels to room extensions to provide insight as to the purpose of their stay (for example, Honeymoon, conference, wedding) or to provide language identification so calls are answered in the guest’s language.
- Enable Automatic personal wakeups for specific VIP or non-VIP guest rooms so that personal wake-up calls for the guests are automatically set.
- Change the language of phone display prompts and applications for the phones in a guest room at 'check in' from the attendant console or the PMS.
- Select up to fifteen different languages including English, French (Canadian), French (European), Italian, German, Spanish (European), Spanish (LA), Dutch, Portuguese (Brazil), Portuguese, (European), Romanian, Russian, Swedish, Polish, Simplified Chinese, and Arabic on IP phones only.

- Set Do Not Disturb (DND) from the Guest Services Application or the PMS for individual rooms or suites to prevent calls from ringing guests’ phones.
- Listen to room monitor extensions on analog phones.
- Enable Call Blocking to prevent calls from being made between guest rooms.
- Restrict the types of calls that guests can make from room extensions.
- Use Message Registration to calculate the total cost of calls made from individual room extensions.
- Print Automatic Wake-up, Room Status, and Message Registration reports.
- Access logs generated by the system during operation of the Hotel/Motel feature.

Property Management System

MCD can work independently or in conjunction with a Property Management System (PMS). PMSs provide

- Reservation control
- Centralized accounting and billing
- Call logging

IP-enabled PMS applications can communicate directly. 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 guest information is changed in the PMS system, the PMS sends messages to the PBX.

Mitel supports connections to a range of PMS systems.

Clustered Hospitality

Clustered Hospitality provides hotel/motel feature functionality across MCD/3300 ICPs clusters A cluster is comprised of a single Hospitality Gateway and one or more Hospitality controllers.The Hospitality Gateway is the interface to the PMS and the Guest Services Application (GSA) on the 5550 IP Console, and can also host guest room extensions.

The following figure depicts a typical clustered hospitality environment.

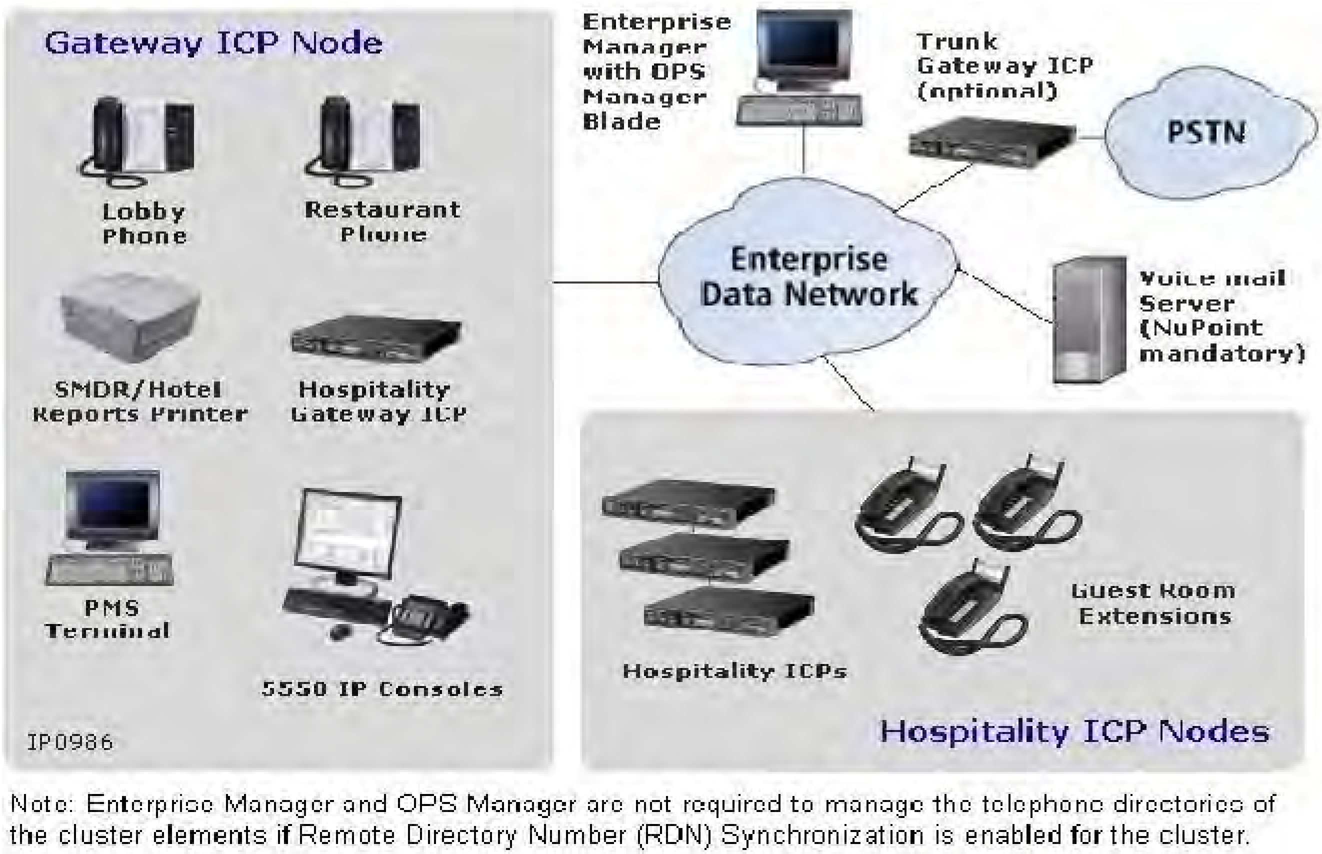


Figure 13: Hospitality Deployment Example

Clustered Hospitality supports

- Hotel logs and reports via a networked printer
- Shared Telephone Service (STS), available if all members reside on the same MCD as the linked suite.
- Configuration of room extensions and suites from any MCD in the cluster
- Resilient hotel room extensions
- PMS GRS General Reset/Get Reservation Status (a PMS function that synchronizes its check-in/check-out data with the MCD controllers to ensure the data on both systems matches on a cluster-wide basis)

Centralized Hospitality Deployment

In addition to Standalone and Clustered Hospitality solutions, MCD accommodates large-scale analog Centralized Hospitality deployments.

The architecture provides scalability and centralization. MCD can automatically extend calls across a cluster to analog ports on AX nodes.

A single IP controller functions as the Suite Hospitality Controller and one or more AX nodes provide connectivity for analog devices. The Suite Hospitality Controller performs processing and management tasks, hosting the Hotel and Motel Features and Reports, PMS and SMDR interfaces, GSA, and attendant consoles.

All call processing is performed locally. Calls to the suite ring all members (who are local or across the cluster). Although the analog ports may be situated on other nodes in the cluster, call processing is managed locally within the Suite Hospitality Controller as a single, standalone hospitality solution.

With this architecture, Suite Services continue to be supported, including wakeup calls and MWI notification. The central Suite Hospitality Controller supports direct connections to Attendant Console Guide Services, PMS and Call Accounting Packages, and to IP and SIP phones.

5540 IP Console

MCD continues to support IP phones. In this architecture, remote ports are considered to be local and 5540 IP guest services can be used in large-scale deployments. For deployments that require over 1000 ports, MCD now uses a smaller footprint as phone appliances are clustered.

SMDR Data

This architecture greatly simplifies the number of SMDR data streams required to support call accounting and billing applications because there is now only one SMDR stream from which to collect data. Since the call control is processed on the central Suite Hospitality Controller, there is no need for SMDR consolidation.

Capacity

Each fully loaded AX node supports a maximum of 288 ONS ports and the central MCD ISS can support a maximum of 5000 devices and over 2000 suites.

Redundant CPU Platform

MCD now supports the Stratus ftServer 2600 and 4500 industry standard servers from Stratus Technologies. The Stratus servers have a fully redundant hardware architecture in which both CPUs function in lockstep, enabling MCD to operate on a CPU redundant hardware IP platform. All key hardware components, such as the CPU, memory, motherboards, I/O, hard discs, are duplicated: processing is uninterrupted in the event of a component malfunction.

Although it appears there are two system, Stratus ftServer presents users with a single view. A single copy and installation of the operating system and MCD instance are required.

Tenanting

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

You can allocate consoles, CO trunks, and dial-in trunks individually to tenants or share them between tenants. You can switch to night service centrally, or on a tenant-to-tenant basis. You can block calls through the system 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 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 or 112), a Caller's Emergency Services ID (CESID) is sent from the MCD system to the Public Safety Answering Point (PSAP). Note PSAP is relevant only 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, mobile devices, and generic SIP devices.

You can configure CESID for mobile directory numbers. Although any hot desk user can have a CESID, only External Hot Desk Users (EHDUs) on external trunks can make use of it. Regular hot desk users and EHDUs logged in to MiNET devices will continue to use the CESID associated with their set's registration DN.

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 can be moved from one location to another, by the user, without the need to manually update to the CESID database because MCD automatically updates CESID.

In order to update CESID automatically, the network environment must have Layer 2 (L2) switches that are all configured for Cisco Discovery Protocol (CDP), Spanning Tree Protocol (STP), Link Layer Discovery Protocol (LLDP), or all of them. The system automatically updates CESID 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.

Device Move Detection over the LLDP is only supported by the 53xx and 5540 sets.

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. 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 93.
- If Emergency Call Routing is configured in MCD, the user picks up the teleworker phone and simply dials the emergency number.

Multi-Level Precedence and Preemption

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

- Specify a precedence level when they make a call
- Preempt calls that have a lower precedence level

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

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

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. For example, a call with a precedence level of FLASH can preempt a call with a precedence level of IMMEDIATE. You can designate users as either preemptable or non-preemptable. Non-preemptable users can still assign precedence levels to calls.

Enterprise Licencing

Mitel Enterprise Licensing (Licence Sharing) is an additional Licence Manager capability available for MCD Enterprise Systems.

Enterprise Licensing allows a customer to easily move licences around their solution with no interference from Mitel, the Solution Provider, or the AMC. By activating Enterprise Licensing, the customer is setting up MCD licensing to work as a single solution rather than a group of individual licensed nodes.

There are many advantages to licensing a group as a single entity including licence flexibility, ease of administration, and lowered cost of ownership.

There are 3 different types of licence management available for MCD systems:

Standalone Systems

Standalone Systems are licensed with a single Application Record for each system. MCD Licence Manager controls the licensing on individual systems.

Non Shared Enterprise Systems

Enterprise Systems licensed with individual Application Records cannot share licences. This licensing model is how all existing Enterprise systems that have upgraded to MCD 5.0 appear. Each system is linked to its own Application Record and is controlled individually by Licence Manager.

Enterprise customers who do not wish to invoke Enterprise Licensing can continue to licence their systems individually.

Shared Enterprise Systems

MCD 5.0 introduces shared Enterprise System licencing. Enterprise licensing allows groups of MCD systems to be amalgamated into a single Application Group at the AMC. The customer chooses a single system within the group of systems to act as the Master Licence Manager Designated Licence Manager (DLM) which connects to the AMC Application Group. All the underlying Group systems licensing is controlled by the DLM.

MCD System Functionality

This chapter describes MCD system functionality. For details on system configurations, refer to the *Mitel Communications Director Engineering Guidelines.* For detailed descriptions of hardware components, refer to the *Mitel Communications Director Hardware Technical Reference Manual*.

3300 IPC Hardware Overview

3300 ICP Controllers provide the voice, signaling, central processing, and communications resources for the system.

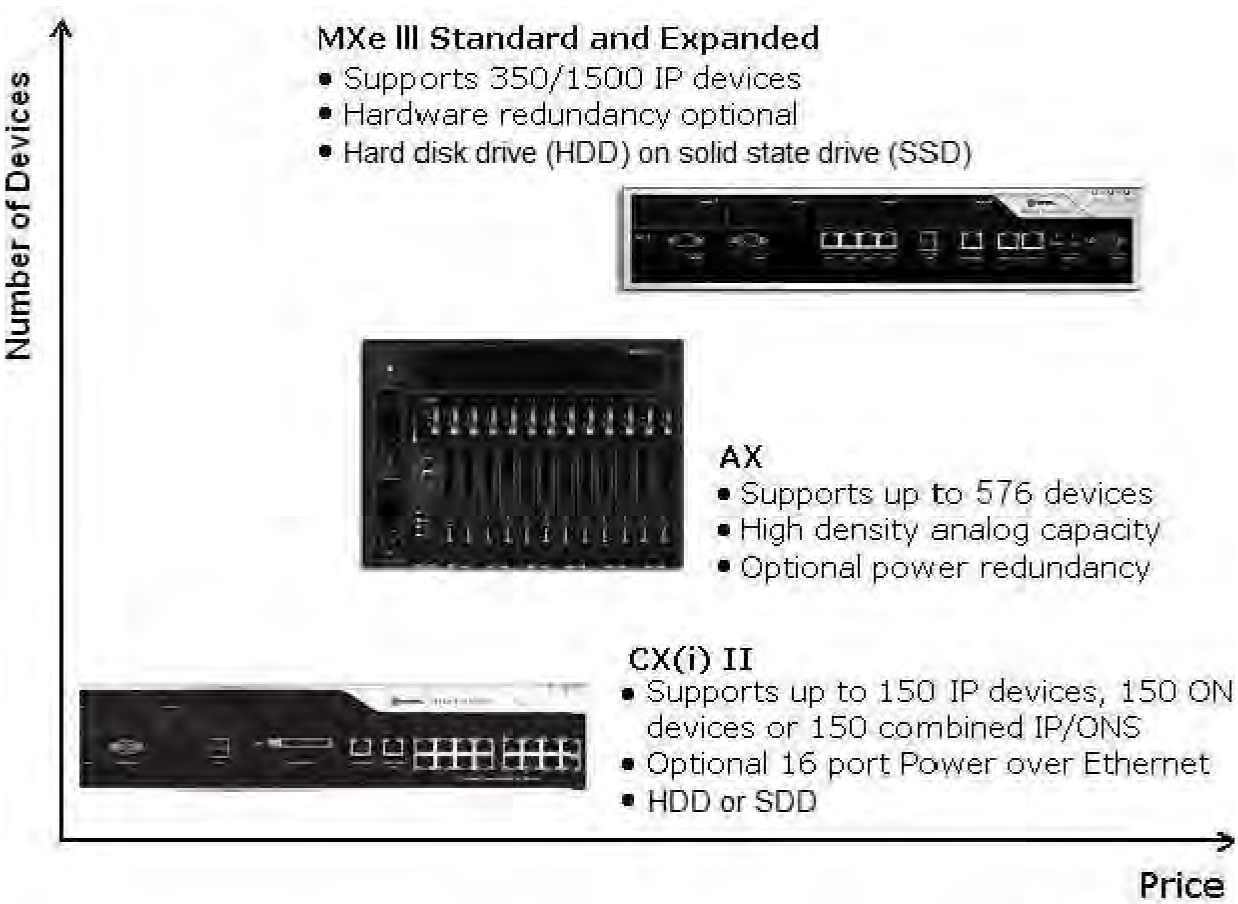


Figure 14: 3300 ICP Controllers — Scaling to Site Requirements

Mitel offers five types of controllers that scale to meet the requirements of small-to-large 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 288 IP sets, for a combined total of 576 devices. Up to 300 IP devices can be supported under low traffic conditions.

- **MXe III Standard Controller**: supports a maximum of 300 IP devices or 350 ONS devices (or a combined maximum of 350 IP/ONS devices)
- **MXe III Expanded Controller**: supports a maximum 1400 IP devices or 1500 ONS (or a combined maximum of 1500 IP/ONS devices)

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 Dual FIM, Dual T1/E1, T1/E1 Combo Card, and Quad BRI modules, 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:** the LEDs are located on the front or rear of the units for 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 controller capacities, refer to the controller configuration tables in the *3300 Integrated Communications Platform Engineering Guidelines*.

CX II and CXi II Controllers

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

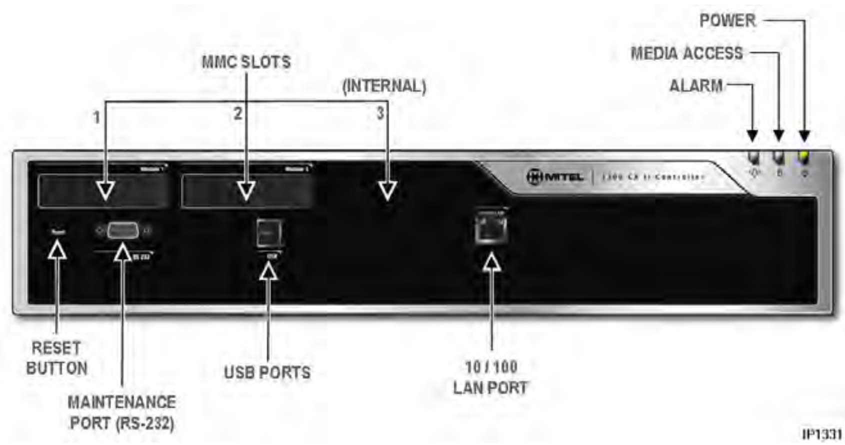
The CX II and CXi II Controllers support

- 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 BaseT LAN port (RJ-45 connector)

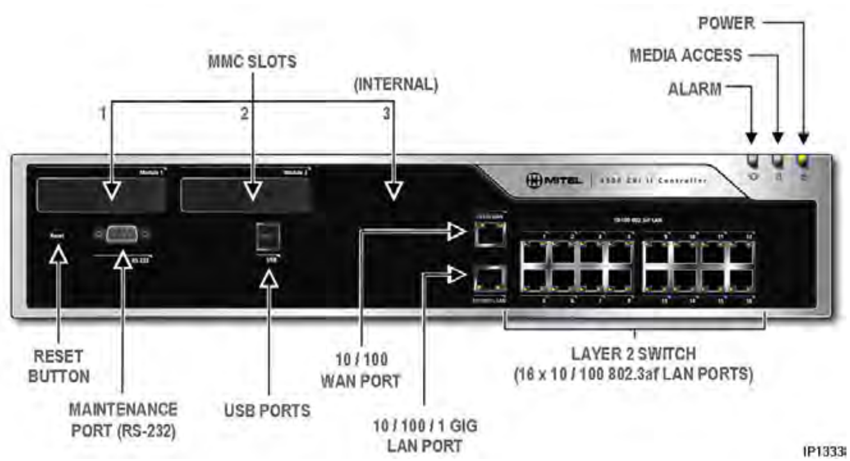
- Sixteen 10/100 BaseT LAN ports connected to an internal Ethernet Layer 2 switch (CXi II Controller only)
- SATA solid state drive or SATA hard drive for software storage

Optionally, you can install

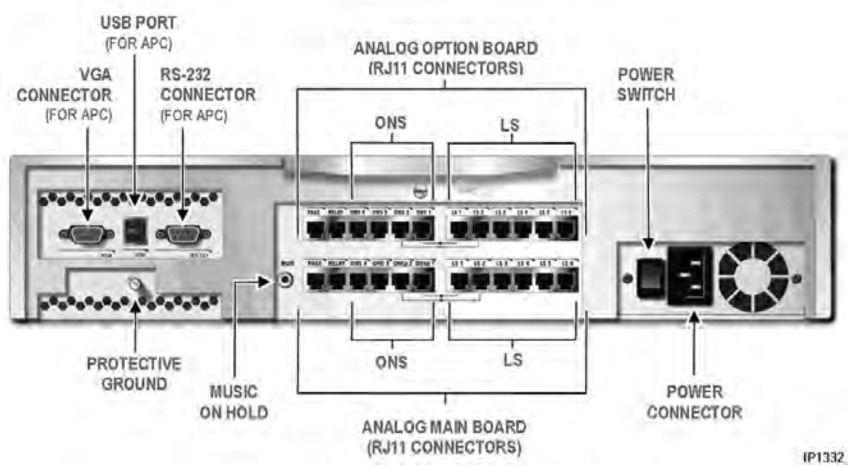
- The Analog Option Board (AOB): provides six LS trunks ports with CLASS support, four ONS ports, one System Fail Transfer circuits and one paging circuit
- One DSP II module for FAX Relay (T.38) / compression
- One or two T1/E1 Combo modules for digital trunking
- One or two Quad BRI module for BRI trunks
- A Quad Copper Interface Module (CIM) for connection of up to three Analog Service Unit IIs (ASU IIs)



CX II FRONT PANEL



CXi II FRONT PANEL



REAR PANEL

Figure 15: CX II/CXi II Controllers

AX Controller

AX Controller provides support for IP devices and analog devices and is ideal in situations that require a high density of analog devices. AX Controller be deployed as a standalone system or in a network of systems to provide additional analog support.

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 AX is installed in a low traffic environment (for example, Hospitality), it 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)
  - DSP II (internal or external)

Optionally, you can install

- A second AC Power Supply Unit (PSU) for power redundancy
- Line cards

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

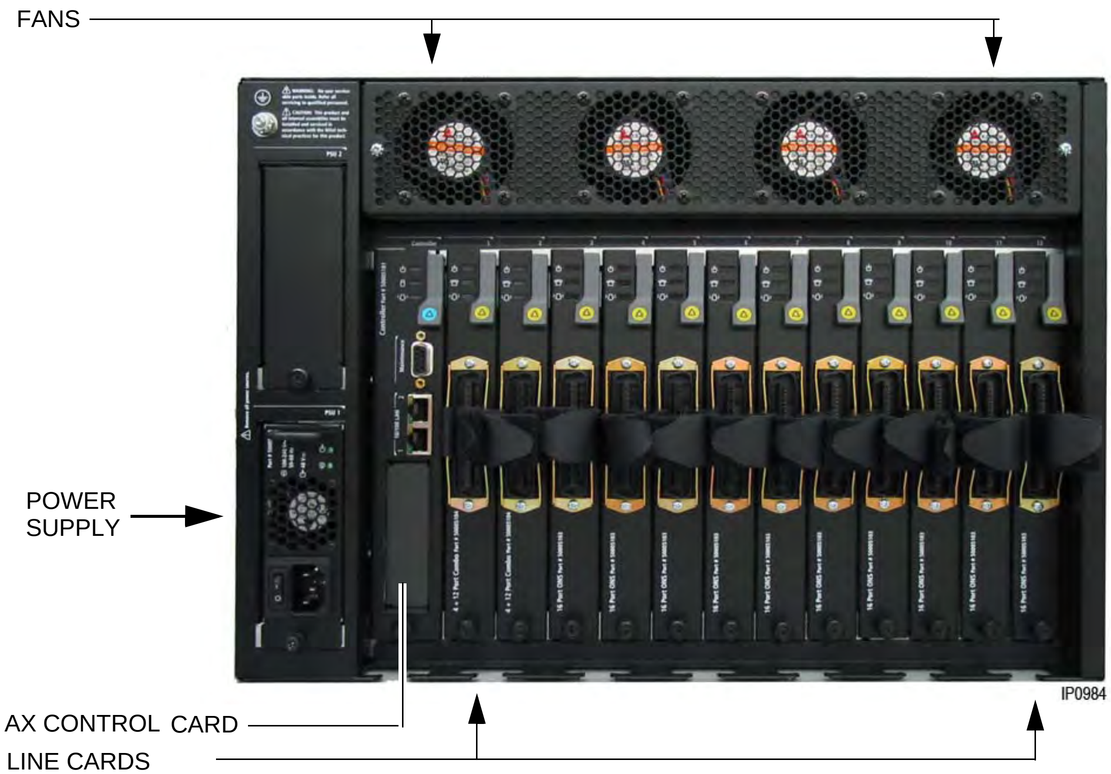


Figure 16: AX Controller Rear View

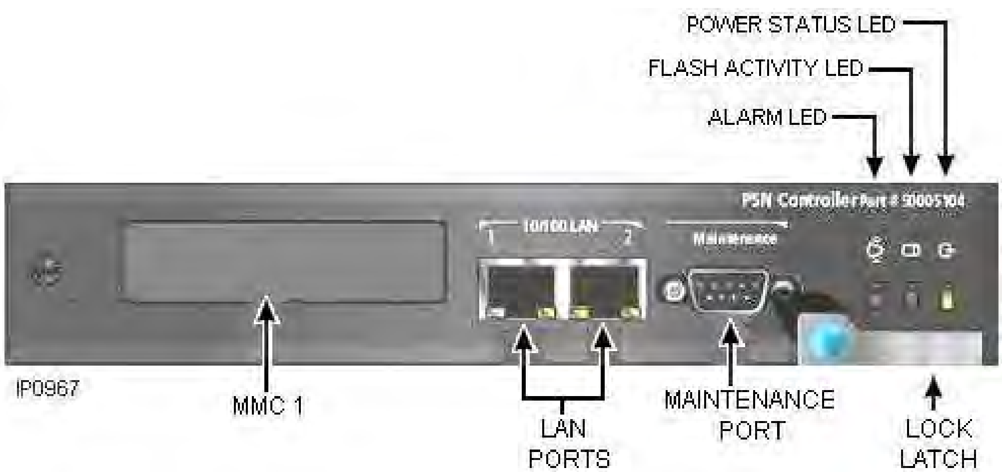


Figure 17: AX Controller Control Card

MXe III Controller

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

The MXe III Controller supports up to

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

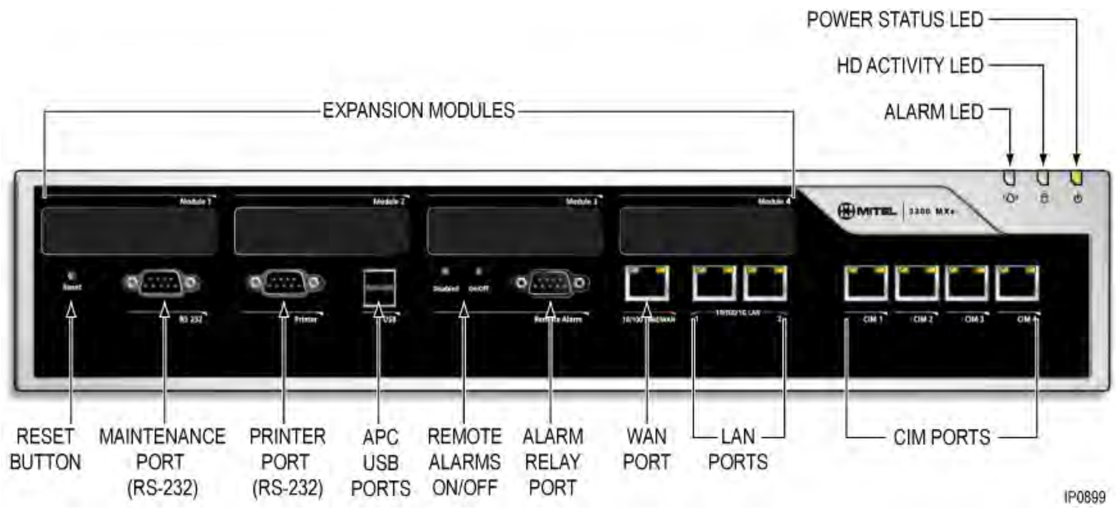
The MXe III Controller can host up to seven bays (North America only) providing connectivity for 96 ONS or OPS devices per bay. Only BCCIII-equipped bays are supported. Trunk cards are not supported.

The MXe III Controller provides

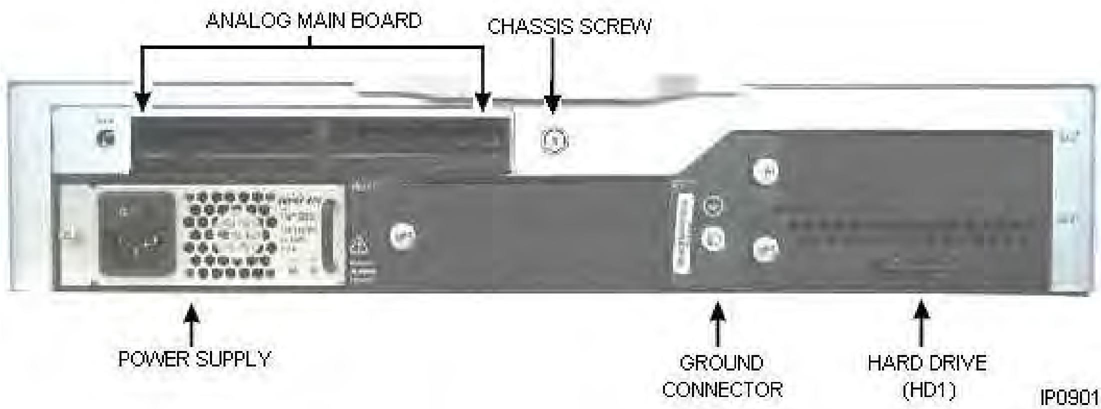
- Two 10/100/1000 BaseT Ethernet LAN ports (RJ-45 connector)
- One 10/100 BaseT Ethernet WAN port (RJ-45 connector)
- Four externally accessible slots and two internal slots for optional modules
- Four CIM ports
- An Analog Main Board that provides 6 analog trunks and 4 analog extension ports
- An alarm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

Optionally, you can install

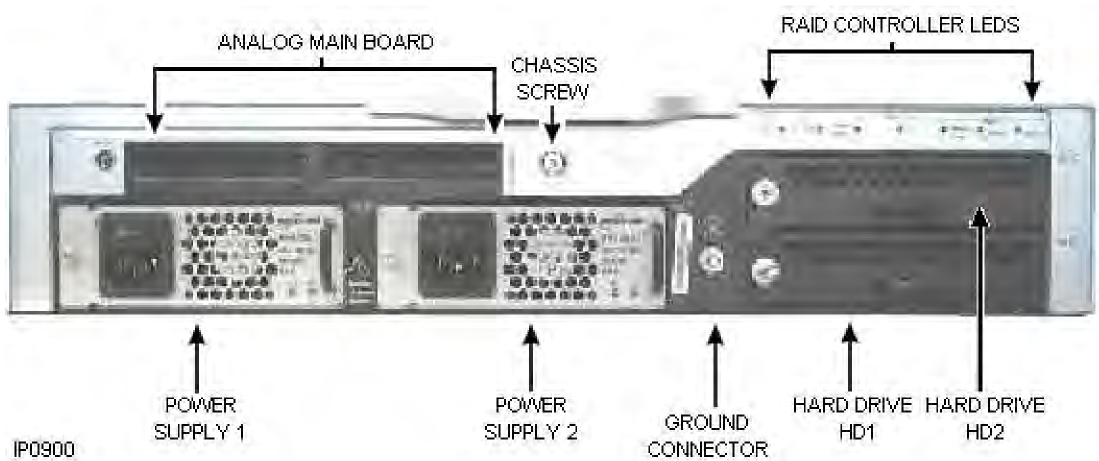
- MXe II Expanded Processor Package to upgrade from standard capacity (350 devices devices and 64 E2T channels) to expanded capacity (1500 devices and 128 or 192 E2T channels)
- Two Quad DSP modules for G.729a compression
- Two octal DSP II modules for G.729a compression and T.38 FAX support
- Up to four Dual FIMs for connecting NSUs, peripheral units, and bays
- Up to four Dual T1/E1 modules
- Up to three T1/E1 Combo modules
- Up to three Quad BRI modules
- 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 18: MXe III Controller

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 III Controllers use a single processor to perform the Real Time Controller (RTC) functions and the Ethernet-to-TDM (E2T) functions. The expanded MXe III Controller has 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 available only for CX/CXi II systems. Instructions on how to calculate system DSP requirements are provided in the *3300 Integrated Communications Platform 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.

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	0	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
ISS VMCD	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.
- The MXe III 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.

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 platform delivers 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:

- The 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.
- The 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 phone 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 phones only; pulse or rotary dial phones are not supported.

Analog Main Board/Analog Option Board

The MXe II and CX (i) (II) Controllers support the Analog Main Board (AMB). In addition, the CX (i) (II) can support the Analog Option Board (AOB).

The Analog Main Board supports

- Six Loop Start (LS) trunks
- Four On-Premise (ONS) lines (the first 2 ports are surge-protected)
- 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 enables 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 digital trunk modules and Network Service Unit provide connectivity to digital trunks for public or private networks. The following table summarizes digital trunk support for the controllers.

Table 2: Digital Trunk Support

Controller	Available External Slots	Dual FIMs	R2 Network Services Unit	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	

Dual Fiber Interface Module (FIM)

The Dual FIM module allows you to connect

- NSUs to the MXe III Controller and AX Controller
- Peripheral Cabinets and bays to the MXe III Controller

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.

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 phone display screen.

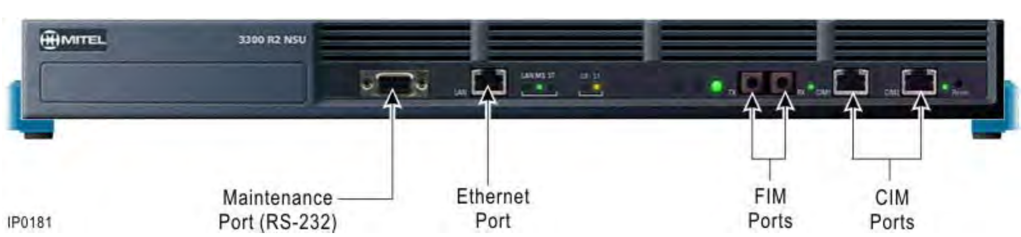


Figure 19: R2 Network Services Unit

Dual T1/E1 Module

The Dual T1/E1 module, supported in the MXe III and AX, 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.

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) Module

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

The BRI module has four ports that support 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

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

Bay Support

The MXe III Controller can support up to seven Bay cabinets. The 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 the MXe III using FIM or CIM cables.

Note: SX-200 Bays are supported in North America only.

Bay Peripheral Cards

The Peripheral interface cards connect peripheral devices (such as SUPERSET™ phones) to the system. They are located in slots 1 through 8 and include the

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

Each ONS port requires an ONS licence on MCD.

Digital Network Interface Line Card (North America Only)

The Digital Network Interface (DNI) line card provides an interface from the system to Mitel SUPERSET phones.

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 phones. 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.

MCD Network Support

Voice Networking Gateway Solutions

The use of IP telephony can result in cost savings and improve the number and quality of voice-related applications. Many end customers recognize the benefits of these enhancements but are hesitant to replace their entire voice infrastructure to gain benefits at a particular site. In such situations, using MCD as a gateway into IP telephony has many advantages.

By integrating MCD into an existing third-party PBX, customers retain their previous investment in communication equipment while taking advantage of the benefits of a superior IP telephony solution. MCD can connect to the third-party PBX using a variety of methods, building the gateway to IP telephony. This enables customers to use IP Networking, Collaboration, Mobility, and virtual Contact Center applications, as illustrated below.

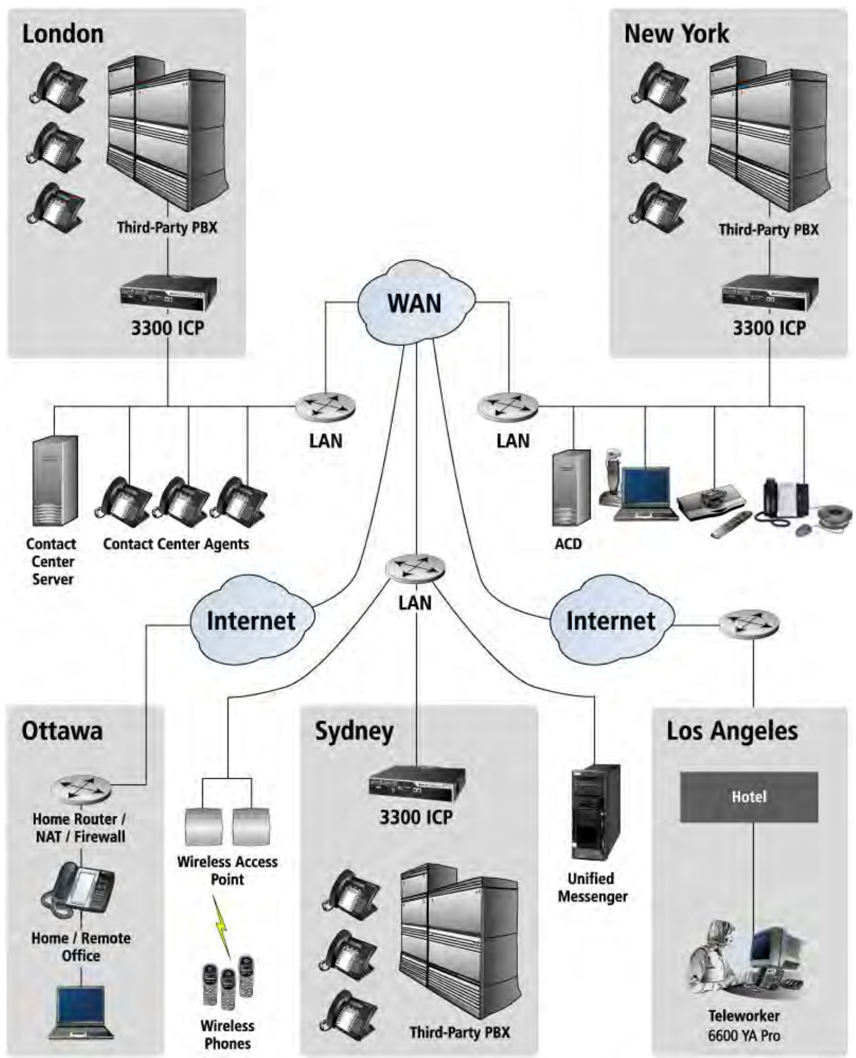


Figure 20: MCD as Gateway

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 Applications Suite” on page 51
  - “Mitel NuPoint Unified Messaging” on page 53
  - “Mitel Unified Communicator Express” on page 54
  - “Mitel Unified Communicator Advanced” on page 55
  - “Mitel Intelligent Directory Application” on page 56
  - “Mitel Live Content Suite” on page 57
  - “Mitel Applications Builder” on page 57
  - “Mitel Live Business Gateway” on page 58
  - “Microsoft Lync” on page 59
  - “Customer Interaction Solutions” on page 67
- “General Business Solutions” on page 71

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 Applications Suite

Mitel Applications Suite (MAS) integrates with Mitel Border Gateway (MBG) to provide small and medium-sized businesses with a set of advanced IP applications. MAS consolidates the installation and management of these applications on a single server to make it simpler to manage the information flow among customers, partners, and suppliers.

Co-residency is provided for multiple Mitel applications, including

- NuPoint (unified messaging)
- Speech Auto-Attendant,
- Unified Communicator (UC) Mobile
- Mitel Border Gateway with Teleworker, SIP Trunk Proxy and Web Proxy services
- Audio and Web Conferencing (AWC)
- Mitel Customer Service Manager (CSM)
- Business Dashboard

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.

MAS can be deployed as software on an industry standard server, as software pre-installed on a compact PC server, or as a virtual appliance.

MAS Support

MAS is supported as a virtual appliance within the VMware® vSphere™ environment for MCD. Virtual Mitel Applications Suite (Virtual MAS) leverages VMware VSphere 4.0 or 4.1 to enable businesses to consolidate Mitel's leading unified communications applications in the data center.

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:** Benefit from high quality audio and video that enables people to 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:** Send e-mail invitations with access codes, dial-in numbers, Web links and all the details participants need for effective meetings from a Web-based interface.
- **Lower Costs:** Reduce costly and inefficient travel, while avoiding the high costs of out-sourced conferencing services.
- **Easy Management:** This is deployed as part of the Mitel Applications Suite.

Audio and Web Conferencing provides

- 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

Mitel NuPoint Unified Messaging

Mitel NuPoint Unified Messaging™ is a scalable, integrated voice and fax unified messaging system that uses can access anywhere, anytime. NuPoint provides access to a host of flexible and customizable applications including Call Director, Speech Auto-attendant, and the Microsoft Office Live Communications Server® 2005/Office Communications Server 2007 integration. Simple and cost-effective configuration, implementation, administration, and management help streamline system management and deliver lower total cost of ownership.

Users can receive their voice mails and emails through one interface. On their desktops, users manage messages in Outlook. While on the road, they can phone into the Mitel NuPoint server and listen to their voice mails and emails. This simplifies the end user experience and increases productivity.

Mailbox users access these capabilities through the NuPoint Telephone User Interface (TUI) or Outlook (using the NuPoint Outlook Client Plug-in). When using the Outlook Client Plug-in, users are able to record, playback, forward and reply to voice mail messages and view fax messages. Text to Speech (TTS) enables users to listen to their email messages through their NuPoint Voice Mail.

NuPoint is available as part of the MAS implementation and as a standalone solution, and can scale to provide voice applications to large enterprises that demand high capacity, high availability, and resilient services. You can network two active NuPoint 640s to a single direct-connect storage array, enabling the 640's to store all data that relates to NuPoint users and NuPoint system data in one shared database. If a single 640 fails, the remaining one in the cluster continues to operate.

The NuPoint 640 has been engineered so there can be no single point of failure. It ensures high availability and minimizes unplanned downtime, and can be further integrated into an organization's data center infrastructure.

You can use NuPoint to

- Place calls to people/departments quickly and efficiently by speaking their names or phone numbers
- Page a mailbox owner when a new voice mail message arrives. NuPoint supports SMS notification to cellular phones. SMS notification text-messages users when they receive new voice messages.

- Allow callers to leave a voice mail message or input a call back number which is then displayed on the mailbox owners pager
- Schedule automatic wake-up calls to any phone at any date and time
- Record voice messages and have them automatically distributed to multiple users
- Deliver new, unplayed voice messages to an on- or off-system phone number of choice
- Route calls to predetermined destinations based on time of day, day of week, or day of year
- Property management integration and custom hotel prompts
- Configure up to six fax channels/ports for each NuPoint server. The Fax feature works in a network configuration where the NuPoint server is integrated directly with an MCD system or with another PBX.
- Perform Mailbox Maintenance, System Maintenance, Report Generation, and Call Director management from a web-based console
- Support integration with up to four MCD systems
- Enhance InBand integration to permits users to interface with legacy PBXs, using enhanced inband with DTMF
- Control voice mail functions through context-sensitive keys on the phone

For more information, refer to *Mitel NuPoint Unified Messaging General Information Guide* on Mitel edocs.

Mitel Unified Communicator

Mitel Unified Communicator enhances business communication and collaboration with co-workers, customers, and partners, helping employees to make better 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 system tray access to 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 IP Phones, resulting in a converged infrastructure that enhances 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 deployments.

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

Figure 21: UC Express Call Control

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 "launch pad" 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.

Figure 22 illustrates a call being escalated to a video call from the voice call and chat windows.

Figure 22: Unified Communicator Advanced

Mitel Intelligent Directory Application

Mitel Intelligent Directory Application is a free desktop application that enables users to access contact, calendar, and presence information for both corporate and personal directories directly on their phones.

Intelligent Directory Application provides a simple, intuitive, on-screen, searchable directory of both corporate (Microsoft® Active Directory®) and personal contacts (Microsoft Outlook®) on the phone's display, 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 (Office Communication Server 2007) to 5320, 5330, 5340, and 5360 IP Phone displays. Presence icons that appear on these phone displays provide presence indication for all corporate contacts.

Mitel Live Content Suite

Mitel Live Content Suite is a sophisticated, yet easy-to-use, web portal application for personalizing new live content applications and telephony functions on Mitel IP Phones.

Live Content Suite enables customers to create and publish dynamic and personalized information to users, transforming Mitel 5360 IP Phones into media information appliances. This improves communication with employees and enables them to readily access information.

Live Content Suite provides multi-language support and enables administrator to copy the programming from a source phone to a set of target phones, and permit specific users to program phones for their work groups.

Live Content Suite includes:

Live Desktop Portal

Live Desktop Portal is an intuitive, web-based phone programming portal. It provides users with a replica view of their phones and enables them to easily drag and drop content, applications, speed-dials, and telephony functions to touch-screen keys on their phones.

Live Blogger

Live Blogger delivers custom, live content to users' phones using standard blog tools. Live Blogger enables users to receive information from a central blogging source. Live Blogger uses a web server to provide the required functionality.

Live Applications

Live Applications are pre-packaged applications that deliver dynamic content to phones enabled with Live Content Suite. The continually expanding list of Live Applications includes Live Twitter® Reader, Live Weather, Live Flickr®, Live RSS List View, and Live RSS Page View.

Supported Mitel Phones

Live Content Suite supports the following Mitel IP Phones:

- Release 1.0: Mitel 5360 IP Phone
- Release 1.1:
  - Mitel 5320, 5330, 5340 IP Phones, providing full Live Content and Live Desktop Portal personalization capabilities
  - Mitel 5304, 5312, 5324, 5320, 5330, 5340, 5212 and 5224 IP Phones, providing full Live Desktop Portal telephony programming and administration capabilities

Mitel Applications Builder

Mitel Application Builder is an easy-to-use online wizard for creating hospitality applications that run on Mitel 5360 IP phones. The wizard guides you through the steps necessary to create a customized, on-phone hospitality application for Mitel 5360 IP Phone sets. The builder creates a hospitality guest application based on your input and selections.

Online Demo and On-Phone Capabilities

After you create an application, you can send an email link to customers that provides an online simulation of the application. You can also demonstrate the application to customers using a Mitel IP 5360 phone.

Ability to Push Live Advertising to Guest Phones

When used in conjunction with Mitel Live Content Suite, Mitel Live Application Builder enables hotel operators to create and manage live advertising content by simply editing a blog.

Supported Mitel IP Phones

Live Application Builder supports the Mitel 5360 IP Phone.

Mitel Live Business Gateway

Mitel Live Business Gateway enables Microsoft Office Communications Server 2007 to communicate with a MCD host platform. Users can see the telephony status and presence of other users on the network and can make calls from Microsoft Office Communicator 2007 using Mitel IP desktop phones. They can access key business resources quickly and efficiently and be notified with pop-ups when other users finish calls (become available).

Live Business Gateway integrates with Microsoft Outlook and Office to track all voice and IM conversations and any missed calls. Users can view the telephony status of business users and make calls from within these applications with the click of a mouse. Out of office messages and calendar information can be synchronized with Communicator 2007, enabling users to readily communicate their availability and whereabouts.

The combination of Office Communications Server 2007 and 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 Communication Server 2005 and Communicator 2005; however, in this document reference is made only to Office Communications Server 2007 and Communicator 2007.

This integrated solution from Mitel and Microsoft includes

- Microsoft Office Communicator 2007 R2
- Microsoft Office Communications Server 2007 R2
- Mitel Live Business Gateway 3.1 or greater
- Mitel 3300 IP Communications Platform (ICP) Controller

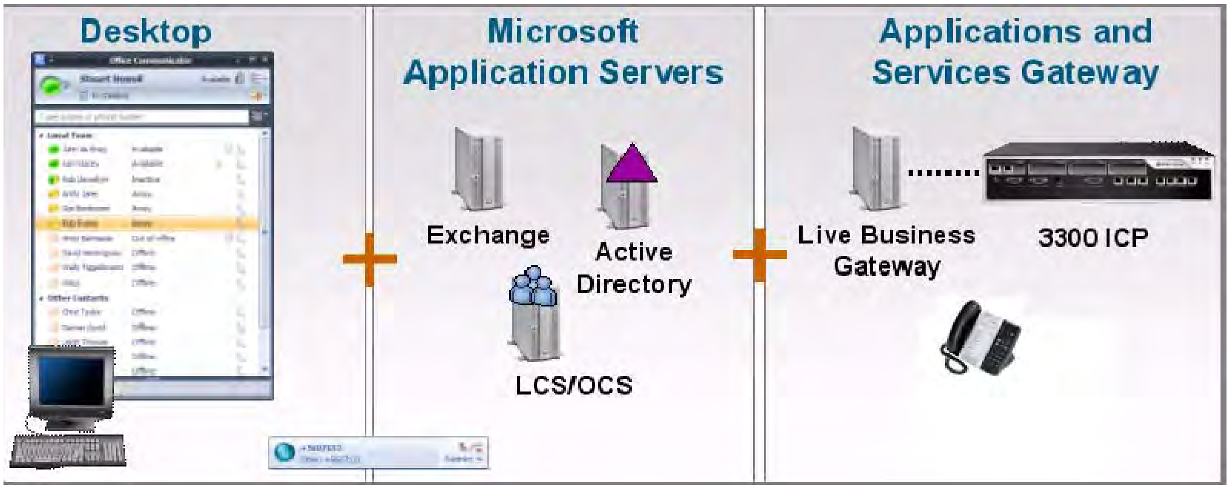


Figure 23: Live Business Gateway Integration

Microsoft Lync

Customers who choose Microsoft Lync as their preferred Unified Communications solution, often require integration to their existing IP - PBX, or if choosing a new IP - PBX, need to ensure that telephony integration with Lync is possible. Most businesses that deploy Lync use it for instant messaging, PC presence and collaboration, and a feature rich IP-PBX, such as MCD for telephony.

Customers who have deployed Live Communications Server 2005, Office Communications Server 2007 or Office Communications Server 2007 R2, are being encouraged by Microsoft to upgrade to Lync.

MCD release 4.2 SP1 and greater is Microsoft Lync Direct SIP certified. Mitel has successfully completed testing of Live Business Gateway release 3.2 with MCD 4.2 and Microsoft Lync. LBG 3.2 and Direct SIP are now fully supported by Mitel when used with Office Communications Server 2007 R2 or Lync.

Mitel offers a wide range of solutions, that in conjunction with MCD, provide telephony and presence integration enhancements to Lync and Office Communications Server 2007 R2.

Current telephony integration solutions include

- Mitel Live Business Gateway release 3.2
- Direct SIP
- Mitel 5550 IP Console presence integration for OCS 2007 R2
- UCA presence integration / federation
- Mitel Speech Auto-Attendant presence integration for OCS 2007 R2

Mobility Solutions

Mitel's Mobility Solutions are as follows.

Mitel Border Gateway Teleworker Service

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

MBG requires the following components.

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)	• 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)

You can configure the MBG teleworker service at the head office using a 5304, 5312, 5320, 5324, 5330, 5340, or 5360 IP Phone. Using a two-click process, the phone is set to operate in teleworker mode. The phone 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 figures 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.

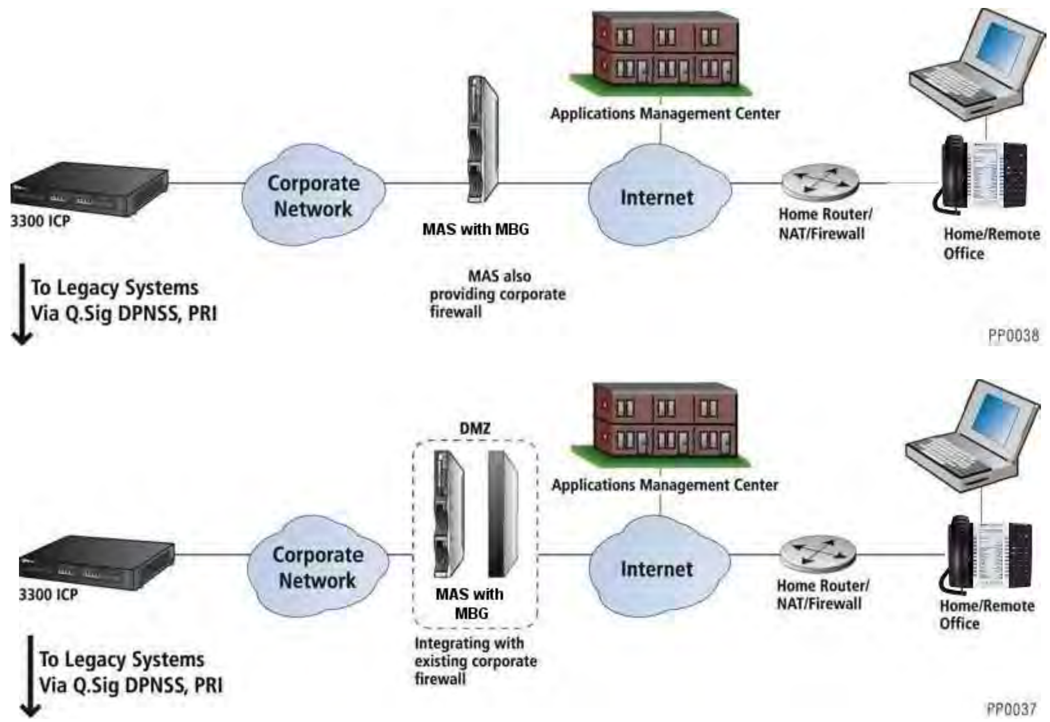


Figure 24: MBG Teleworker Service

Unified Communicator Mobile

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

UC Mobile extends commonly-used PBX features, such as hold and transfer, to cell phones so mobile workers can access Mitel's rich telephony features and applications while they are on the go. When configured as a twin, a cell phone acts as an extension of the enterprise desk phone, providing a single number contact and an integrated mobile and desktop experience.

A user can readily change the device/number that is twinned to his/her primary extension and take advantage of advanced capabilities delivered with 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 is only available within MAS.

Wireless Support

Mitel offers full-featured, integrated wireless IP solutions to suit your application, geographic location, and technology preferences. From DECT and Wi-Fi/802.11 solutions to Bluetooth Devices and DECT Cordless Devices for Mitel 5330, 5340, and 5360 IP Phones, Mitel's wireless IP phone devices provide users with the complete range of MCD features.

MCD supports the following wireless devices. See "Desktop Application Phones" on page 84 for more information on these devices.

Wireless phone	Wireless infrastructure	Comments
SpectraLink 8020, 8030 Wireless Handsets	Polycom certified enterprise grade Wi-Fi networks (www.polycom.com)	Available through Mitel/CommSource in select regions and through Polycom resellers globally; integrated over SIP
Mitel 5603, 5604, and 5606 Wireless Handsets	IP-DECT	Available globally; integrated over SIP
Mitel 5610 Handset and IP DECT Stand	IP-DECT	Available globally; integrated over SIP

The DECT Cordless and Bluetooth devices 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, provides the complete range of MCD features

IP-DECT System (Global)

The Mitel IP-DECT System (Global) can be deployed in any locality where the operation of devices in compliance with the European DECT or the North American DECT standards is permitted.

The IP-DECT System comprises the following components:

- 3300 ICP Controller
- Base stations for wireless coverage
- 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
- 5610 DECT: IP phone and stand for Mitel 5300 Series IP Phones
- 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 5603 and 5604 handsets are programmed on the 3300 ICP as specific device types; the 5606 series handsets are programmed as regular SIP extensions.

The system supports up to 1,000 users and 1,000 base stations per master, to a maximum of 10,000 users per system. See the diagram below for a typical configuration.

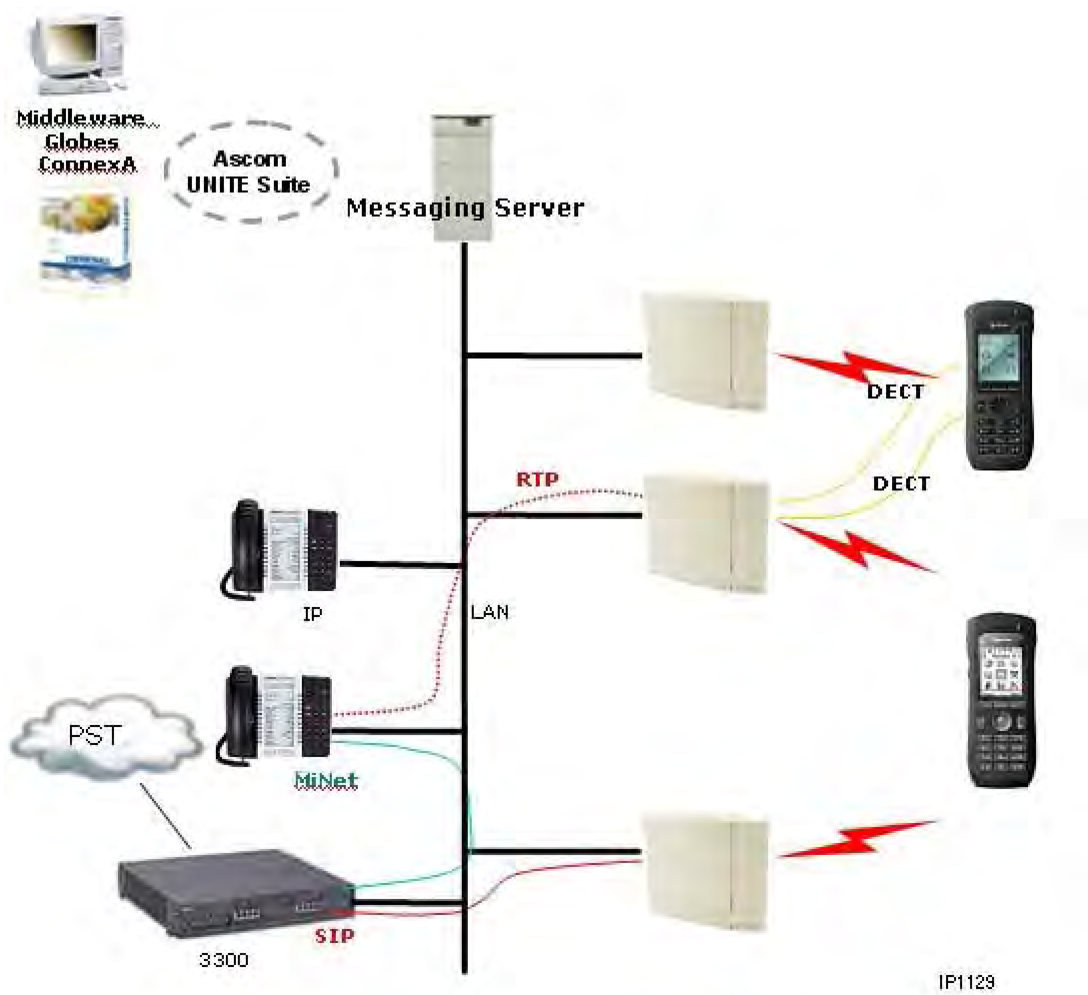


Figure 25: IP-DECT System (Global)

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
- 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 and base stations.

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

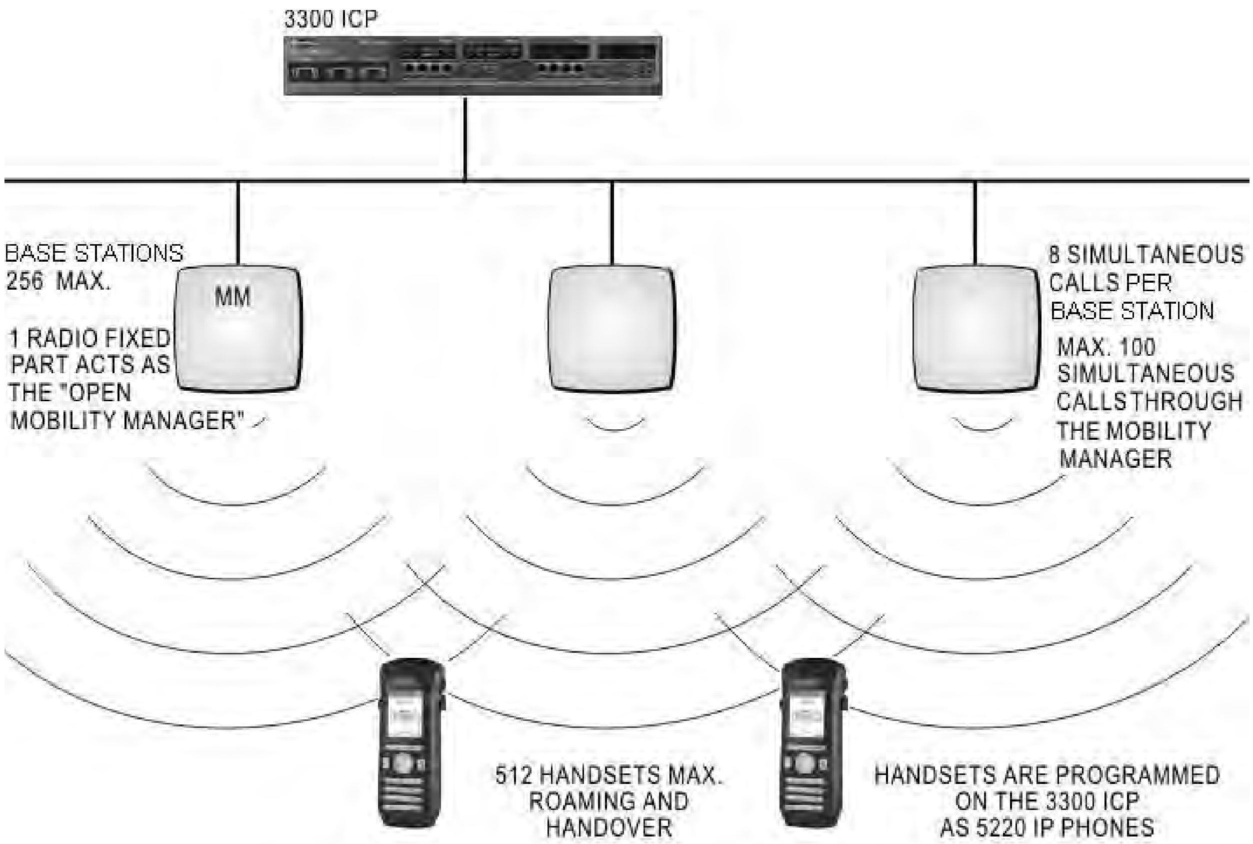


Figure 26: IP DECT Wireless Solution (EMEA version)

IP DECT Handset and DECT Stand

The Mitel 5610 Handset and DECT Stand are peripheral devices for 5300 Series IP Phones. To support up to eight 5610 Handsets for localized mobile communication, you simply snap the DECT Stand onto a phone. The handsets have vibrant color screens and full dial pads, 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 enabling customers to build wireless infrastructures using a choice of more than 27 different brands of Access Points. SpectraLink enables customers to consider out-of-building, campus-wide or even municipality-wide wireless voice networks. In addition, the SpectraLink Wireless Voice Solution enables 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 66). In addition, Mitel provides the optional SpectraLink Open Application Interface for two-way messaging.

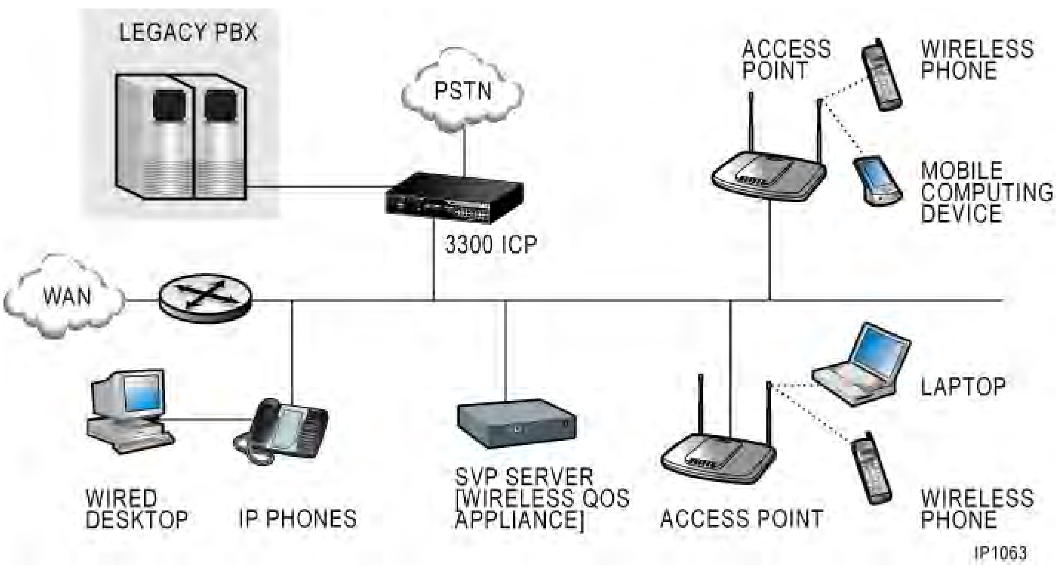


Figure 27: SpectraLink IP Integrated Wireless Voice Solution

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) enables handsets to function as two-way messaging devices: they provide integration with other enterprise systems and enable mobile workers to access critical information.

Customer Interaction Solutions

The Mitel Customer Interaction Solutions product suite combines 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 67
- "Applications for Formal Contact Centers" on page 68
  - "Mitel Contact Center Business Edition" on page 68
  - "Mitel Contact Center Enterprise Edition" on page 69
  - "Commander Contact Centre" on page 69
- "Applications for Informal Contact Centers" on page 70
  - "Mitel Customer Service Manager" on page 70

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. MCD's integrated ACD functionality is enhanced by the Mitel Contact Center Solutions product suite. Designed for formal, small-to-enterprise sized contact centers, Contact Center Solutions enables customers to streamline operations and improve efficiency. It is described in more detail below, and 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 can log in and log out of any phone with their unique Hot Desk ID and password. The system applies the agent's personal profile to the set. After they log in, agents can make themselves present in, or absent from, any one of their groups through a pre-programmed feature access key (FAK). If agents are not present in any of their groups, they can still access their prime line keys. They can log in and log out using feature access codes (FACs), with flexibility in the choice of sets. These capabilities are all controlled by Class of Service (COS) settings in MCD, ensuring that ACD administrators have full control over permission assignment.

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

With dynamic license allocation, customers can purchase the number of concurrent licenses required for their operation.

For a description of the 3300's Hot Desking capabilities see "Hot Desking" on page 17

Applications for Formal Contact Centers

A formal contact center is typically an organization that operates one or more call centers that are critical to their business. Formal contact centers can be large or small. They 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 formal contact center solutions provide a formal way of dealing with incoming calls and support a range of basic to advanced functionality and price points. They are built on the sophisticated call routing functionality of the ACDII application in MCD.

Mitel's formal contact center solution supports dynamic Extensions for agents, extending ACD features to all IP, SIP, and external devices, and enabling External Hot Desk Agents (EHDAs) to be on 3rd party endpoints, such as cell phones, on analog phones, or at home. The solution provides ACD dimensioning for active agents, agent skill groups, dial out of queue points, and RADs.

Support for Mitel Contact Center Solutions in Virtualized Environments

Mitel's formal contact center solutions take advantage of the improved performance, ease of use, and comprehensive management capabilities of data center virtualization. These solutions are fully supported in virtualized (VMware) environments, where virtual appliances are created to simplify configuration and installation. All of the advanced contact center and general business functionality is supported, with the exception of web callbacks and Music on Hold (due to their requirement for a sound card), and Multimedia Contact Center (coming soon).

Mitel's formal contact center solutions include

- "Mitel Contact Center Business Edition" on page 68
- "Mitel Contact Center Enterprise Edition" on page 69
- "Commander Contact Centre" on page 69

Mitel Contact Center Business Edition

Mitel Contact Center Business Edition is designed for organizations that need to process incoming calls in a formal way, and require advanced capabilities with minimal customization. Contact Center Business Edition is designed for single-site contact centers that have 25 or fewer agents and focused application needs. Contact Center Business Edition provides

- An award-winning graphical agent desktop
- A Core set of historical and real-time reports
- Consolidated agent and queue management
- Rich voice automatic call distribution (ACD) functionality

- Contact center management tools
- Contact center schedueling for automatic agent schedueling based on business rules and required skills
- Schedule adherence to verify agents are adhering to their schedules
- Real-time agent and queue control
- Call accounting
- Automatic call distribution
- A browser-based IVR solution that provides advanced call routing and self service
- Inbound multimedia: ACD for e-mail, web chat, fax, SMS, and walk-in
- Desktop phone and softphones
- CRM screen-pops
- 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 Contact Center Enterprise Edition is a scalable, resilient, and virtual solution for 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, 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. 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.

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 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 look up customer data or information about a call (who the caller is, the caller ID, and the 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 among 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. This 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 "Third-Party Developer Support" on page 72). 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:** provide tools to improve team productivity including call management and 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. With Call Accounting, you can

- 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 you to charge back departments, employees, and customers using markup or discount pricing
- Mitel Traffic Analysis (optional module): enables you to determine if the organization is using its incoming, outgoing, and bi-directional trunks efficiently

General Business Solutions

Emergency Response Adviser

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 dispatch emergency personnel (for fires, medical emergencies, etcetera).

Emergency Response Adviser enables you to alert mobile personnel on their phones or pagers. Emergency Response Adviser can simplify the generation of data files necessary for keeping the PSAP up to date with physical plant changes, which is an essential part of emergency services management.

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 with 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:** enables switch-to-application server communication for multiple switches
- **MiAUDIO:** enables an application to process voice on multiple Mitel phones
- **MiTAI Browser Tool:** ensures the connection is functioning properly to make function calls and to view events from the API
- **MiTAI Server Logger Tool:** connects to the MCD host platform and downloads log files, and captures all MiTAI server incoming and outgoing messages for debugging purposes
- **MiTAI Client Logger Tool:** enables you to access MiTAI application information, collect MiTAI API information in a log file, and capture MiTAI client data on incoming and outgoing messages for debugging
- **MiAUDIO Test Tool:** enables you 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 that require 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 developers to produce applications (for example, voice mail or automated call routing systems requiring DTMF detection), that require sophisticated capabilities beyond standard call handling.

MiAUDIO

With MiAUDIO, developers can include the processing of phone 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 and handle multiple phones, trunk devices, and routing queues. Applications written for MiAUDIO permit third-party call control (outside of the "conversation"). MiAUDIO targets server applications that control multiple devices and handle things such 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 enables 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 part of the MBG software blade. 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 Mitel 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. Mitel HTML Toolkit provides Application Programming Interfaces (APIs) for developers to customize these large-screen display phones.

HTML Toolkit enables developers to build graphical applications for 5320, 5330, 5340, and 5360 phones using standard web authoring tools. They can tightly integrate the phones into their business processes and deliver tailored functionality for a wide range of business applications that target horizontal and vertical market sectors. Custom applications provide simple navigation and enhanced usability of display phones and meet organizational objectives (for example, sales, branding, and process improvement).

The HTML Toolkit also provides notification applications for Mitel 5304, 5312 and 5324 IP phones. HTML Applications are also supported over MBG (Teleworker). With HTML Toolkit 2.2, core applications such as Mitel Intelligent Directory and Mitel Live Content Suite will run on supported phones anywhere using MBG.

Examples of applications that can be developed with the HTML Toolkit include

- **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

Administration Tools

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

Enterprise Manager

Enterprise Manager is a management tool that 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.
- Management of a network of MCD systems via OPS Manager, the network management tool (OPS Manager is integrated into Enterprise Manager).
- 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, NuPoint, 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*.

MCD Software Installer

MCD Software Installer is a Windows-based, standalone software tool that enables technicians to automatically upgrade software on one MCD system or on multiple MCD systems simultaneously. MCD Software Installer provides a single interface that allows technicians to define which installation steps to perform during the process. MCD Software Installer re-synchronizes with the AMC during software upgrades.

MCD Software Installer 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, 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.

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	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)	Included	Included	Included	Included	Included	Included
5300 Series Handset Version 4	Yes	Yes	Yes	Yes	Yes (Wideband)	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.722.1**	G.711, G.729a, G.722.1**	G.711, G.729a, G.722.1**
Voice QoS (802.1p/q)	Yes	Yes	Yes	Yes	Yes	Yes	Yes
Encryption	128 bit AES*	128 bit AES*	128 bit AES*	128 bit AES*	128 bit AES*	128 bit AES*	128 bit AES*
802.1x Support	Yes	Yes	Yes	Yes	Yes	Yes	Yes
CLASS B Support	Yes	Yes	Yes	Yes	Yes	Yes	Yes
Headset Jack	No	Yes	Yes	Yes	Yes	Yes	Yes
Peripherals (Modules) Support	No	No	Yes	No	Yes	Yes	Yes (Does not include PKM support)
IP DECT Stand/GigE Stand Support	No	Yes	Yes	Yes	Yes	Yes	Yes***

Page 1 of 5

*Advanced Encryption Standard

** Audio coding: ITU-T Rec. G.722.1 Annex C, licensed from Polycom®. Applies to 5330, 5340 and 5360 IP Phones only

*** 5360 IP Phone supports embedded Gigabit

Powering Options	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)	2.03W	2.43W	2.43W	3.20W	3.2W	3.2W	4.2W*
Power Consumption (Typical)	2.88W	3.23W	3.23W	4.3W	4.8W	4.8W	7.4W*
Power Consumption (Maximum)	3.45W	3.87W	3.87W	5.3W	5.8W	5.80W	7.8W*

* 10/100 MB Mode values. GB Mode values: Idle - 4.8W; Typical - 8.6W; Maximum - 9.2W

Display	5304	5312	5324	5320	6330	5340	5360
Color	No	No	No	No		No	
Size (pixels)	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)	320 x 240 (1/4 VGA)	800 x 480 (7in.)
Number of Pixels (w x h)	160 x 28	160 x 28	160 x 28	160 x 320	160 x 320	160 x 320	480 x800
Pixel Size	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	.19 x .19 mm
Illumination	Reflective Backlit White	Reflective Backlit White	Reflective Backlit White	Reflective Non-backlit	Transmissive FSTN with White LED Backlight	Transmissive FSTN with White LED Backlight	TFT Color with LED Backlight
Contrast Adjust	Yes	Yes	Yes	Yes	Yes	Yes	No
Display (soft) Keys	No	No	Yes	Yes	Yes	Yes	Yes
Auto Dimming	Yes	Yes	Yes	N/A	Yes (Programmable)	Yes (Programmable)	Yes (Programmable)

Page 2 of 5

Backlight Off Capability	No	No	No	N/A	No (Screen saver)	(Screen saver)	
Chinese Character Support	Yes	Yes	Yes	Yes	Yes	Yes	Yes
Function Keys	**5304**	**5312**	**5324**	**5320**	**5330**	**5340**	**5360**
Number of Programmable Feature/Line Appearance Keys	9	12	24	8 (Self-labelling)	24 (Self-labelling)	48 (Self-labelling)	48 (Self-labelling)
Fixed Feature Keys	2	10	10	10	10	10	10 plus 9 quick launch icons on Gadget Sidebar
Softkeys	0	0	3	3	3	6	6
Multi-line	Yes	Yes	Yes	Yes	Yes	Yes	Yes
Hold	No (Definable)	Yes	Yes	Yes	Yes	Yes	Yes
Redial	No (Definable)	Yes	Yes	Yes	Yes	Yes	Yes
Cancel	No (Definable)	Yes	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)	Yes	Yes	Yes	Yes	Yes	Yes
Transfer/ Conference Key	No (Definable)	Yes	Yes	Yes	Yes	Yes	Yes
Call Forward (On/Off) Key	No (Definable)	Yes (Definable)	Yes (Definable)	Yes (Softkey)	Yes (Softkey)	Yes (Softkey)	Yes (Softkey)
Call Me Back Key	No (Definable)	Yes (Definable)	Yes (Definable)	Yes (Softkey)	Yes (Softkey)	Yes (Softkey)	Yes (Softkey)
Phonebook/ Directory Key	No (Definable)	Yes (Definable)	Yes (Definable)	Yes (Via Menu Key)	Yes (Via Menu Key)	Yes (Via Menu Key)	Yes (Via Menu Key)
Microphone Key	No	No	No	No	No	No	No
Mute Key	No	Yes	Yes	Yes	Yes	Yes	Yes
Speakerphone	No	Yes	Yes	Yes	Yes	Yes	Yes

Page 3 of 5

Program/ Superkey	No (Definable)	Yes	Yes	Yes (Via Menu Key)	Yes (Via Menu Key)	Yes (Via Menu Key)	Yes (Via Menu Key)
Desktop User Tool	Yes	Yes	Yes	Yes	Yes	Yes	Yes
Indicators	**5304**	**5312**	**5324**	**5320**	**5330**	**5340**	**5360**
Feature/Line Appearance LEDs	2	12	24	8	24	48	48
Message Waiting LED	Orange	Orange	Orange	Orange	Orange	Orange	Orange
Hold	N/A	Yes (Flashes Orange)	Yes (Flashes Orange)	Yes (Flashes Orange)	Yes (Flashes Orange)	Yes (Flashes Orange)	Yes (Flashes Orange)
Hold Button	N/A	Red	Red	Red	Red	Red	Red
Line LED Color	Orange	Orange/Green	Orange/Green	Orange/Green	Orange/Green	Orange/Green	Orange/Green
Ringer LED	Orange	Orange	Orange	Orange	Orange	Orange	Orange
Microphone/ Mute LED	No	Orange	Orange	Orange	Orange	Orange	Orange
Acoustic Functions	**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	Yes	Yes	Yes	Yes	Yes	Yes
Handsfree: Half Duplex Full Duplex	None	Full Duplex	Full Duplex	Full Duplex	Full Duplex	Full Duplex	Full Duplex
Wideband Audio Hardware	No	No	No	No	Yes	Yes	Yes
On-Hook Dialing	No	Yes	Yes	Yes	Yes	Yes	Yes
On-Hook Call Announce (Paging Receive Capability)	Yes	Yes	Yes	Yes	Yes	Yes	Yes
Off-Hook Call Announce	No	Yes	Yes	Yes	Yes	Yes	Yes
							Page 4 of 5

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	Yes	Yes	Yes	Yes	Yes	Yes
System Software Requirements	**5304**	**5312**	**5324**	**5320**	**5330**	**5340**	**5360**
3300 ICP	Release 9.0 (UR1) or later	Release 9.0 (UR1) or later	Release 9.0 (UR1) or later	MCD Release 4.1 (SP1) or later	5330: Release 8.0 (UR3) or later	Release 9.0 (UR1) or later	MCD Release 4.1 (SP1) or later
Mitel Border Gateway telworker service	Release 5.2 (SP1) or later	Release 5.0 or later	Release 5.0 or later	5320: Release 5.2 (SP1) or later	5320: Release 5.2 (SP1) or later	Release 4.1 or later	Release 5.2 or later
SIP Software Platform	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 5 of 5

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 following phones to be located off-site and still be supported by MCD:

- **5304 IP Phone:** an entry-level display phone that 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 28: Display Phones

Desktop Application Phones

Mitel desktop application phones provide innovative features and applications. Users can access features quickly using programmable self-labeling keys. A large graphics display and an intuitive softkey interface provide easy-to-use applications such as PhoneBook and Call History. The following phones are ideal for enterprise executives, managers, ACD agents, ACD supervisors, and teleworkers:

- **5320 IP Phone:** provides 8 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 and six call state sensitive intuitive softkeys on a large color touch-sensitive display.

These phones provide a built-in HTML toolkit for desktop applications development (See "HTML Toolkit for Mitel 5320, 5330, 5340, and 5360 IP Phones" on page 73 for additional details). The following Mitel phone applications are currently available:

- **Mitel Intelligent Directory Application:** provides access to Active Directory and Outlook contacts on the 5320, 5330, 5340 and 5360 IP Phones. It also provides presence information on 5320, 5330, 5340 and 5360 IP Phones for contacts via Microsoft Office Live Communications Server.
- **Mitel Live Content Suite:** enables customers to create and publish dynamic and personalized information to users, transforming 5320, 5330, 5340 and 5360 IP Phones into media information appliances.

5320 IP Phone

5330 IP Phone

5340 IP Phone

5360 IP Phone

Figure 29: Desktop Application Phones

Mitel UC360™ Collaboration Point

Mitel® UC360 Collaboration Point is an all-in-one multimedia collaboration appliance that provides multi-party audio and video conferencing, in-room presentation display, and remote collaboration for the personal office meeting space. Featuring a compact design and an easy-to-use touchscreen interface, the UC360 makes collaboration a natural part of every work day.

Figure 30: Mitel UC360 Collaboration Point

Key features of the UC360 Collaboration Point include

- Superior audio conferencing capability including a beam forming microphone array
- Built-in presentation display capability via HDMI interface that supports connection to high definition LCD display/projector
- Built-in MS Office readers and editors
- Remote desktop access (no need to bring laptop to give a presentation)
- Support for multiple file transfer methods, including USB Flash Drive, SD Card,

- Dropbox™ and Google® Docs
- Audio conferences for up to four parties
- High Definition video conferencing for up to four parties with an integrated conference bridge
- Support for integration with Active Directory and LDAP
- Ability to display Remote Presentations

Mitel 5505 Guest IP Phone

The Mitel 5505 Guest IP Phone meets the needs of Hospitality customers who are looking to deploy IP to guest rooms. The 5505 Guest IP Phone base provides the physical features hotel guests have come to expect such as, a high quality full-duplex speakerphone, programmable speed dial keys, a large area for custom branding and dialing instructions, a cordless handset locator, and a physical ringer volume switch.

The 5505 Cordless Handset provides industry leading features by virtue of its built-in two line backlit display and a built-in alarm clock that can be easily set. With an operating range of up to 50 meters (150 feet) from the phone base, the 5505 Cordless Handset is ideal as a second phone for a guest room or a suite of rooms.

The 5505 Guest IP Phone provides DECT to SIP gateway functionality and the 5505 Cordless Handset.

DECT to SIP gateway features include/support

- A built-in DECT/DECT 6.0 interface with support for 1 base cordless handset plus up to 3 extension handsets (appears as one extension to the PBX)
- Single line SIP VoIP protocol features
- A High Speed Internet / PC Port side mounted for easy guest access
- Powering via Power Over Ethernet (802.3af)
- A full duplex speakerphone with On-Hook Dialing support
- A 12-button dial pad
- 6 fixed feature keys: Volume up/down rocker, speakerphone, microphone mute, messages, end call, handset locator
- 5 programmable speed dial keys
- Joining of a handset call with a base unit speakerphone call
- MCD Resiliency support: can still make calls upon failover to a secondary controller

5505 Cordless Handset features include

- A 2-line illuminated display with automatic dimming
- A 12 button dial pad
- 9 fixed keys: Talk, Hang-up / Power, Messages, Volume up/down, Speakerphone, Soft Key 1 & 2, Mute, Flash

- Message waiting indication via phone display
- A Built-in Speakerphone
- Support for multiple languages
- Guest programmable options: Alarm clock, phonebook, customizable ringer volume, choice of ringer melodies, and language selection

Figure 31: Mitel 5505 Guest IP Phone

Mitel 5560 IPT

The 5560 IPT is a dual display / dual handset, multi-line trading appliance. It's rugged design is suited to the high activity environment of stock trading floors: it provides access to many lines and handles a high volume of calls. The 5560 IPT combines the speed and performance that split-second trading demands, at a fraction of the total cost of ownership of other solutions.

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

The 5560 IPT enables 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 32: 5560 IPT

Wireless IP Phones

The 3300 ICP supports the following wireless phones:

- **SpectraLink 8020, 8030 Wireless Handsets:** featuring a variety of functions, the SIP-enabled 8020/8030 phone is an ideal choice for vertical enterprises, such as Healthcare or Manufacturing, that run enterprise-grade 802.11b, g or wireless networks, that require durable handsets with basic call control capabilities. The 8030 supports push-to-talk functionality for broadcast communication between employees, eliminating the need for two-way radios or walkie talkies. The 8020/30 handsets can be deployed in 802.11a networks and support the display of graphical information such as waveforms. With the introduction of R3.0 firmware, these handsets can be deployed without the SVP server, or can continue to be deployed with the SVP server for deployment behind legacy WLAN networks.
- **5603, 5604, 5606 Wireless Handsets:** IP DECT phones for the IP-DECT Wireless System (Global). These handsets provide voice communication, text messaging, alarm handling, and an extensive set of 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.

SpectraLink 8020

SpectraLink 8030

Mitel 5610

Mitel 5603

Mitel 5604

Mitel 5606

Figure 33: Wireless Phones

IP Phone Accessories

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

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

* Supports embedded Gigabit

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

Mitel IP Programmable Key Modules 12 and 48

The 12- and 48-button Mitel Programmable Key Modules (PKMs) extend the capabilities of the Mitel 5324, 5330, and 5340 IP Phones with additional buttons and LED indicators. With these expansion modules, you can readily add 12 or 48 or up to 96 buttons to the existing programmable keys on the IP Phones.

You can program the 12, 48, or 96 additional personal keys as feature keys, speedcall keys, direct station select (DSS) keys, or line appearance keys. Each key has a line status indicator that works the same way as those of the associated phone. The additional keys can be readily programmed using the phone or by the system administrator.

12-Button PKM **48-Button PKM**

Figure 34: 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 need for an additional LAN port.

The conference unit provides

- Acoustic beam-forming technology that controls near end, far end, and double talk, and locates the 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 35: 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 System Administrator sets the operation mode during system programming. The Line Interface Module has the following modes of operation:

- **LIM Mode:** is recommended for teleworker/remote configurations and 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.

- **Failover Mode:** 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).

DECT Cordless Handset and Headset

The DECT Cordless Handset and DECT Cordless Headset offer corridor mobility for Mitel 5330, 5340, and 5360 IP phone users. The Cordless Handset and Headset enables users to move freely within the office or adjacent offices (up to 300 feet from their desk) while still communicating from their desk phones.

Both cordless devices connect to an IP phone 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.

The DECT cordless accessories provide

- LED Indicators on the Cordless Module, Handset and Headset that indicate connectivity and charging status
- Eight hours of talk time
- 43 Hours of standby time
- An operating range of up to 300 feet (100 metres) in a typical office environment
- An 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 36: Cordless Handset and Headset

Mitel Bluetooth Module

The Mitel Bluetooth Module is a new IP Peripheral which fits discretely into the back of the 5330, 5340 and 5360 IP Phones. The Bluetooth Module supports Mitel's Bluetooth Handset and a vast number of third-party Bluetooth headsets from other manufacturers. The Bluetooth Module enables Mitel IP Phone users to integrate their commercially available Bluetooth headsets with their desk phones: they can enjoy handsfree freedom, similar to using their cell phones. The ability to use a single headset with both a desk phone and cell phone augments Mitel's mobility solution for users who want to leverage the Dynamic Extension capability. The Bluetooth Module enables users to have personal area mobility with a potential range of up to 30 feet from their desks within the office or adjacent offices, while still communicating using their desk phones.

The Mitel Bluetooth Module provides

- An initiate call / end call key
- Volume control keys and a Mute key
- A built-in ringer in the Bluetooth Handset
- The ability to pair up to 6 Bluetooth devices with the Bluetooth Module
- The ability to place outbound calls when mobile with programmable “auto speed dial” upon off hook - Speak@Ease or “0” or secretary speed call
- LED indicators on the Bluetooth Module and Bluetooth Handset that indicate connectivity and charging status
- Eight hours of talk time
- Forty-three hours of standby time
- A battery recharge time of three hours or less
- An operating range of up to 30 feet (10 Meters) from the IP Phone
- An out of communications range warning tone

Figure 37: BlueTooth Module and Handset

Mitel Gigabit Ethernet Stand

The Gigabit Ethernet (GigE) Stand enables 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

Mitel IP DECT Stand

The IP DECT Stand is an accessory peripheral for the Mitel 5300 Series IP Phones. The IP DECT Stand connects to the base of a Mitel 5312 / 5324 / 5320 / 5330 / 5340 or 5360 IP Phone and acts as an IP DECT base station with SIP Gateway functionality. The IP DECT Stand connects to the network through the PC port on the Mitel IP Phone. It supports up to eight 5610 DECT Handsets that act as SIP extensions. The IP DECT Stand

- Supports up to eight handsets / three simultaneous calls
- Is supported across a range of IP phones: 5312 / 5324 / 5320 / 5330 / 5340 / 5360 IP Phones
- Is configured using a web configuration interface
- Has a DECT-based design: DECT 6.0 cordless technology provides higher quality voice transmission, density, and less interference

5610 IP Dect Stand

Gigabit Ethernet Stand

Figure 38: 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 39: 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) that includes screen-based call status and call handling prompts. A telephone keypad and dual handset-headset jack provide fast, efficient attendant call handling.

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

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

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Figure 40: 5550 IP Console

Mitel 5540 IP Console

The Mitel® 5540 IP Console is the ideal attendant solution for small and medium sized businesses. It can be used as an attendant console, a sub-attendant position for departments or workgroups, or as a back-up answering position. It supports a broad range of standard and specialty functions and features including

- A highly visible, four-line, 80-character, backlit, tilt display that shows the date, time, call status information, calling line identification, and calls waiting
- 14 fixed function keys dedicated to basic and enhanced call-handling activities
- 10 softkeys that control access to the attendant features through call state sensitive keys
- Teleworker support with Mitel Border Gateway that enables attendants to work anywhere, anytime
- Third-party cordless headset integrated functions: Call Answer, Call Cancel, Audio Controls, and training mode support
- Access to integrated Mitel hospitality features including room status, guest telephony privileges, and automatic wake-up calls

Multiple language support for global customers: English, French, Spanish, German, Italian, Dutch and Portuguese.

Figure 41: 5540 IP Console

Features

Features of Mitel Communications Director

The following table details MCD features and indicates which features are supported by Resiliency. N/A indicates that a feature is not specifically related to resiliency or a resilient device, but that it 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 phone.	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
		Page 1 of 28

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 External Hot Desk Agents	Mitel supports Dynamic Extensions for agents, extending ACD features to all IP, SIP, and external devices, and enabling External Hot Desk Agents (EHDAs) to be on 3rd party endpoints, such as cell phones, on analog phones, or at home. An EHDA is an External Hot Desk User (EHDU) that is also a member of an ACD group. In a typical work-at-home scenario, the user answers the ACD calls on a single-line residential phone and has a MiTAI-based call center application that provides "screen pops" that contain caller information and client account data.	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 Scaling	Provides increased ACD dimensioning for active agents, agent skill groups, dial out of queue points, and RADs	
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

Page 2 of 28

Feature Name:	Description:	Support while the set is on the secondary controller
ACD Silent Monitor	Allows a supervisor to listen to an agent's phone 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 phone 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
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
		Page 3 of 28

Feature Name:	Description:	Support while the set is on the secondary controller
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
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
		Page 4 of 28

Feature Name:	Description:	Support while the set is on the secondary controller
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
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 phone directory and speed dial list. You use it to save phone 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
		Page 5 of 28

Feature Name:	Description:	Support while the set is on the secondary controller
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	Uploads audio files to the MCD system and uses them for embedded Music on Hold, all Auto Attendant greetings, set greetings, and RAD greetings Uploads 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
Audit Trail	Provides a historical record of changes made to the system (from the System Administration Tool and various other user interfaces and applications) in the Login/ Logout Audit Logs form Assists with troubleshooting problems that arise, pinpointing who, in a multi-administrator system, is responsible for a particular change	N/A
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 Mobile Failover/ (EHDU)	If your desktop phone fails, the Mobile Failover/External Hotdesk User (EHDU) feature reroutes all calls to your mobile device. After the phone returns to service, calls are automatically routed back to the desktop.	
Automatic Phone Lock	The ability to schedule an event to automatically log out Hotdesk users, who are current logged in	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
Backups - Scheduled	Enables you to schedule events to automate the process of backing up the system database to the local hard drive or to an FTP server	N/A
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

Feature Name:	Description:	Support while the set is on the secondary controller
Basic Rate Interface	A basic ISDN service that consists of two 64Kbps channels and one 16Kbps channel. Basic Rate Interface (BRI) is supported on the 3300 ICP by the Quad BRI module.	N/A
Broadcast Groups	See Groups-Key System and Multicall.	Yes
Broker's Call	Allows you to temporarily suspend a phone 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.
CSV File Import/Export - Scheduling	Enables you to schedule events to automate the process of importing and exporting form data in .CSV format	N/A
Calculator	Allows you to use your phone as a basic four-function calculator by using the phone 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
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

Feature Name:	Description:	Support while the set is on the secondary controller
Call Duration Display	Displays the call duration for incoming and outgoing calls, in one minute increments (starting at 0:00)	Yes
Call Forking	Enables a call to be split or forked so that several locations can ring simultaneously. MCD supports forking for outgoing calls over SIP trunks supported for outgoing SIP calls handled by external SIP forking servers	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 phone 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 phone 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
Calling Line Identification	The phone 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

Feature Name:	Description:	Support while the set is on the secondary controller
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
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 phone on the PBX to use CENTREX features.	Yes

Page 9 of 28

Feature Name:	Description:	Support while the set is on the secondary controller
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.
Voice Compression	Allows IP calls in VoIP systems to use less bandwidth than uncompressed calls In addition to the G.711 a/u law and G.729a codecs already supported, Mitel 5330, 5340, and 5360 IP Phones now support the G.722.1 wideband codec.	Yes
Centralized Suites for Analog Devices	Distributes the connections for analog guest room extensions across several elements and centralizes all processing on a single IP node in standalone hospitality environments This implementation can be protected by installing the ICP Hospitality node software on fully redundant platform such as the Stratus® Server for RHEL.	No
Conference	Allows you to connect three or more calls into a single phone 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

Feature Name:	Description:	Support while the set is on the secondary controller
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
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 phone credit limit. The PBX uses an Alert message to notify the PMS when the established phone 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 phones, 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

Feature Name:	Description:	Support while the set is on the secondary controller
Direct Inward Dialing (DID)	Permits incoming calls on designated trunks to directly access predefined stations (or other answering points) on the system	Yes
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 phone 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 phone 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 Transfer to Voice Mail	Transfers an active call directly to the requested party's voice mailbox instead of waiting for the system to transfer it there after ringing the party's phone. Use this feature when you know that the party is unavailable or when the caller only wishes to leave them a voice message.	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 phones 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

Feature Name:	Description:	Support while the set is on the secondary controller
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
Dual PKM 48 Support	The Programmable Key Module 48 (PKM48) provides 48 additional feature keys for phones. 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
File Transfer Support	You can use the Scheduler application to collect and transfer the following file types: • SMDR Records • Audit Trail Logs • phone Directory • Traffic Logs • IDS Synchronization Files	Yes
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

Feature Name:	Description:	Support while the set is on the secondary controller
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
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 phones 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 phone without lifting the handset	Yes
Headset Operation	Allows you to use a Headset to make and receive phone calls	Yes
Hold	Allows you to temporarily suspend a phone call. While the call is on hold, you can use the other phone features. The call can be retrieved either at the original answer point or at another extension.	Yes

Feature Name:	Description:	Support while the set is on the secondary controller
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
Hot Desking -External	Allows users to configure any external phone 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 phone 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
Hotdesk Login Indicator	The Busy Lamp Field (BLF) indicator light does not flash when a hot desk user is logged out. When a hot desk user is logged in, the lamp displays a steady, green light. The Green BLF Lamp for Logged in Hotdesk User Class of Service option controls this capability.	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
		Page 15 of 28

Feature Name:	Description:	Support while the set is on the secondary controller
Integrated Directory Service	The Integrated Directory Service (IDS) feature uses the Lightweight Directory Access Protocol (LDAP) to synchronize user and service data from your corporate directory server to the MCD platform. System Data Synchronization is then used to share the data among the administrative group. Note that in MCD Release 5.0, data is synchronized in one direction only, from the corporate directory server to MCD, and that only one type of directory server is supported: Microsoft Active Directory. Although all users are IDS manageable by default, you can disable the feature for individual users on the User and Device Configuration form.	N/A
Integrated Directory Service - Scheduling	After you have programmed IDS for your cluster or network, you can schedule "full" or "incremental data synchronization events. Full IDS synchronization queries the directory server for new, modified, and deleted user records. Incremental IDS synchronization queries the directory server for new and modified user records.	
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 phones in a hotel environment that must be restricted from dialing various internal numbers	Yes
Inward Dialing Modification	Enables you to alter dial strings contained in inbound SIP calls After adding substitution rules, you can apply them to both "Called Party" and "Calling Party" SIP headers. You can implement this feature as part of the initial system setup when you Program SIP Trunks and Program SIP Phones.	
IP Networking	Enables 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 phone 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

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 phone 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) • Arabic (5312 and 5324 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
Location Based Accounting	Location Based Accounting enables you to automatically determine a device's location based on its IP address. You can attribute calls to specific locations and bill the locations accordingly. The feature involves two components: • zone identification based on IP address • device location information in the 3300 ICP's SMDR records	Yes
Location Based Call Routing	Directs calls made to designated numbers (such as Emergency - 911, Directory Assistance - 411, etc) to appropriate services located in the same zone as the device from which the users are dialing.	N/A

Feature Name:	Description:	Support while the set is on the secondary controller
Location Based Time Zone	Enables 3300 ICP administrators to manage set displays based on the time zone in which the sets are located, independent from the system time zone	
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
Messaging-Advisory	Displays a short advisory message to display-set users who call your phone	No
Messaging-Callback	Allows you to leave a callback message on a phone 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 phone. 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 phones 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
		Page 18 of 28

Feature Name:	Description:	Support while the set is on the secondary controller
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
Multi-Color LED line status	Involves employing color to indicate to a set user whether activity on a line key is of direct concern to the user, or of greater concern to another member of the associated broadcast group	Yes
Multi-device Suite Licence	Simplifies and cost reduces the hospitality solution where hotels require multiple devices in a single suite. Hoteliers can licence hotel rooms as single suites: up to 6 devices can be configured in a suite while consuming only a single System licence.	N/A
Music	Allows you to listen to the Music On Hold music source through the speaker on the phone	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
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 phone 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	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

Feature Name:	Description:	Support while the set is on the secondary controller
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	Switches the system from day service to night service and vice versa 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 Indicator	Enables supported sets to be programmed so the Feature Access Key (FAK) LED goes off during the day and turns on at night. Pressing the key displays the current mode of operation (Day, Night 1 or Night 2).	Yes
Night Service - Scheduled	Enables you to schedule Night Service modes on the MCD system. Allows transitions between all of 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
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

Page 20 of 28

Feature Name:	Description:	Support while the set is on the secondary controller
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
Prevent Call to SIP Devices if in Use	Forwards incoming calls to an alternate destination, such as voice mail, If the SIP Phone user is already engaged in a call	Yes
Post Call Destination	Automatically forwards callers to a specified destination after the called party hangs up.	N/A
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
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
		Page 21 of 28

Feature Name:	Description:	Support while the set is on the secondary controller
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 phones 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
Recorded Announcement Device Support	RADs 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
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

Feature Name:	Description:	Support while the set is on the secondary controller
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 phone 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, or through OPS Manager.	Yes
Ringer Control	Allows you to adjust the volume and pitch of the phone ringer	Yes
Ring Groups	Provides the ability to ring all members of a group simultaneously or sequentially	Yes
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.	N/A
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

Feature Name:	Description:	Support while the set is on the secondary controller
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 phone 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
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	Yes
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

Feature Name:	Description:	Support while the set is on the secondary controller
Station-To-Station Dialing	Allows you to dial any other station directly	Yes
Suite Service	Allows you to group a number of phone lines through interconnected hotel/motel rooms, or suites, for the purposes of billing and sharing phone 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 phone 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
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
phone Directory -Privacy Option	Any extension number in the system phone directory can be designated as private. When an extension number is private, the number is not displayed on other users' phones.	Yes
phone 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
		Page 25 of 28

Feature Name:	Description:	Support while the set is on the secondary controller
Templates	Templates have been introduced to speed the configuration process and ensure that correct settings are applied throughout the enterprise. Common settings, such as the Class of Service and Device Type, can be saved in a template and applied to multiple users and devices. Three new forms are available: Key Templates: Enables you to program line key settings for multiline phones and SIP devices User and Device Templates: Enables you to program a subset of the information normally added on the User and Device Configuration form User Roles: Enables you to link templates with roles. When you add a new user, you are prompted to select a role and its associated template.	N/A
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 phone 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 phone is required.	Yes

Feature Name:	Description:	Support while the set is on the secondary controller
Trunk Answer From Any Station (TAFAS)	Allows you to answer any call that rings a night bell	Yes
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
User Provisioning Roles	The User and Device Configuration form simplifies the creation and management of users, enabling you to modify a wide range of user data without having to make modifications in many separate forms When configuring users, you can apply a default (standard) user role or a unique, customized user role.	N/A
Voice Mail	The system has its own integral voice mail system.	Yes
		Page 27 of 28

Feature Name:	Description:	Support while the set is on the secondary controller
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 phone.	Yes
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
		Page 28 of 28

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.
	Page 2 of 4

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 phones 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.
Mailbox Administration via OPS Manager	Mailbox administration (adds, moves, changes) can be performed using OPS Manager, a standalone application that works seamlessly with the MCD embedded system management.
Networked Voice Mail	Networked Voice Mail allows voice mail users to seamlessly send and receive messages between all the voice mail servers on a network. This includes (but is not limited to): • selecting destination mailboxes using the corporate voice mail directory. • confirmation of destination mailboxes (name or number). • using existing voice mail features such as receipts, distribution lists, replying to a voice mail Networked Voice Mail supports EMEM (networked and clustered), NuPoint Messenger, and other VPIM2-compliant mail servers (G.711 compliant), and is compatible with Hot Desking.
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.
	Page 3 of 4

Feature	Description
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.
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 phone 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 phone 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.
	Page 4 of 4

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 phone display screen if the appropriate Class of Service options are set.

Feature	Description
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

Standard	Feature	Mitel 3300
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.

Feature	Description
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 phone 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 phone 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 phone 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.

Feature	Description
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 phone call. While the call is on hold, the person that placed the call is able to use other phone features. The call can be retrieved from the phone that placed the call or from another phone.
Direct Page	Allows you to page another phone 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.

For secure signaling to SIP devices, Mitel supports Transport Layer Security (TLS) protocol - an upgrade to the SSL protocol. TLS provides message encryption, message integrity, and endpoint authentication. It is used for secure SIP signaling between MCDs and the following endpoints:

- Mitel-branded TLS-capable SIP devices, such as 5603, 5604, 5607
- 3rd-party SIP devices as approved by Mitel's SIP Center of Excellence with TLS interworking

When a TLS set initiates a connection with an MCD, the TLS's Server (unilateral) authentication method is used to authenticate the server (only the server's security certificate is required).

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 phone privileges and restrictions to devices. In addition, public phones 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 Application Suite	Y	Y
3300 ICP Components	Y	Y
3300 Voice Mail	Y	Y
Audio Web Conferencing	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
Visual Workflow Manager (Intelligent Queue)	Y	Y
Call Recording	Y	Y
Unified Communicator	Y	Y
Intelligent Directory Application	Y	Y
Live Content Suite	Y	Y
		Page 1 of 2

North American Region (continued)		
	Canada	**United States**
Phones		
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
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
Gigabit Ethernet Stand	Y	Y
Cordless Module and Accessories	Y	Y
5610 DECT Handset and IP DECT Stand	Y	Y
Bluetooth Module	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
3300 ICP Components	Y	Y	Y
3300 Voice Mail	Y	Y	Y
			Page 1 of 3

Asia Pacific Region (continued)			
	Australia	**New Zealand**	**China**
3300 Wireless	Y	Y	N
Peripheral Cabinet	Y	Y	Y
SX-200 Bays	N	N	N
Applications			
Contact Center Management	Y	Y	N
Interactive Contact Center	Y	Y	N
Contact Center Scheduling	Y	Y	N
Multimedia Contact Center	Y	Y	N
Visual Workflow Manager	Y	Y	N
Call Recording	Y	Y	N
Unified Communicator	Y	Y	Y**
Intelligent Directory Application	Y	Y	Y
Live Content Suite	Y	Y	Y
* **Note:** The Unified Communicator Advanced Collaboration Option is not translated into Chinese.			
			Page 2 of 3

Asia Pacific Region (continued)			
	Australia	**New Zealand**	**China**
Phones			
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*
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
Gigabit Ethernet Stand	Y	Y	Y
Cordless Module and Accessories	Y	Y	N
5610 DECT Handset and IP DECT Stand	Y	Y	N
Bluetooth Module	Y	Y	Y
			Page 3 of 3

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
									Page 1 of 3

EMEA Region (continued)									
	UK	**Spain**	**Portugal**	**Netherlands**	**Italy**	**Germany**	**France**	**UAE**	**South Africa**
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
Visual Workflow Manager	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
Intelligent Directory Application	Y	Y	Y	Y	Y	Y	Y	Y	Y
Live Content Suite	Y	Y	Y	Y	Y	Y	Y	Y	Y
									Page 2 of 3

EMEA Region (continued)									
	UK	**Spain**	**Portugal**	**Netherlands**	**Italy**	**Germany**	**France**	**UAE**	**South Africa**
Phones									
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
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
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
Bluetooth Module	Y	Y	Y	Y	Y	Y	Y	Y	Y
									Page 3 of 3

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
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
Visual Workflow Manager	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
Intelligent Directory Application	Y	Y	Y	Y
Live Content Suite	Y	Y	Y	Y
				Page 1 of 2

Latin America (continued)				
	Argentina	Brazil	Chile	Mexico
Phones				
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
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
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
Bluetooth Module	Y	Y	Y	Y
				Page 2 of 2

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