A MITEL
PRODUCT
GUIDE

MiVoice Business

General Information Guide

Release 9.4 SP1
Document Version 1.0

September 2022

Mitel
Powering connections

Notices

Trademarks

Contents

About this Document 1

This chapter contains the following sections:

- Audience
- Related Documentation

Overview

This guide provides an overview of the MiVoice Business 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

MiVoice Business licensing is available to be purchased in either a onetime model (i.e. CapEx), to which annual Software Assurance needs to be applied, or as a monthly reoccurring expenditure (i.e. OpEx), which includes Software Assurance.

MiVoice Business is available to be deployed as a customer specific dedicated instance on the following hardware platforms:

- Mitel 3300 ICP controllers, including MXe III, MXe III-L, CX II, CXi II,
- Mitel EX Controller
- Industry standard servers (x64 Intel based)
- VMware® vSphere™ and Microsoft® Hyper V™ virtualization platforms
- Public Cloud services as virtual servers including Microsoft Azure and Amazon Web Services (AWS)

What's New in this Document

This section describes changes in this document due to new and changed functionality in the MiVoice Business Release 9.4 SP1. The changes are summarized in the following table.

Table 1: Document Version 1.0

Feature/ Enhancement	Document Updates	Location	Publishing Date
-	The entire document has been updated.	-	September, 2022
-	Reference to Multi-Level Precedence and Preemption (MLPP) is removed.	-	September, 2022

1.1 Audience

This guide is intended for:

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

1.2 Related Documentation

Mitel Product Documentation

To access the product documentation please visit https://www.mitel.com/document-center/business-phone-systems

The following guides provide complete information about MiVoice Business and host platforms:

- 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 MiVoice Business.
- 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 MiVoice Business system for a customer site. The guidelines are intended to highlight specific areas of the product that need to be considered before installation.
- Deployment Guides

Overview 2

This chapter contains the following sections:

Mitel MiVoice Business provides businesses of all sizes with a scalable, feature-rich communications system using a single stream of software. MiVoice Business is designed to meet the needs of businesses that have from 5 to 130,000 users, whether they are single-site deployments or multi-site networks that span many countries across onsite deployments, private cloud, public cloud or a mix of all of these. In addition MiVoice Business is available to be purchased using the preferred purchase method of the customer through either capital expenditure or subscription licensing models.

2.1 Platforms

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

- Mitel 3300 ICP controllers, including MXe III, MXe III-L, CX II, CXi II, and AX
- Mitel EX Controller
- Industry Standard servers
- Microsoft Azure virtual servers
- VMware® vSphere™ and Microsoft® Hyper V™ virtualization platforms
- Amazon Web Services (AWS) virtual servers

2.2 Modular Platform Design Provides Scalability and Flexibility

Deploying MiVoice Business 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. MiVoice Business also has the ability to scale with multiple nodes being able to be clustered together to create a logical system with up to 130000 devices.

MiVoice Business Deployment Options

MiVoice Business can be deployed to support a broad spectrum of customer required configurations. For example, you could:

- Centralize in a data center (VMware, Hyper-V, Axure, AWS) and deploy endpoints over the internet, on network (e.g. via SD-WAN).
- Implement a 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.
- A distributed architecture where larger offices have their own independent controllers.
- A combination of centralized primary controller with onsite resilient controllers to provide local support.
- Cluster an entire network of controllers (up to 999 nodes) to function as one large system.
- A network comprised of a mix of any type of supported platform available at the software release level.

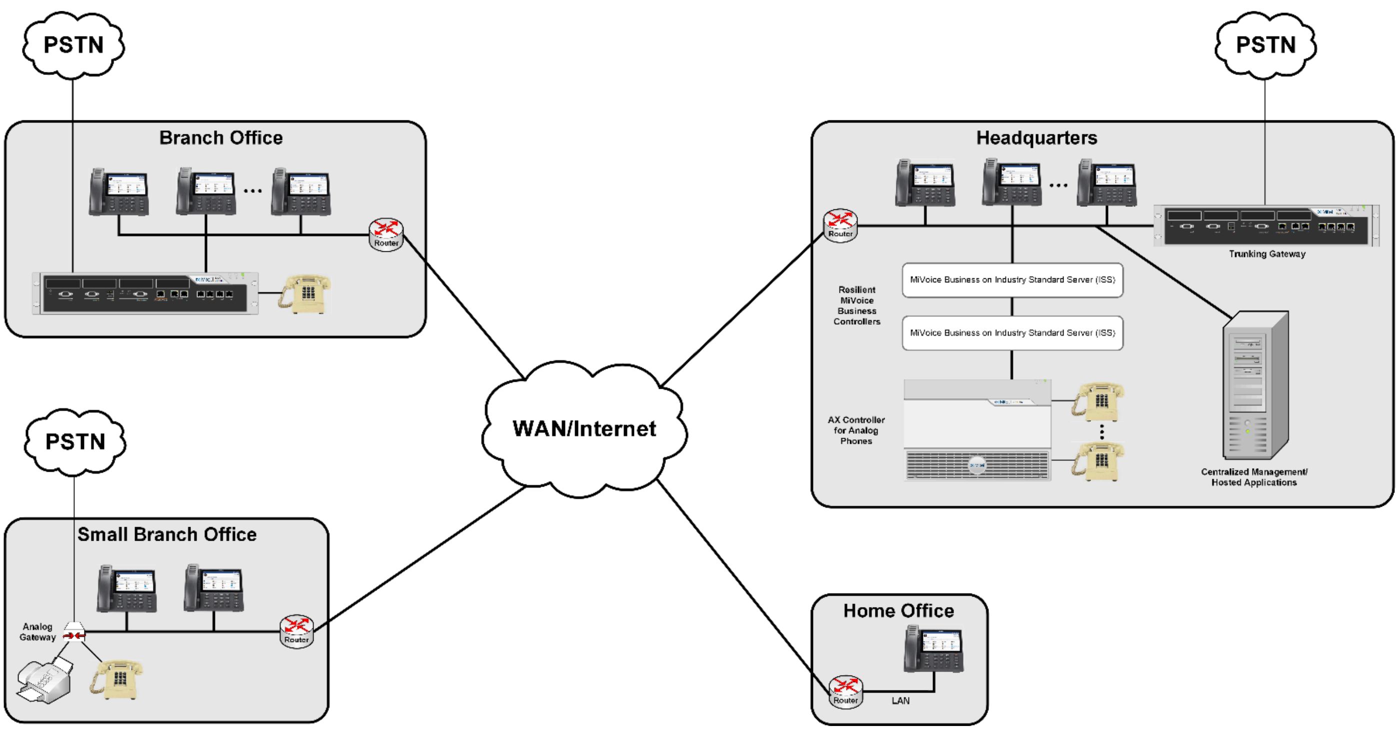


Figure 1: Example of MiVoice Business Centralized Site Configuration with Local Controller.

Figure 1 shows an example MiVoice Business configuration with a centralized instance in the main location, a local controller in the branch office (which may be configured as a primary of resilient node in the network) and a small branch office devices connected over the customer's WAN with home users connected over the Internet. This is just one example of the flexibility of deployment that the MiVoice Business supports

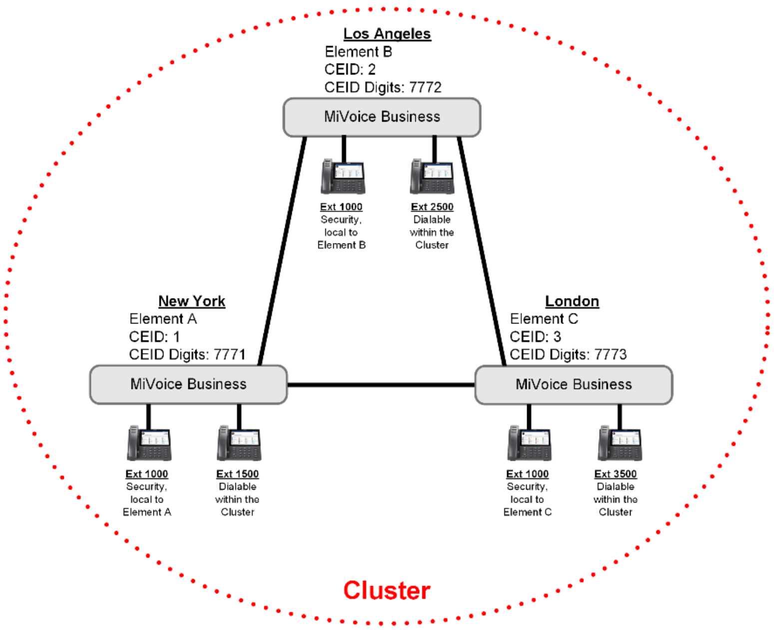
MiVoice Business is complemented with a range of off-board applications including Unified Messaging, Collaboration, Conferencing and Teamwork applications as well as a range of third-party developed applications available through the Mitel's API program – Mitel Solutions Alliance (MSA).

Clustering

For larger organizations or multi-site deployments, up to 999 MiVoice Business controllers, of any platform type, can be deployed in a clustered configuration over a private network (e.g. VPN) creating a single logical system and delivering extensive transparent features, services, and applications. These controllers use a peer-to-peer communications protocol to share control, management and administration data between systems to ensure consistency of features and applications without incurring high management costs.

On large sites, key functionality may be hosted by dedicated "task-specific" controllers with all users connected over the IP network. For example, a large organization might have the following setup:

- 5000 devices on an industry standard server with a backup MiVoice Business Controller to provide resilient support at a different location.
- An MXe III/MXe III-L or EX Controller acting as a TDM trunking gateway with connections to the traditional tele phone network for outside access.
- A dedicated off board Unified Messaging server for voicemail, automated attendant, and unified messaging.



Wide Area Networking Support

As shown n Figure 1 MiVoice Business users and devices can be deployed across a routed network (On-net), across Virtual Private Networks (including MPLS and SD-WAN) and “Over the Top” across the Internet using Mitel’s MiVoice Border Gateway. A customer may have any combination of networking topologies.

Networking With Industry-Standard Protocols

Customers can be assured that their investment in a Mitel solution will be developed and expanded into the future as the solution uses open standards such as Session Initiation Protocol (SIP) and supports IEEE (e.g. 802.1p/Q) and IETF (e.g. DSCP) standards for networking. MiVoice Business meets the complete communications solution needs of today with the flexibility to deliver more in the future while also supporting an extensive list of legacy protocols and devices through dedicated hardware. MiVoice Business can be deployed in public cloud, private cloud or onsite in a distributed, centralized or hybrid architecture as a customers prefer while always providing a dedicated instance of the application for the customer.

Reliability Through Redundancy and Resiliency

Mitel supports VMware® tools to enable voice and business applications to run together in highly available environments. Mitel's unified communications features run as virtual appliances on the VMware virtualization platform and supports VMware High Availability, vMotion and Storage vMotion capabilities. Plus with the ability to also be installed in AWS and Microsoft Azure MiVoice Business can make use of the inherent capabilities of the cloud providers’ infrastructure.

Added to this are Mitel’s IP set resiliency capabilities. IP Set Resiliency automatically transfers support for an IP phone to an alternate controller in the network in the event that the phone cannot communicate with its primary controller. The MiVoice Business instance can be in any of the supported deployment platforms as the feature set is the same. By taking advantage of IP networking, resiliency provides an extremely flexible solution to enhance system reliability in addition to that of any underlying redundancy capabilities that the platform provides. It uses resources that are spread across the network to optimize resources and ensure there is no single point-of-failure.

2.3 Applications that Enhance Productivity

MiVoice Business supports 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 contact centers.

MiVoice Business 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 through the use of industry standard APIs. More information is available at https://www.mitel.com/developer/mitel-solutions-alliance.

2.4 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:

- Industry leading range of IP Phones
- Wireless Phones (DECT and WiFi)
- Session Initiation Protocol (SIP) Phones
- Attendant Consoles
- Conference Units
- Analog Terminal Adapters
- Phone Accessories (includes a range of headsets, and additional button/key modules)
- Softphones
- PC, MAC and Mobile Clients

2.5 Extensive System Feature Set

MiVoice Business has one of the industry's most extensive list of end-user and system features that support effective and efficient communications. The system administrator can enable or disable features through the browser accessed System Administration Tool and can create unique Classes of Service to define levels of feature support for each different group of users creating templates for the same user type or unique Classes of Service as required. For example, a Class of Service can be created to provide executives with advanced calling privilege that standard users don't have access to 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 and can enable or disable system settings across the entire system or network. The range of features is described later in this document.

2.6 Migration Made Easy

Mitel has a long history in voice communications and continues to support a host of protocols which facilitate a smooth migration to Mitel Voice over IP solutions whether a customer's existing PBX is from Mitel or an alternate supplier.

MiVoice Business can be added to an existing mixed vendor network and provide a migration for the customer over their chosen time period and through their preferred purchase method. For example adding a contact center to an existing system. MiVoice Business can also be deployed as a network gateway to link multiple legacy PBX's together, eliminating costly private circuits. And it can be deployed 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. Plus with the MiVoice Business subscription licensing capability the initial outlay for a customer is very approachable to add these new features.

2.7 Technical Training (Partner Certification & End User)

Mitel offers Channel Partners MiVoice Business Technical Certification training that is required to gain access to Mitel Technical Support and forms part of the requirements for the Mitel Partner Program.

MiVoice Business certification training is delivered via remote leader led training with course dates and pricing detailed in Mitel's Learning Management System (LMS) which is accessed via the MiAccess Web Portal. Technical training course options available at time of publishing include:

- 5-day MiVoice Business Core Installation & Maintenance Course Remote Leader-led (Entry Level for Mitel Technical Support)
- 4-day MiVoice Business Network Clustering and Resiliency Installation & Maintenance Course Remote Leader-led (Advanced level)

Mitel also offers end user training courses for customers including videos, user guides & leader-led chargeable private training. Customers should work through their Channel Partner to ascertain the best option for them. Customer training available includes:

- Desktop Devices (Phones & applications)
- System Administration, Moves Adds and Changes

Use the link below to find more information of the range of end user training available from Mitel. https://www.mitel.com/services/mitel-training

MiVoice Business Software Overview 3

This chapter contains the following sections:

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 support, such as POTS and TDM PSTN trunks, the EX and 3300 ICP platforms include TDM/IP Gateway capabilities. 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.

3.1 Licensing

The MiVoice Business license strategy delivers simplicity and flexibility while maintaining cost effective- ness. MiVoice Business can be deployed on purpose-built hardware, on industry standard servers or virtualization platforms, or on public cloud platforms.

Mitel offers multiple user licensing levels providing a customer the ability to select the preferred licensing level to match the user’s roles in their organization.

MiVoice Business can be procured as a capital expenditure (CapEx) or as a subscription (OpEx). Licenses are either perpetual or subscription. With Subscription licensing the focus of the license is at the user level and capabilities such as SIP trunks or Teleworker functionality are bundled in with the user license.

Licensing Management

There are three types of license management available for MiVoice Business systems:

- Standalone Systems - are licensed with a single Application Record for each system.
- Non Shared Enterprise Systems – these are Enterprise Systems licensed with individual Application Records that do not share licenses across MiVoice Business instances. Each system is linked to its own Application Record.
- Enterprise Licensing – where a solution is programmed to allow multiple MiVoice Business systems to be amalgamated into a single Application Group record. More details follow.

Enterprise Licensing

Enterprise Licensing (License Sharing) is a License Manager capability available for MiVoice Business Enterprise Systems that allows a customer to easily move licenses around their solution. With Enterprise Licensing a can move licenses across MiVoice Business instances in their network with no involvement from Mitel or their Solution Provider. It requires that an instance in the network be configured as the Designated License Manager (DLM). By activating Enterprise Licensing, the customer is setting up MiVoice Business licensing to work as a single solution rather than a group of individual licensed nodes.

All licenses in a DLM must be of the same procured type. That is they must all be perpetual or all be subscription within the DLM. Enterprise customers who do not wish to invoke Enterprise Licensing can continue to license their systems individually (the non-shared model).

System Type Licensing

MiVoice Business systems are activated either as:

- Standalone (not available in all markets) or
- Enterprise Systems.

The underlying software running the two system types is the same. The different system types allow Mitel to present the MiVoice Business solution to different markets and customer segments while using the same software stream.

While a Standalone system may be upgraded to an Enterprise system when desired Mitel encourages the use of Enterprise licensing over Standalone licensing as it already includes the ability to expand to multiple nodes (i.e. clustering) at the Enterprise level.

The Enterprise System includes the inherent networking capabilities of MiVoice Business along with full User and Device resiliency.

MiVoice Business User Licensing

Individual MiVoice Business User Licenses include:

- User License: enables an IP device to be fully activated for all features or can be used for a Hot Desk User as the user logs on to a phone that does not have a license.
- External Hot Desking license 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 license: permits concurrent usage license—one required for each concurrent agent log in.
- Single line license: one license required per analog or single-line SIP termination.
- Voice mail licenses: one license is required per MiVoice Business mailbox, also used for auto attendant applications.
- Multi-device User License: intended for users who have a range of up to eight devices: desk phones, soft phones, mobile phones, in-building wireless phones.
- Suite License: used in Hospitality solutions where up to six phones can be added to a single suite using only a single Suite license.

Unified Communications and Collaboration Licensing

Unified Communications and Collaboration (UCC) licensing is a combination of MiVoice Business and Mitel application licensing. UCC licensing simplifies the licensing process as it bundles the MiVoice Business and application user licenses together into one purchasable bundle. Instead of ordering a MiCollab license (see MiCollab section), PBX user licenses, and multiple individual applications licenses for each user, with UCC licensing you can order a single UCC license per user. UCC licensing offers the following benefits:

- Improves the competitive offering.
- Simplify the sales, ordering and deployment process.
- Rewards customers who purchase more functionality.
- Focuses on core end user needs.

UCC Licensing types include:

- UCC Entry User license: The UCC Entry target user is an office-based worker. It includes a Multi-Device User license, MiCollab voicemail, a desktop or web client with Chat capability, presence, Outlook calendar presence integration, and AWV collaboration Participant capabilities.

- UCC Standard User license: The UCC Standard target user is a mobile information worker. UCC Standard includes everything from UCC Entry plus a mobile client, additional Teleworker licenses, and AWV licensing to create, manage, and lead conferences.

Trunking and Compression Licenses

Individual Trunking and Compression Licenses include:

- SIP Trunk license – licensed per simultaneous SIP Trunk call.
- G729 Compression licenses – required for TDM calls that need to be compressed when connecting to an IP user or SIP trunk.
- T.38 Licenses – used with Fax to IP conversion.
- Digital Link License – Licensed per E1 or T1 link E1.

3.2 Compression

Bandwidth optimization is a key requirement of VoIP systems: MiVoice Business supports G.711, G.722.1, Opus and G.729A Codecs. It compresses calls using the G.729A Codec. Compression via G.729A reduces the bandwidth of a call from 64 kbps (G.711) to 8 kbps (G.729A) 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. Groups of devices are placed into zones based upon their location in the customer network to compress calls between zones if needed. Opus and G.722.1 Codecs also use less bandwidth than G.711.

Mitel IP Phones include support for 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 (e.g. 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.

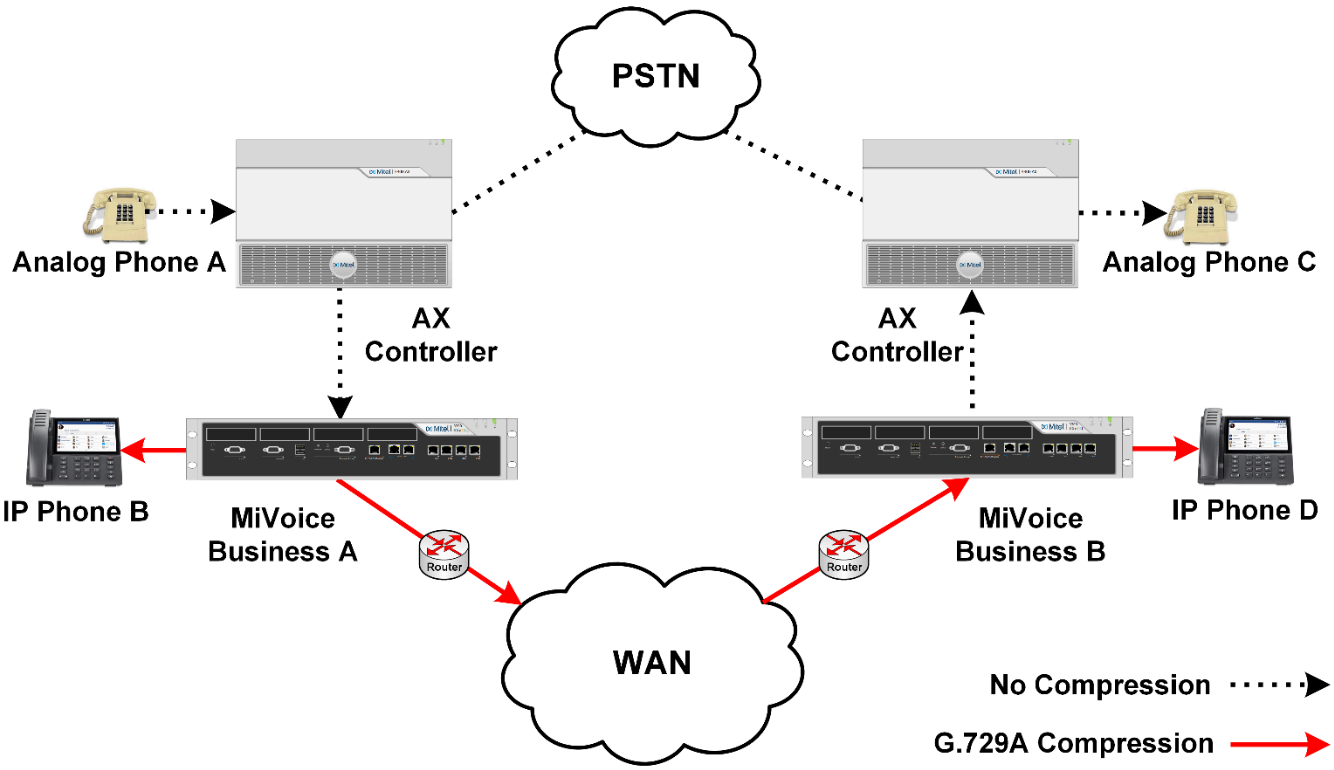
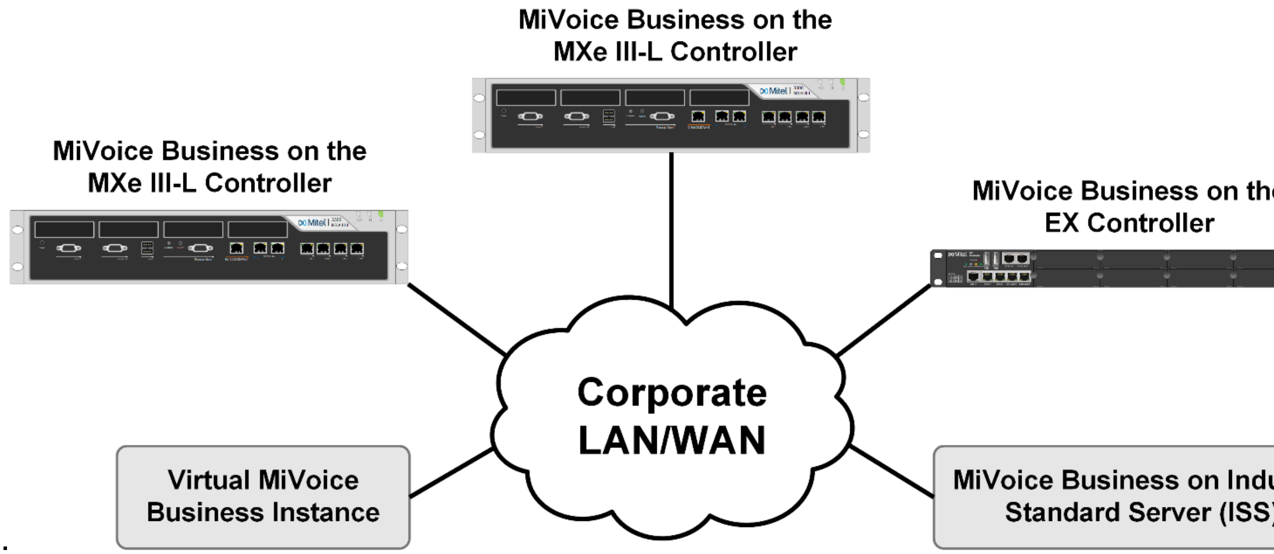


Figure 2: Example of Voice Compression Between MiVoice Business systems.

3.3 IP Networking

IP Networking enables multiple MiVoice Business systems to be networked together. Instead of leasing dedicated voice circuits, calls can be routed over the existing LAN/WAN infrastructure. The Mitel IP Networking implementation uses point-to-point topology to optimize network



resources.

Figure 3: Example of IP Networking - Point to Point Topology Illustrating Various Possibilities.

IP Networking also supports the clustering of instances in a single location to provide greater resiliency than that of a single controller operating autonomously independent of the underlying platform. You can seamlessly network geographically separated controllers to share information and services in a transparent and cost efficient manner. A total of 2000 IP network connections are supported from any one node and up to 2000 connections can be defined between any two nodes.

3.4 SIP Trunking

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

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

Mitel operates a SIP interoperability team that undertakes testing activity and certifies and publishes integration with SIP service providers, devices, gateways, services and applications that connect to MiVoice Business. This information is available to Mitel Channel Partners through the MiAccess Portal.

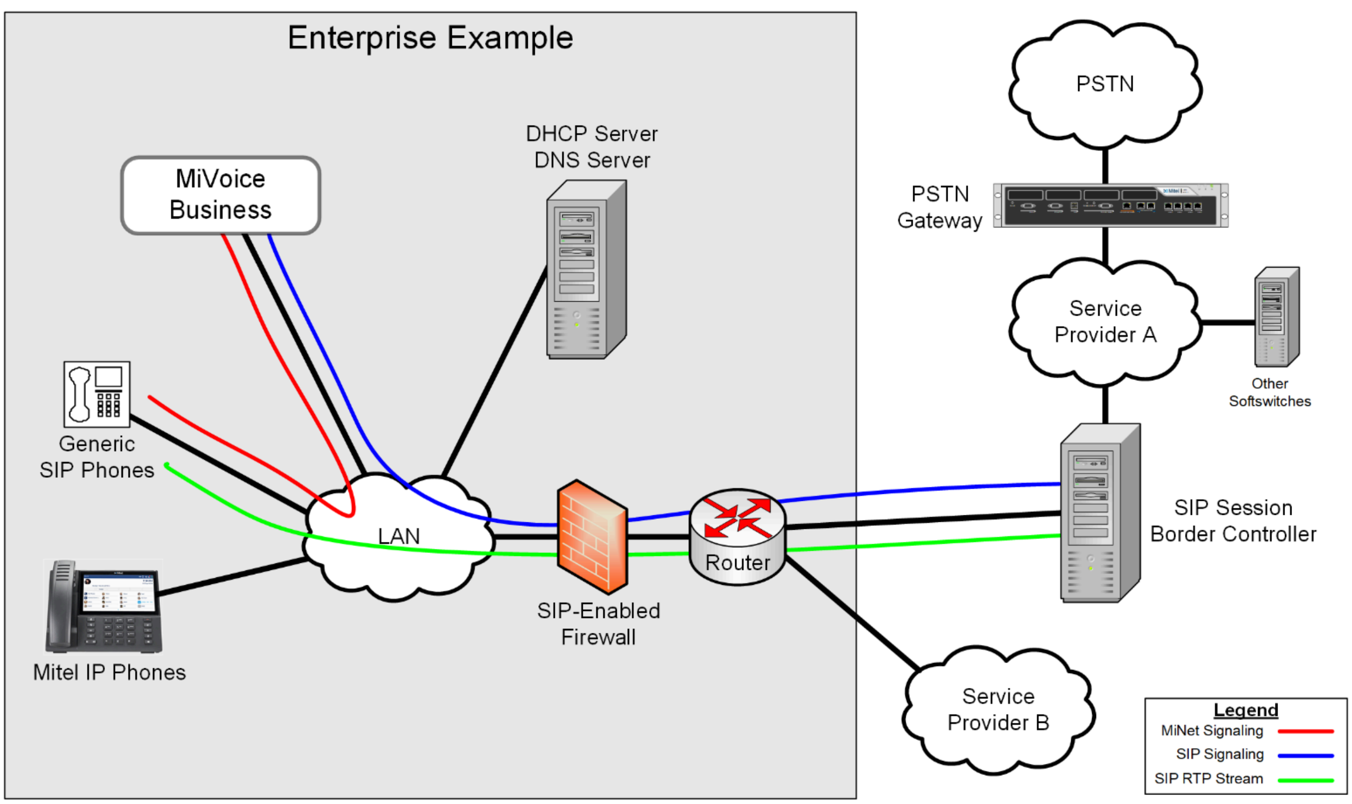


Figure 4: Example of a SIP Trunking Setup

3.4.1 Configurable Real-time Transport Protocol (RTP) Packetization

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

3.4.2 Malicious Call Trace

For incoming SIP calls that are tagged as Malicious Calls, MiVoice Business 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.

3.4.3 FAX Support

You can configure the MiVoice Business 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 MiVoice Business systems.

Optionally Mitel also provides off board T.38 capabilities through SIP Analog Terminal Adapters (ATA) such as the TA7102, TA7104, and TA7108.

3.5 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 MiVoice Border Gateway teleworker service) hosted by a centralized MiVoice Business 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.729A Codec. Compression is applied to calls between the zones of IP Phones.

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).

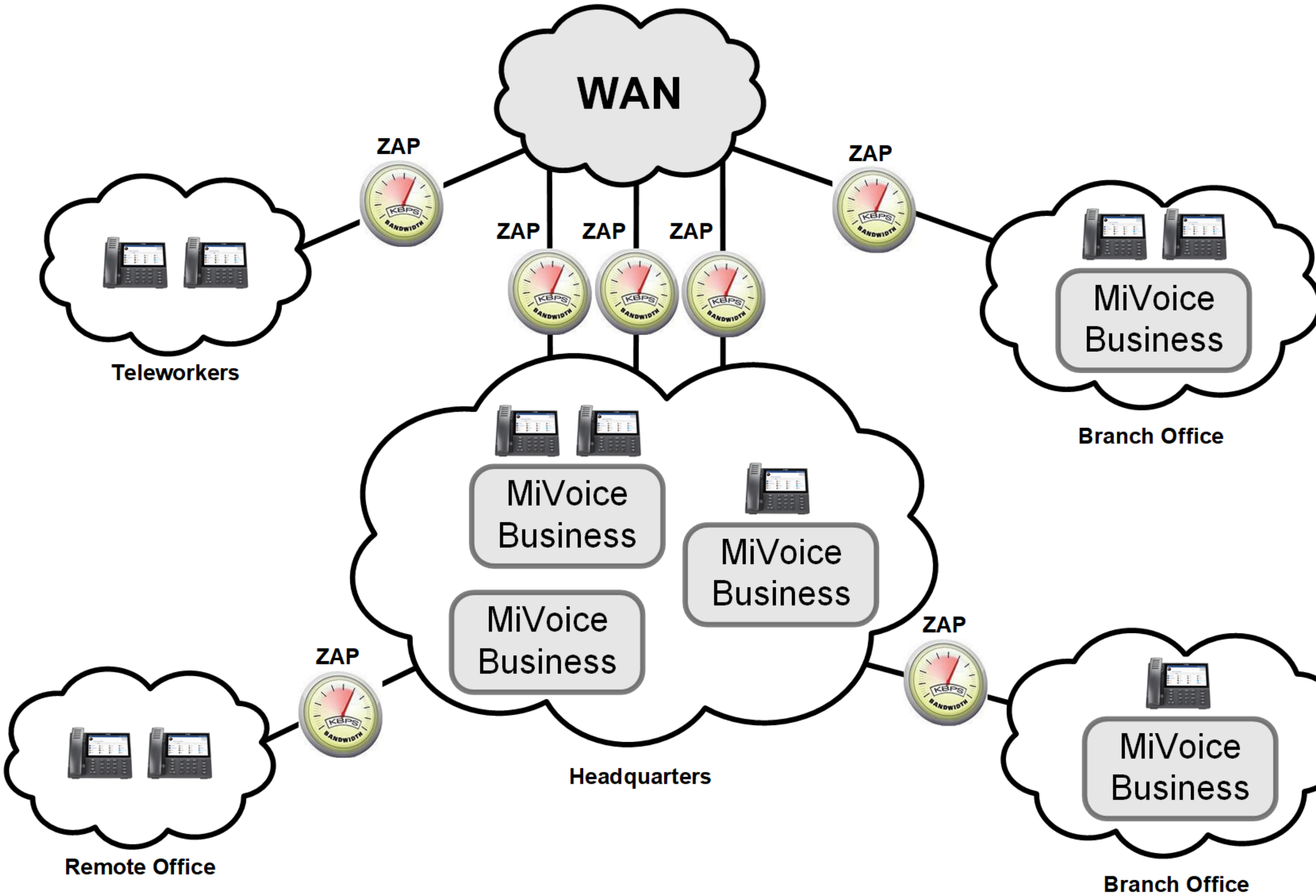


Figure 5: Example of Bandwidth Management Zone Access Points

3.6 Resiliency

Resiliency on MiVoice Business 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 MiVoice Business or network failures. The Resiliency solution preserves system functionality in the event of network difficulties by distributing network intelligence throughout “resilient” nodes in the cluster that 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, internal 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.

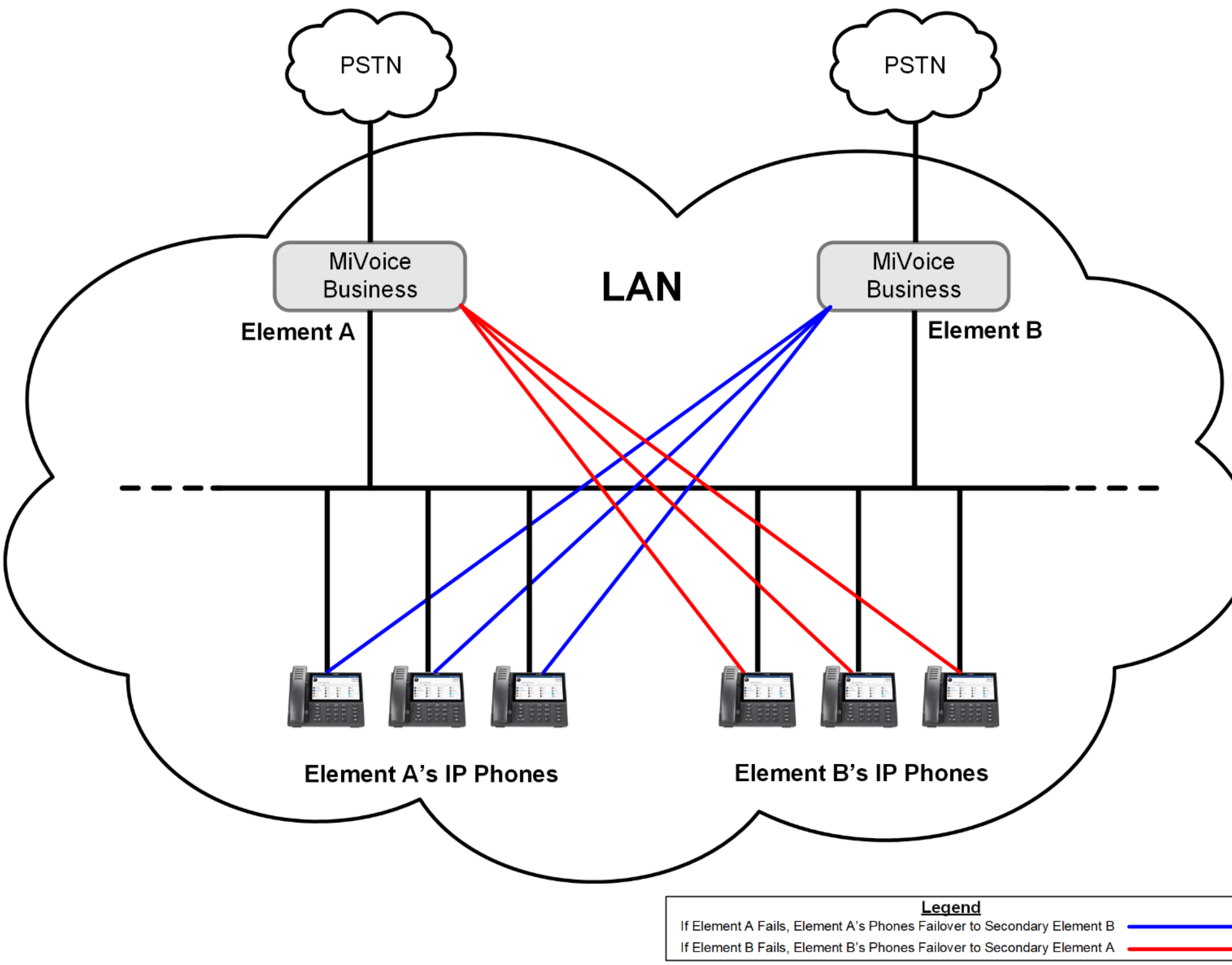


Figure 6: Resilient Configuration

Resilient solutions add another layer of availability to a customer's solution and are more flexible than redundant only solutions. MiVoice Business resilient networks enable you to route around failed or otherwise inaccessible portions of an IP network.

MiVoice Business resiliency is platform independent meaning that a customer can choose to deploy their MiVoice Business nodes across any of the supported underlying platforms e.g. it is possible to have a primary MiVoice Business deployed in Azure with a resilient node in AWS.

Because any controller in the network can act as a secondary controller, Mitel Resiliency can be referred to as an “N +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)
- Call center controller
- IP network gateway
- PSTN gateway
- Voice mail server

For detailed information on Resiliency, see the MiVoice Business Resiliency Guidelines.

3.6.1 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

3.6.2 Devices that Support Resiliency

Mitel IP devices that support resiliency include:

- All 6900 series IP Phones
- All 5300 series IP Phones
- 5540 IP Console
- MiVoice Business Console
- Teleworker service sets

For detailed information on Resiliency, see the MiVoice Business Resiliency Guidelines.

3.7 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.

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
- The Scheduler tool can be used to automatically log out Hot Desk Users at a set time.

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 MiVoice Business.

MiVoice Business also 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.

3.7.1 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 Hot Desk 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.

3.7.2 Multi-device Capability

MiVoice Business provides a multi-device capability whereby a user can have their single number contact them at multiple locations at the same time e.g. their desk, home, and mobile device. The capability is most beneficial for users that are highly mobile.

There are two primary ways of achieving this:

- By licensing the user as a Multiple Device User whereby the user consumes one Multi-device Users (MDU) license. Up to 8 devices (internal or external to the MiVoice Business) may be called simultaneously, but the user can answer and talk on only one at a time (i.e.one busy all busy)
- Or by using a Personal Ring Group (PRG) where up to 8 devices can also be called but each of the members of the PRG consumes an IP User license, but all the devices may be used at the same time in conversation.

3.8 Embedded Unified Messaging

Embedded Unified Messaging (UM) enables users to receive and manage voice messages through either SMTP forwarding or IMAP-enabled e-mail clients such as Outlook. With IMAP enabled, 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 mutually exclusive with Standard Unified Messaging. Only one or the other can be enabled for a mailbox.

3.9 Embedded Voice Mail

MiVoice Business includes an integrated fully-featured voice mail system. The number of embedded voice mail ports differs between platform types with up to 120 ports being available on the on server based platforms. Please check Engineering Guidelines for latest information. The embedded voice mail system supports:

- A maximum of 750 mailboxes on appliances and 5000 on servers.
- A storage time of 450 hours with 13 or 14 GB partition (130 hours if backup is required) and 130 hours with 4 GB partition (30 hours if backup is required).

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.

With the server based platforms the Call Flows feature may be used in place of the MLAA. Call Flows is a collection of call-processing actions programmed to control how an incoming call is handled and adds powerful call-processing capabilities to the Embedded Voice Mail system. With Call Flows and administrator can create automated attendant and call processing applications for their organization, for departments within the organization, and for individual mailboxes and extensions.

3.10 Embedded System Management

Embedded System Management (ESM) is the browser accessed administration tool used by MiVoice Business and includes the following end user tools:

- Desktop Tool
- #unique_32/unique_32_Connect_42_id21CD8200B5Z
- System Administration Tools

3.10.1 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.

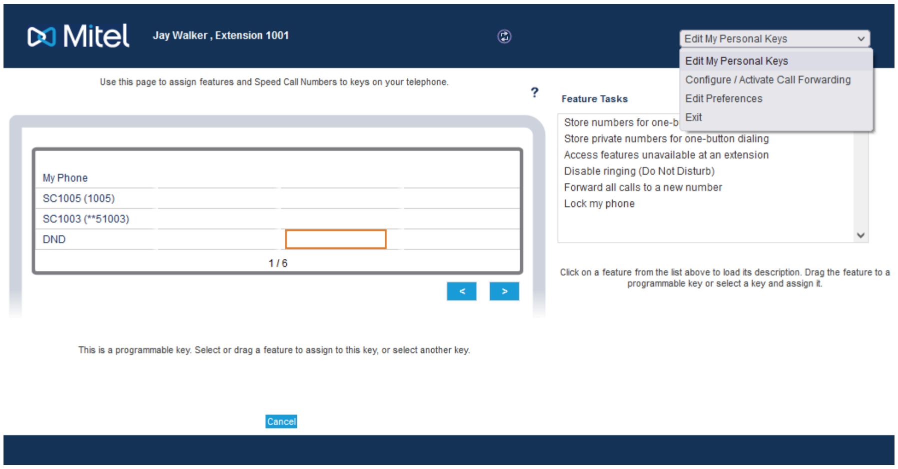


Figure 7: Example of Desktop Tool Interface

3.10.2 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 and provides access to Maintenance Logs, Software Logs, and Login and Logout Audit Logs. 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 trouble- shooting 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 MiVoice Business 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 MiVoice Business 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 group of MiVoice Business network elements (up to 20) effectively and conveniently without the need for a management tool such as Mitel Performance Analytics. 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 MiVoice Business System Administration Tool Help available in the Mitel Document Center.

Note:

- Network elements must be grouped together within an SDS Administrative group to use this feature.
- The System Administration Tool is available in English only.

The User and Service 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 service data using .CSV files. These spreadsheets contain built-in validation similar to ESM data entry rules which helps reduce errors.

The following figure illustrates the User and Service Configuration Form.

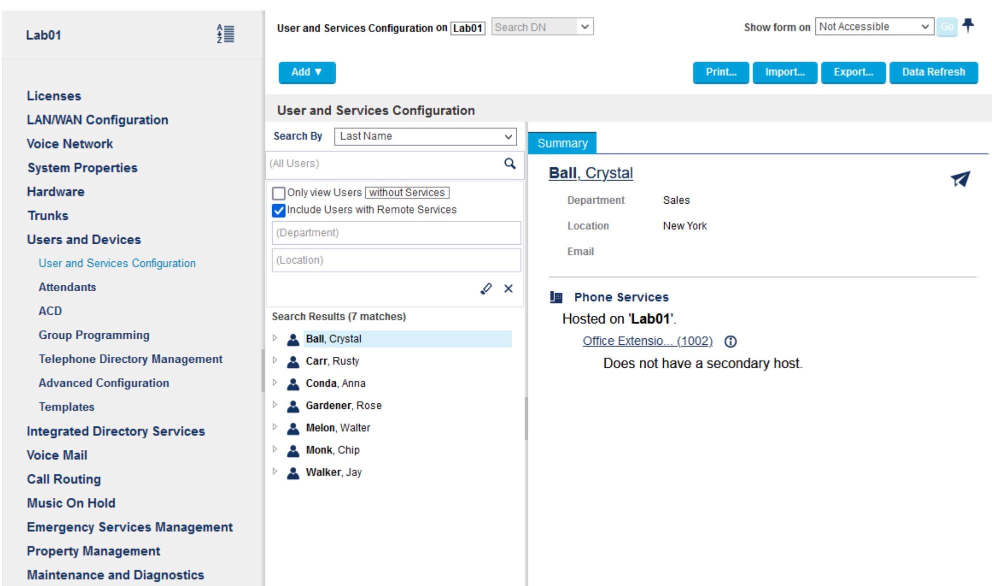


Figure 8: Example of User and Services Configuration Form

3.10.3 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.

3.10.4 Remote Alarms Notification

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

MiVoice Businesssupports 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).

The SNMP agent in MiVoice Business communicates with SNMP- Network Management Stations and supports industry-standard MIB-II definitions as well as proprietary SNMP extensions. The MIB is available for download.

3.10.5 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.

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

3.10.6 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 MiVoice Business 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.

3.10.7 System Data Synchronization

The System Data Synchronization application enables administrators to synchronize database information among a network or cluster of MiVoice Business systems. Database changes made to a platform in the network or cluster are applied to the other platforms System Data Synchronization is an enabling technology that:

- Reduces the time to provision and administer multiple MiVoice Business 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 MiVoice Business nodes with the existing network.
- Enhances security management across the network by allowing accounts and passwords to be managed centrally.

3.10.8 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 MiVoice Business system to the Public Safety Answering Point (PSAP).

> **Note**:
>
> PSAP applies terminology applies to North America, only.

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 data- base within MiVoice Business be kept up to date.

CESID is supported on analog sets, Mitel's IP sets, 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 analog phones change offices or relocate within a building, a manual update is required to the MiVoice Business database and the phone move is typically managed by the Tele- coms/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 MiVoice Business automatically updates CESID.

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

MiVoice Business RAY BAUM'S Act and Kari's Law (USA Only)

The USA government has adopted rules for implementing two federal laws that strengthen emergency calling: Kari's Law and Section 506 of RAY BAUM'S Act.

The RAY BAUM'S Act classifies devices into:

- Fixed MLTS devices - devices that connect to a single end point (e.g., a desk or office phone) and are not capable of being moved to another endpoint by the end user, although they may be capable of being moved to a different endpoint by a professional installer or network manager.Non-Fixed MLTS devices - devices that the end user can move from one endpoint to another without assistance.

The MiVoice Business solution implements RAY BAUM'S Act support in conjunction with NG911 providers as the MiVB 911 solution alone does not satisfy the legislated requirements for the RAY BAUM'S Act for all non-fixed devices. The customer therefore requires a valid service agreement with an NG911 provider. Mitel does not provide this service agreement directly. 911 calls are sent via SIP trunk to the NG911 provider and the NG911 provider will redirect to the appropriate Public Safety Answering Points (PSAPs) based on the Civic Address of the location as identified by the NG911 provider.

For Kari's Law requirements direct dialing of 911, the MiVB can be pre-configured to directly dialing of 911 (emergency calls), without having to dial any prefix or access code. However, in order to support Kari's Law local notifications, the solution uses the NG911 provider's notification application. The MiVoice Business notifications (including Mitel Revolution) provide supplemental information and are not sufficient to satisfy Kari's law on their own.

For more detail, please see MiVoice Business RAY BAUM'S Act Solutions Guide.

3.11 MiVoice Business System Hardware

MiVoice Business is available on multiple platform types including public cloud, private cloud, industry standard servers, VMWare as well as purpose built appliances from Mitel.

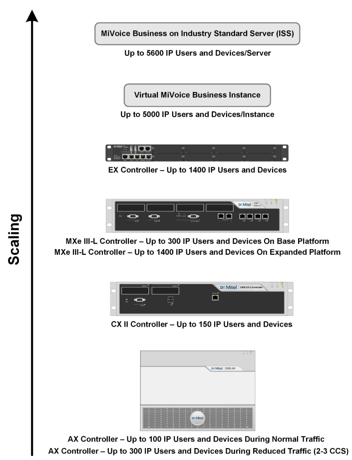


Figure 9: MiVoice Business Hardware Platform Scaling

An example of the purpose built appliances are the Mitel EX controller and the 3300 ICP range.

EX Controller

The EX Controller is a hardware platform that supports the MiVoice Business call processing software and has built-in Digital Trunk, Analog Trunk, and Analog Phone capabilities. It combines the virtual capability of MiVoice Business with SIP Gateway capabilities. A Virtual MiVoice Business instance runs on a Kernel-based Virtual Machine (KVM) environment on the EX Controller which provides a virtual operating environment to run the Virtual MiVoice Business instance. Only one MiVoice Business Instance is allowed per EX Controller.

The EX Controller has eight card slots for Analog Phones, Analog Trunks, PRI Digital Trunks, and a DSP. It supports up to 1400 IP users and provides analog capabilities of up to 28 Foreign Exchange Subscriber (FXS) or Foreign Exchange office (FXO) ports or 8 T1/E1 ports.

There are two EX Controller variants available:

- EX Controller with 4 GB RAM and 60 GB storage

- EX Controller with 8 GB RAM, 120 GB storage, and Dual PSU

The architecture and deployment of the EX Controller is different from MiVoice Business 3300 Integrated Communication Platform (ICP) controllers and hardware is not interchangeable.

Figure 10: Ex Controller

Supported Hardware Configuration

The EX Controller has eight slots for accommodating the following cards:

- 1 port PRI (T1/E1) card
  - Supports R2
  - Does not support DPNSS or Q.SIG
- 1 DSP card (required one per chassis when using an FXS or FXO card)
- 4 port FXS card
- 4 port FXO card
- 4 port FXO card – Australian variant

Note:

The EX Cards are not the same as the MMC Modules used in 3300 ICP systems.

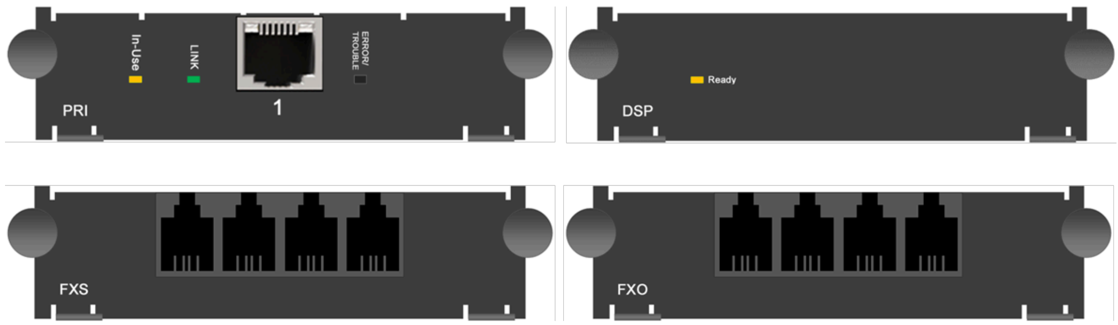


Figure 11: EX Controller Card Slots and Modules

With an EX Controller, there are always enough TDM to IP Channels to cover the Digital and Analog Ports because the interface cards themselves have this functionality. For example, an EX Controller with eight PRI Cards can use all channels simultaneously for Ethernet to TDM Conversion and associated Echo Cancellation. The interface cards in the EX Controller can also handle Compression on all channels and T.38 Fax is enabled for all channels by default.

This is non-blocking since the IP Media Endpoints are either IP Phones and/or EX interface cards which have enough TDM to IP Gateway resources to cover all channels. Therefore, the EX Controller fully supports up to 240 gateway channels, all of which can also support Compression and T.38 Fax.

MiVoice Business System Functionality

4

This chapter contains the following sections:

- 3300 ICP Hardware Overview
- 3300 ICP Processors, Cards, and Modules
- Analog Expansion Support

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

4.1 3300 ICP Hardware Overview

The 3300 ICP range is a range of purpose-built hardware appliances for the MiVoice Business solution. The controllers provide the voice, signaling, central processing and communications resources for the system. Mitel offers several types of controllers that scale to meet the requirements of small-to-large sites:

- CX II: 2nd generation version of the CX controller that provides support for up to 150 devices without the need for additional DSP resources.
- AX Controller: optimized for analog devices, this unit supports a maximum of 100 IP devices or a maximum of 288 ONS devices (or a combined maximum of 300 devices). Note that when installed in a low traffic environment (for example, Hospitality), the AX can support 288 analog sets and 300 IP sets, for a combined total of 575 devices. Up to 300 IP devices can be supported under low traffic conditions.
- MXe III/MXe III-L Standard Controller: supports a maximum of 300 IP devices or 196 ONS devices (or a combined maximum of 350 IP/ONS devices)
- MXe III/MXe III-L Expanded Controller: supports a maximum 1400 IP devices or 576 ONS (or a combined maximum of 1500 IP/ONS devices)

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 T1/E1, T1/E1 Combo Card, 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.

4.1.1 CX II Controllers

The CX II comes with an embedded Analog Main Board that supports 6 analog trunks and 4 analog extension ports. The CX 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 Controller supports

- Up to 150 IP devices, or up to 150 combined IP/ONS devices
- The Analog Main Board (AMB): provides six LS trunk ports with CLASS support (CLASS is available in North America and Latin American only), four ONS ports, a single Music-on-Hold port (one source supported), a single Paging port (one paging zone), and two System Fail Transfer circuits.
- The Analog Option Board (AOB): provides six LS trunks ports with CLASS support, four ONS ports, one System Fail Transfer circuits and one paging circuit
- One 10/100 BaseT WAN port (RJ-45 connector)
- One 10/100/1000 BaseT LAN port (RJ-45 connector)
- SATA solid state 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
- A Quad Copper Interface Module (CIM) for connection of up to three Analog Service Unit IIs (ASU IIs)

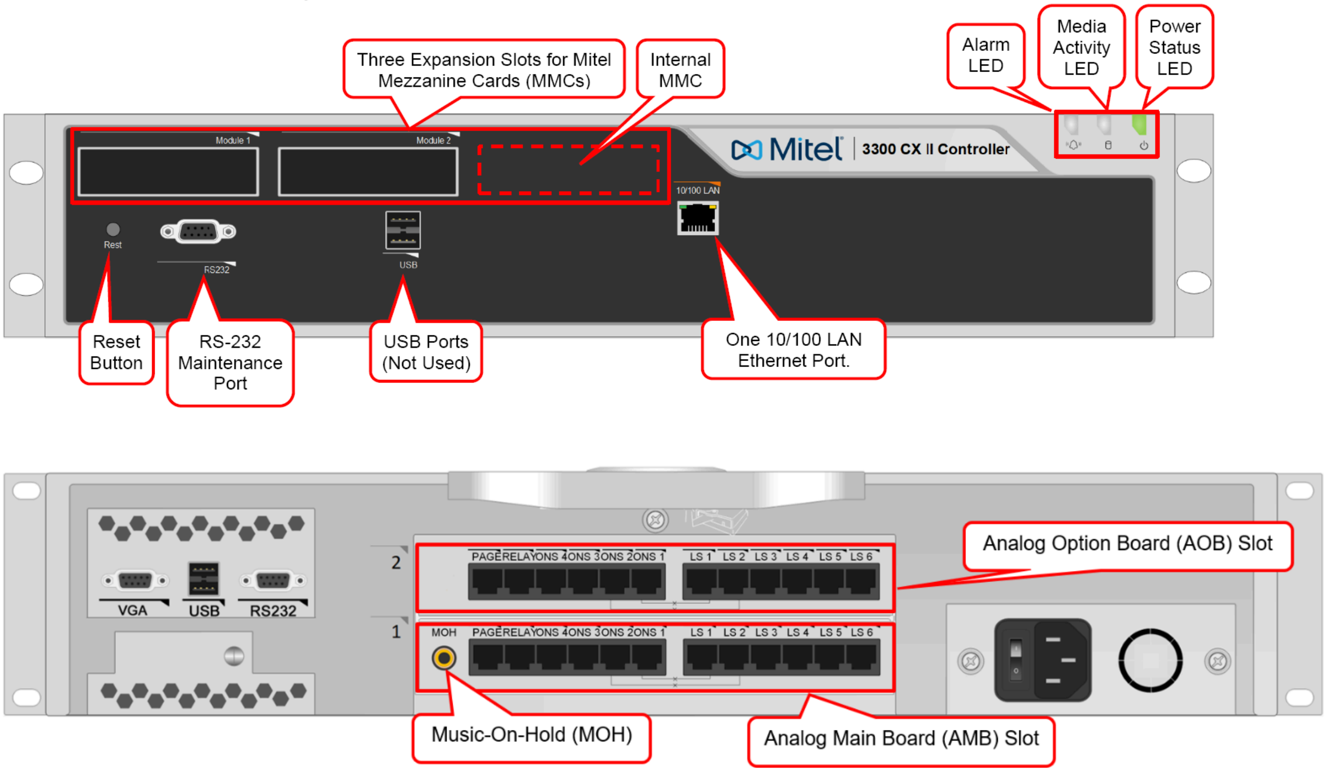


Figure 12: CX II Controllers

4.1.2 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.

At normal traffic levels the AX Controller supports a maximum of 100 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 300 IP sets, with a combined capacity of 575 devices.

The AX Controller provides support for:

- 12 line card slots to support analog phones and trunks.

The following cards are available:

- 24-port ONS line card
- 4 + 12 port combo card (4 analog trunks and 12 ONS ports)
- Two 10/100 BaseT Ethernet LAN ports (RJ-45 connector)
- One externally accessible expansion slot and one internal expansion slot for up to two of the following optional modules:
- Quad DSP (external or internal)
- Echo Canceller (external or internal)
- Dual T1/E1 (external)
- T1/E1 Combo (external)
- DSP II (internal or external)

Optionally, you can install

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

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.

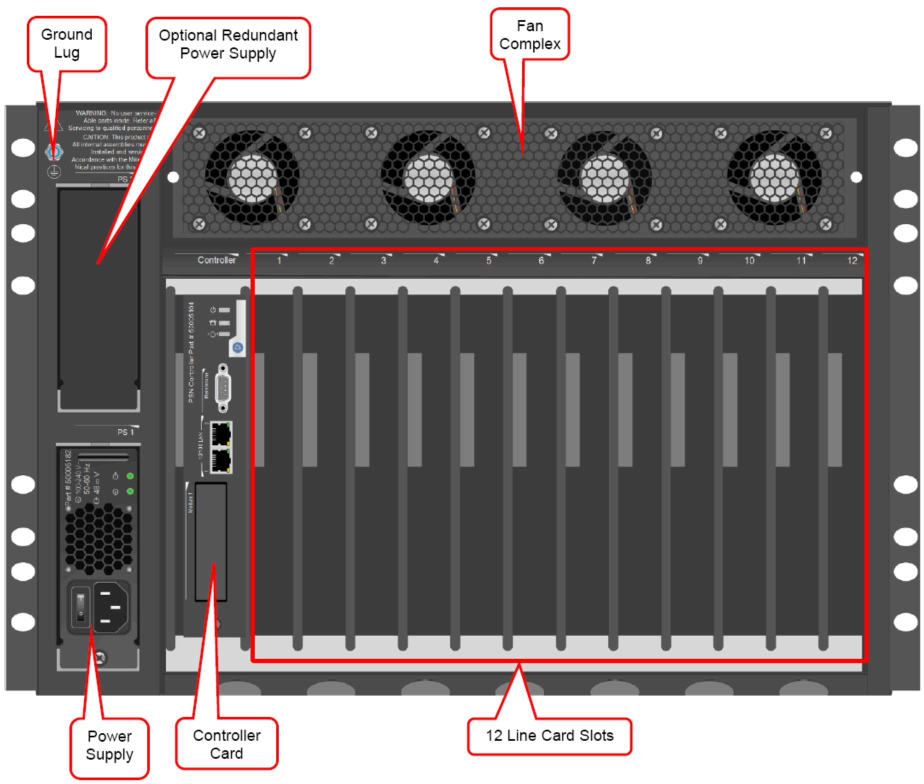


Figure 13: AX Controller Rear View

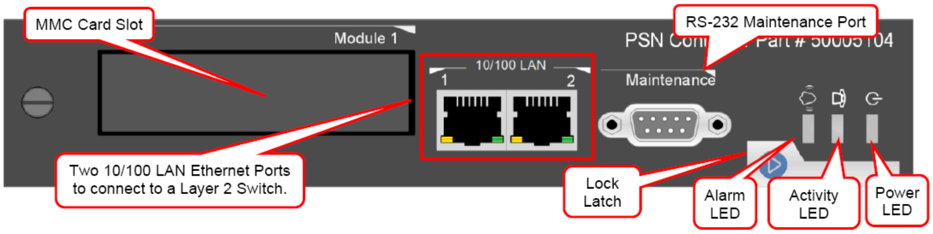


Figure 14: AX Controller Control Card

4.1.3 MXe III/MXe III-L Controllers

The MXe III/MXe III-L Controllers are available in two capacities: standard and expanded. Both versions include an embedded Analog Main Board and redundant cooling fans.

The MXe III/MXe III-L 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/MXe III-L Controllers provide:

- Two 10/100/1000 BaseT Ethernet LAN ports (RJ-45 connector) **(MXe III only)**
- Two 10/100 BaseT Ethernet LAN ports (RJ-45 connector) **(MXe III-L only)**
- One 10/100 BaseT Ethernet WAN port (RJ-45 connector) **(MXe III only)**
- Four externally accessible slots and two internal slots for optional modules
- Four Copper Interface Module (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 III Expanded Processor Package to upgrade from standard capacity (350 devices and 64 Ethernet to TDM (E2T) - channels) to expanded capacity (1400 devices and 128 E2T channels)
- Two DSP II modules for G.729A compression
- Two octal DSP II modules for G.729A compression and T.38 FAX support
- Up to four Dual T1/E1 modules
- Up to three T1/E1 Combo modules
- Power and disk drive redundancy with the addition of a RAID (Redundant Array of Independent Disks) controller, a second solid state drive, and a second AC PSU

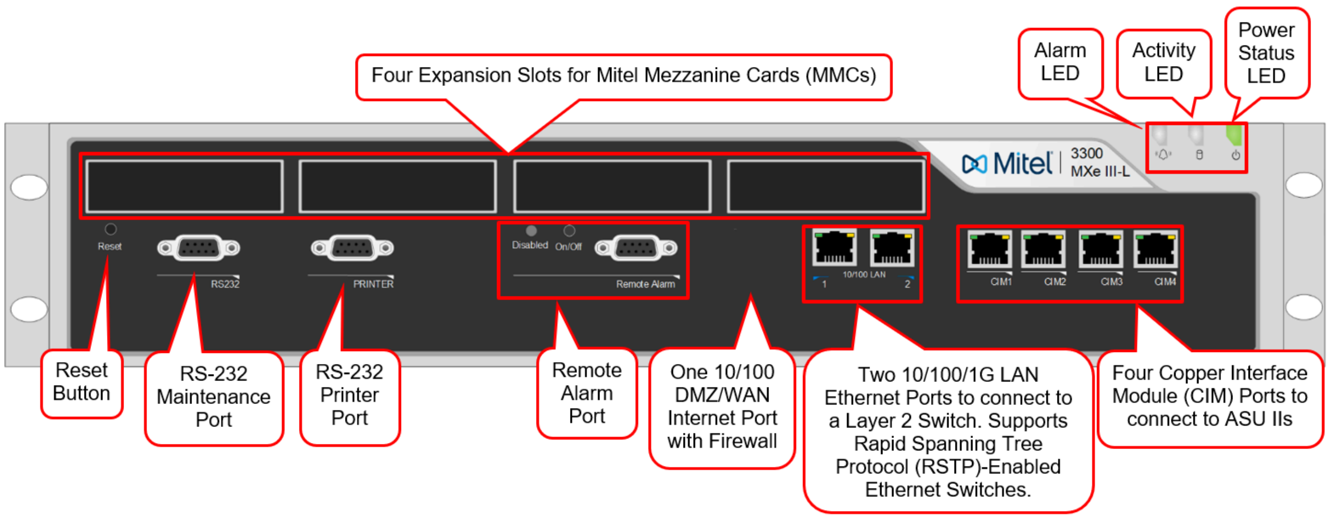


Figure 15: Front Panel of the MXe III-L Controller

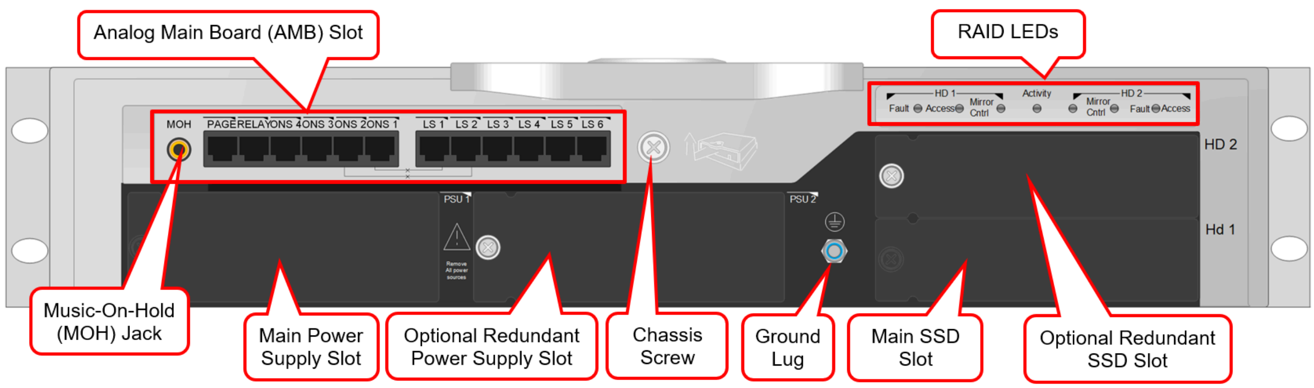


Figure 16: Rear Panel of the MXe III-L Controller

The MXe III-L is available in North America, the United Kingdom, Australia, New Zealand, and Middle East and Africa.

4.2 3300 ICP 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.

4.2.1 Processors (E2T/RTC)

The CX II/CXi II, AX, and standard MXe III/MXe III-L Controllers use a single processor to perform the Real Time Controller (RTC) functions and the Ethernet-to-TDM (E2T) functions. The expanded MXe III/MXe III-L 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.

4.2.2 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.

4.2.3 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.

4.3 Analog Expansion 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 2: Analog Support

Controller	Quad CIMs	ASU IIs	Analog Main Board	Analog Option Board
CX II	1	3 with one Quad CIM installed	1	
AX	0	Not Applicable – up to 288 internal		
MXe III/MXe III-L	2	4 without any Quad CIMs installed 8 with one Quad CIM installed 12 with two Quad CIMs installed	1	Not supported
MiVoice Business for Industry Standard Servers / MiVoice Business Virtual	0	Not supported		

4.3.1 Quad Copper Interface Module (CIM)

A Quad CIM MMC provides four CIM ports that allow you to connect ASUIIs 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/MXe III-L 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.

4.3.2 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-Premises 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.

4.3.3 Analog Main Board/Analog Option Board

The MXe III/MXe III-L and CX (II) Controllers all support the Analog Main Board (AMB). In addition, the CX (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.

MiVoice Business Network Support 5

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 MiVoice Business as a gateway into IP telephony has many advantages.

By integrating MiVoice Business 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. MiVoice Business 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.

In addition to MiVoice Business Mitel can enable customer migration through the use of applications such as contact center, unified messaging and mass notification with dual integrations.

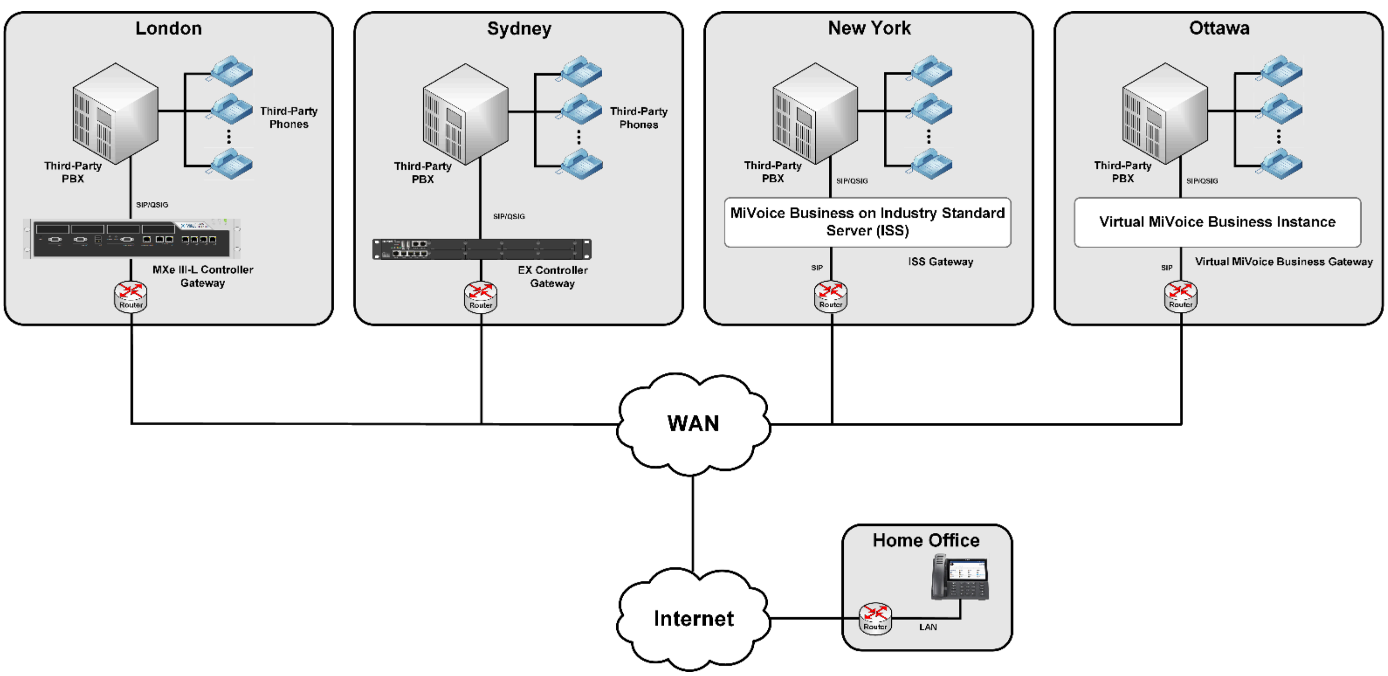


Figure 17: Example of MiVoice Business as Gateway

Applications 6

This chapter contains the following sections:

- Mitel Unified Communications with MiCollab
- MiCollab Unified Messaging
- Mitel Unified Communicator
- Desktop Devices
- Features

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

- MiCollab
- Mitel Collab Advanced Messaging
- Microsoft Teams Support
    - Support for Microsoft Direct Routing (requires external SBC)
    - Mitel Assistant for Microsoft Teams
    - Call2Teams Interworking (third party solution)
- MiVoice Border Gateway
- Mitel Business Analytics
- Mitel Performance Analytics
- Mitel MiTeam Meetings
- Mitel Revolution
- MiContact Center Business
- Mitel Interaction Recording

Some applications are embedded in the system software and others are supported externally. For more information, see the sections that follow.

6.1 Mitel Unified Communications with MiCollab

Mitel's MiCollab enhances the MiVoice Business by adding teamwork, conferencing, chat and collaboration tools with co-workers, customers, and partners, helping employees to make better decisions, be more responsive, and deliver greater value to their clients.

MiCollab integrates with MiVoice Business to provide small and medium-sized businesses with a set of advanced IP applications. MiCollab 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:

- MiCollab Unified Messaging
- MiCollab Speech Auto Attendant
- MiCollab Mobile Client
- MiVoice Border Gateway with Teleworker, SIP Trunk Proxy and Web Proxy services
- MiCollab Audio, Web, and Video Conferencing

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

MiCollab can be deployed in all of the same models as the MiVoice Business including as software on an industry standard server and in public and private cloud as a dedicated instance virtual appliance.

6.1.1 Flow Through Provisioning

Flow Through Provisioning

Flow Through Provisioning is a feature that allows an administrator to provision and manage users and services for a network of MiVoice Business servers from the MiCollab User and Services (USP) application. Updates made to the following data are synchronized between the MiCollab and MiVoice Business systems using System Data Synchronization (SDS):

- User and Services DataNetwork Elements
- Network Elements
- Departments and Locations
- Roles and Templates

Both MiVoice Business and MiCollab are updated following any user- or service-affecting changes on either platform. However, although changes made to users, phone services, roles, and templates on a MiVoice Business system are distributed to the other systems (including MiCollab), the recommended best practice is to perform all user and service provisioning from the MiCollab USP application.

Note:

MiCollab services can be added, modified, or deleted only from the MiCollab USP application.

The feature provides single sign-on to the administration interfaces for the Mitel communications network. After you sign into the MiVoice Business System Administration Tool, you are granted Reach Through access to the MiCollab USP application, and vice versa. Direct links allow you to access specific programming forms on one platform from forms on the other platform.

For details and for information on how to configure Flow Through Provisioning, refer to the “MiCollab Installation and Maintenance Guide” on the Mitel Customer Documentation web site.

6.1.2 MiCollab Support

MiCollabis supported as a virtual appliance within the VMware® vSphere™ environment for MiVoice Business. Virtual MiCollableverages VMware VSphere 4.0 or 4.1 to enable businesses to consolidate Mitel’s leading unified communications applications in the data center.

6.1.2.1 MiCollab Audio, Web, and Video Conferencing

MiCollab Audio, Web, and Video Conferencing provides a feature-rich, cost effective IP-based collabo- ration 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. Key benefits of the solution are as follows:

- 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 outsourced conferencing services.
- Easy Management: This is deployed as part of MiCollab.

6.2 MiCollab Unified Messaging

MiCollab Unified Messaging™ is a scalable, integrated voice and fax unified messaging system that uses can access anywhere, anytime. Formerly known as NuPoint it provides access to a host of flexible and customizable applications including Call Director and Speech Auto-attendant. Simple and cost-effective configura tion, 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 use their MiCollab Mobile Client Visual Voicemail capability or call into the application and listen to their voice mails. This simplifies the end user experience and increases productivity.

Mailbox users access these capabilities through the Telephone User Interface (TUI) or Outlook (using the 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.

MiCollab Unified Messaging can be used to:

- Place calls to people/departments quickly and efficiently by speaking their names or phone numbers.
- Notify a mailbox owner when a new voice mail message arrives. It supports SMS notification to cellular phones. SMS notification text-messages users when they receive new voice messages.
- 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 server. The Fax feature works in a network configuration where the server is integrated directly with a MiVoice Business system or with another PBX.
- Perform Mailbox Maintenance, System Maintenance, Report Generation, and Call Director management from a web-based console.
- Control voice mail functions through context-sensitive keys on the phone.

For more information, refer to MiCollab General Information Guide in the Mitel Document Center.

6.3 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.

6.3.1 MiCollab Client

MiCollab Client is a comprehensive unified communications client that integrates presence and avail- ability, secure Instant Messaging (IM), audio conferencing, and video and data collaboration with the call control capabilities of MiVoice Business. MiCollab Client provides a home page that allows users to quickly launch commonly used URLs and directly communicate with their favorite contacts.

MiCollab Client enables also users to manage contact information, determine the presence and availability of colleagues, and set their own call-handling policies at the desktop or on their mobile device.

Figure 18: MiCollab Client

6.3.2 Mobility Solutions

Mitel's Mobility Solutions are described in the following sections.

6.3.2.1 MiVoice Border Gateway Teleworker Service

The MiVoice Border Gateway 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.

MiVoice Border Gateway requires the following components:

Head Office	Remote Site
• Server installed with MiCollab software and the MiVoice Border Gateway software blade or Server installed with Mitel Standard Linux software and the MiVoice Border • Gateway software blade • Static IP address	• 6905, 6910, 6920, 6930, 6940, 6970 • 5304, 5312, 5320, 5324, 5330, 5340, 5360 • MiCollab Soft phone • Home router with Network Address Translation (NAT) and local DHCP • Broadband connectivity (static IP address is not required)

You can configure the MiVoice Border Gateway teleworker service on any of the current Mitel IP sets. Using a simple process, the phone is set to operate in tele worker mode. By following the screen prompts the phone keypad is used to enter the IP address of the MiVoice Border Gateway installed at the head office. The phone can then be used off-site and plugged into any broadband Internet connection. When the phone is powered up, it automatically establishes a secure encrypted connection with the MiVoice Border Gateway 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 MiVoice Border Gateway teleworker service configurations.

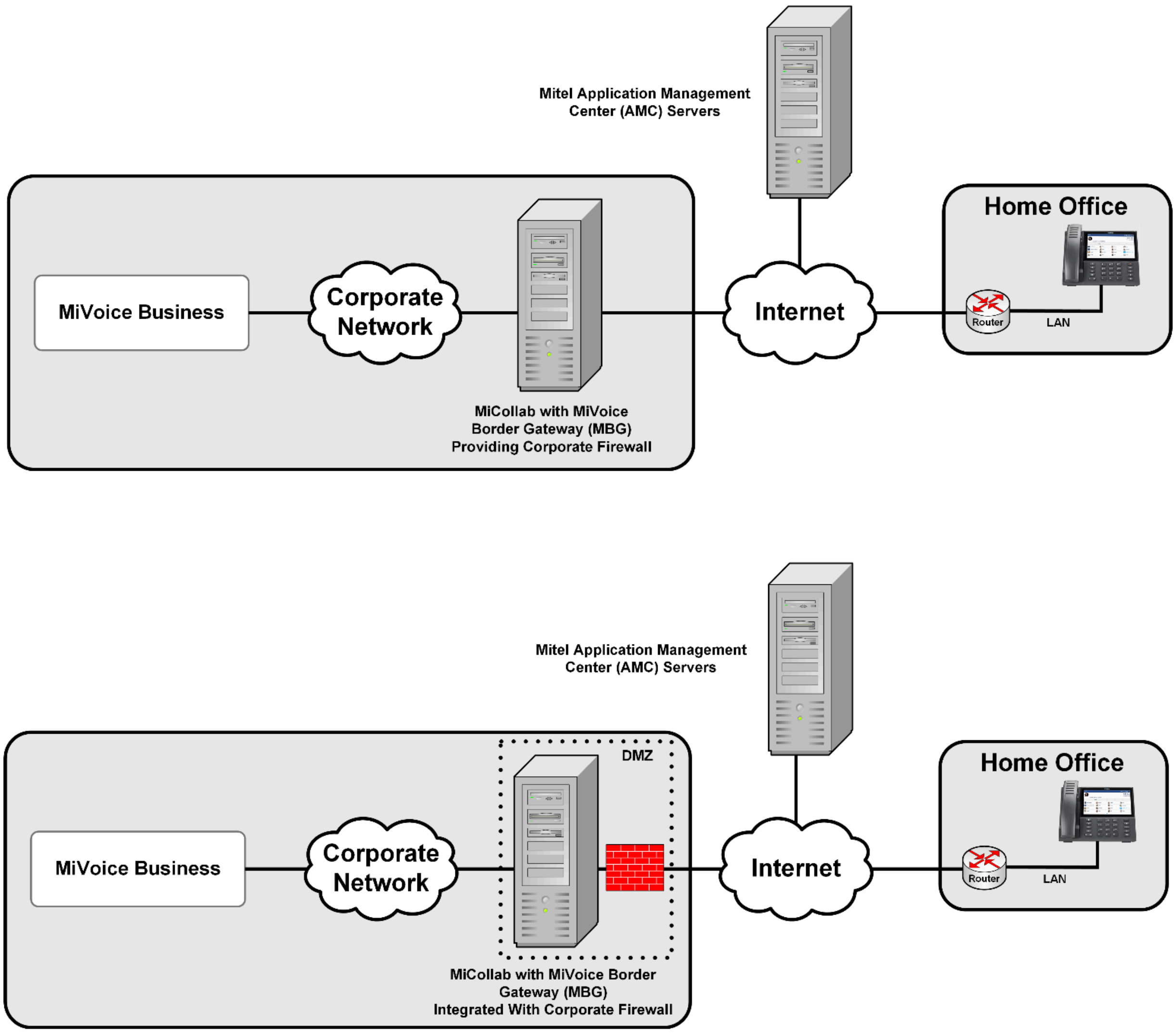


Figure 19: MiVoice Border Gateway Teleworker Service (Internet Facing and DMZ)

Mobile Unified Communications

In addition the MiCollab softphone that is available for mobile or desktop device MiVoice Business, when a user is licensed accordingly, enables users to twin their desk phone with any permitted internal or external PSTN connected phone including their cell phone. Calls arriving at the desk phone can ring the cell phone simultaneously, until one or the other is answered. If calls are unanswered, they are forwarded to the enterprise voice mail. MiVoice Business 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 when combined with MiCollab an integrated mobile and desktop experience.

A user can readily change the device/number that is twinned to their primary extension and take advantage of advanced capabilities delivered with MiCollab and outbound mobile calls can be placed by the PBX, extending Single Number capability and delivering cellular long distance cost savings.

6.3.2.2 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 Cord- less Devices for Mitel 6930 and 6940 IP Phones., Mitel's wireless IP phone devices provide users with the complete range of MiVoice Business features.

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

Complete IP Network integration: When integrated with a MiVoice Business system, provides the complete range of MiVoice Business features

MiVoice Business supports the following DECT devices.

IP-DECT System

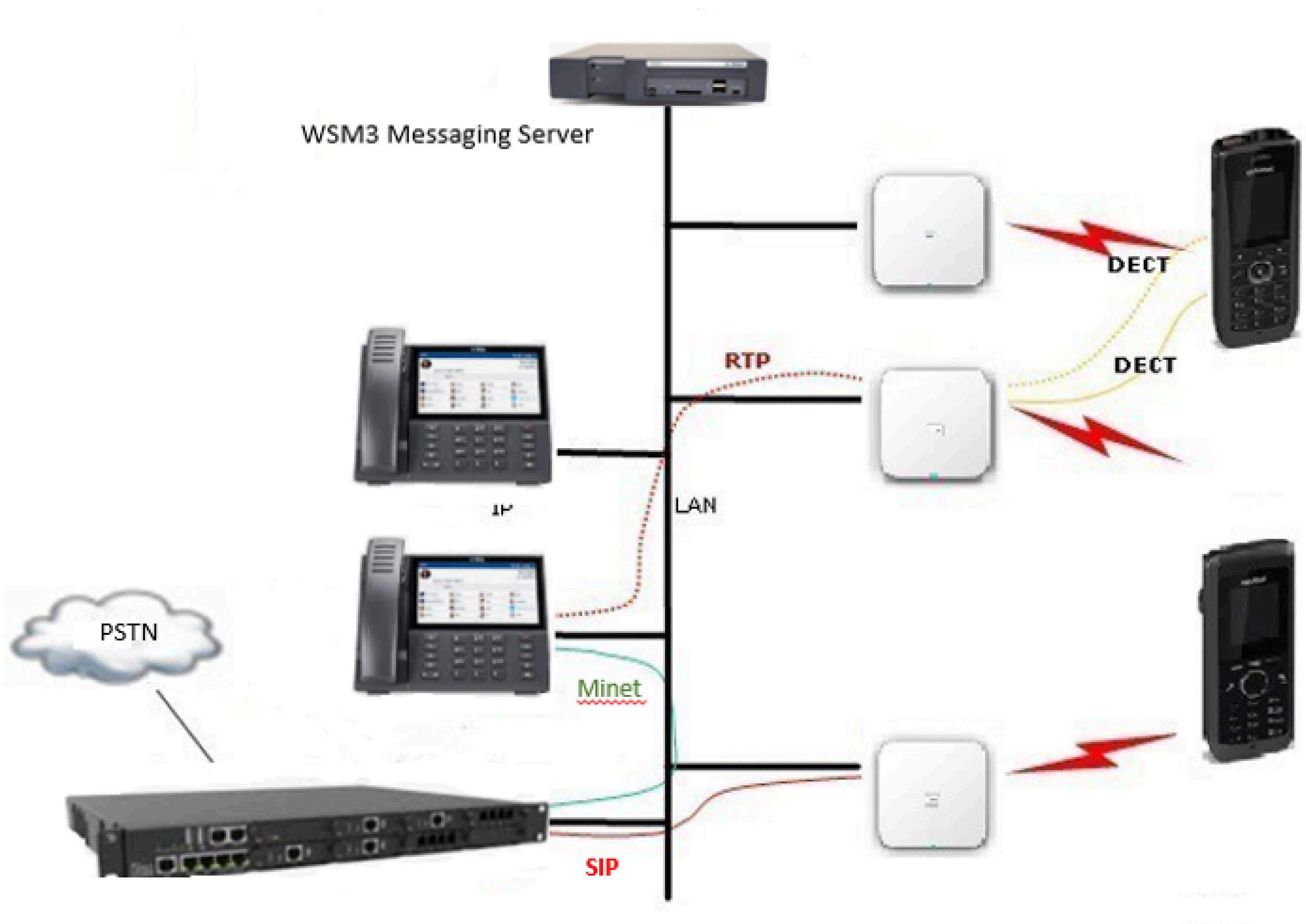
Wireless phone	Wireless infrastructure	Comments
Mitel 5613, 5614 and 5607 Wireless Handsets	IP-DECT	Available globally; integrated over SIP
Mitel 5634 Wireless Handsets	Wi-Fi/802.11	--

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.
- Complete IP Network integration: When integrated with an MiVoice Business system, provides the complete range of MiVoice Business features.

- The Mitel IP-DECT System 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:
  - MiVB Instance (Picture EX system)
  - Base stations for wireless coverage
  - 5613: wireless handset for office environments
  - 5614: wireless handset for healthcare environments
  - 5607: wireless handset for industrial and manufacturing environments
  - Services and Messaging gateway (WSM)
  - Full Range of Accessories

The base stations connect to the MiVB instance through the LAN or MBG and communicate using to the 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 5613 and 5614 handsets are programmed on the MiVB as specific device types.



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.

SIP-DECT Wireless Solution

Wireless phone	Wireless infrastructure	Comments
Mitel 612, 622 and 632 Wireless Handsets	SIP-DECT	Available globally; integrated over SIP

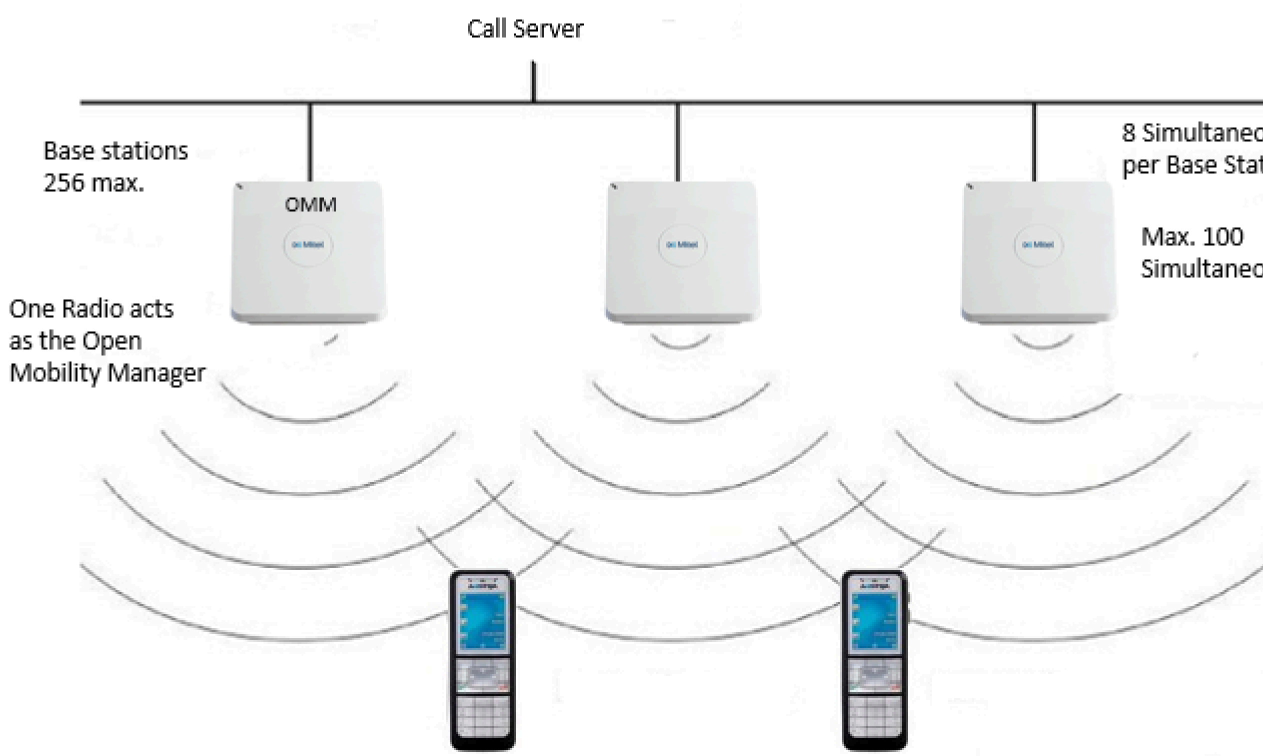
The Mitel Internet Protocol Digital Enhanced Cordless Telecommunications (SIP-DECT) wireless solution is A. It consists of the following components:

- MiVB instance
- Base stations for wireless coverage

- 6x2 SIP-DECT: wireless handset for office/healthcare environments
- 6x2dt SIP-DECT: wireless handset for industrial/manufacturing environments
- Open Mobility Manager (SIP-DECT wireless solution administration application)

The base stations connect to the MiVB instance through the LAN or MBG. 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.

Wireless IP Phones

MiVoice Business supports the following wireless phones:

- 5603/5613, 5604/5614, 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 MiVoice Business.

- Mitel 612, 622, and 632 DECT Handsets: Mitel 600 family of handsets, from the entry-level 612 to the ruggedized 632 and the in-between 622, offer exceptional voice quality and data transmission along with the latest in DECT security standards.

Figure 20: Wireless Phones

6.3.2.3 Mitel StreamLine

Mitel StreamLine delivers Ethernet and Power over Ethernet services over a single pair of telephony-grade wire with four times the reach of traditional data switches. In addition, StreamLine switches come standard with the flexibility of power sharing, load balancing, hot swappable power supply and power sharing among multiple daisy chained units.

Mitel StreamLine ensures you can IP-enable legacy voice environments – quickly, easily, efficiently and cost-effectively. When wiring issues become a barrier to migrating your business communications system to IP, Mitel StreamLine can help to seamlessly integrate heritage building sites with legacy wiring infrastructure, including remote and isolated sites such as warehouses and cruise ships. Experience a solution that delivers no cable replacement, no business disruption, no security risks, no connectivity limits and no wasted

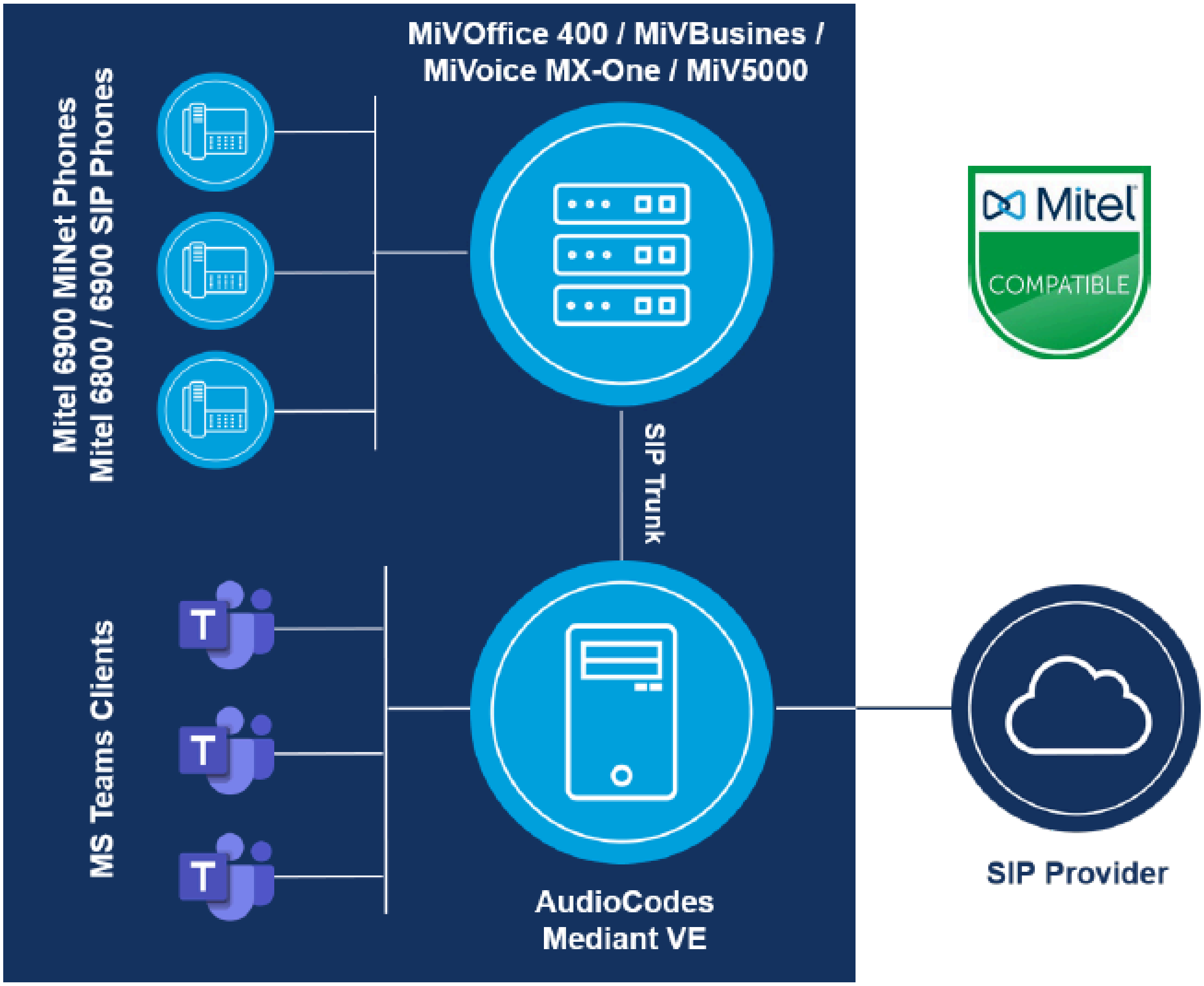
budgets.

Mitel StreamLine extends Power over Ethernet (PoE) to areas not previously reachable, up to 1,200ft (365m), and you'll enable employees to work wherever they want, resulting in more productivity and employee satisfaction. Mitel StreamLine simplifies network design by transforming outdated technology (TDM/PBX, analog/digital endpoints and more) to full IP paths with power. Avoiding two-wire rip/replace eliminates risk and disruption to businesses and networks.

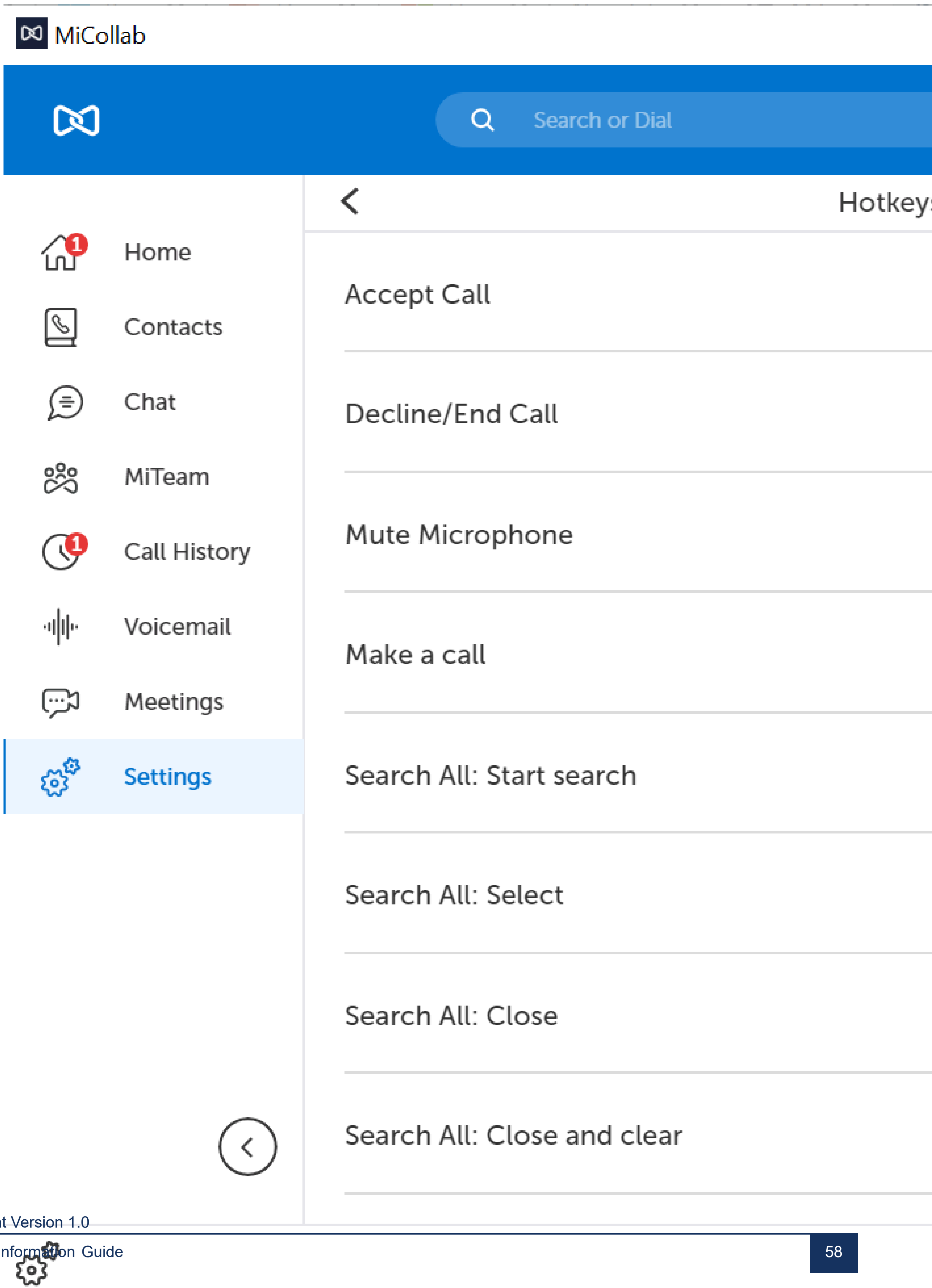
6.3.2.4 Mitel for Microsoft Teams

A customer can make the most of their investment in Microsoft Teams with MiVoice Business by easily combining their Microsoft Teams account with their MiVoice Business having access to the collaboration of Microsoft Teams with Mitel's enterprise telephony features at your fingertips.

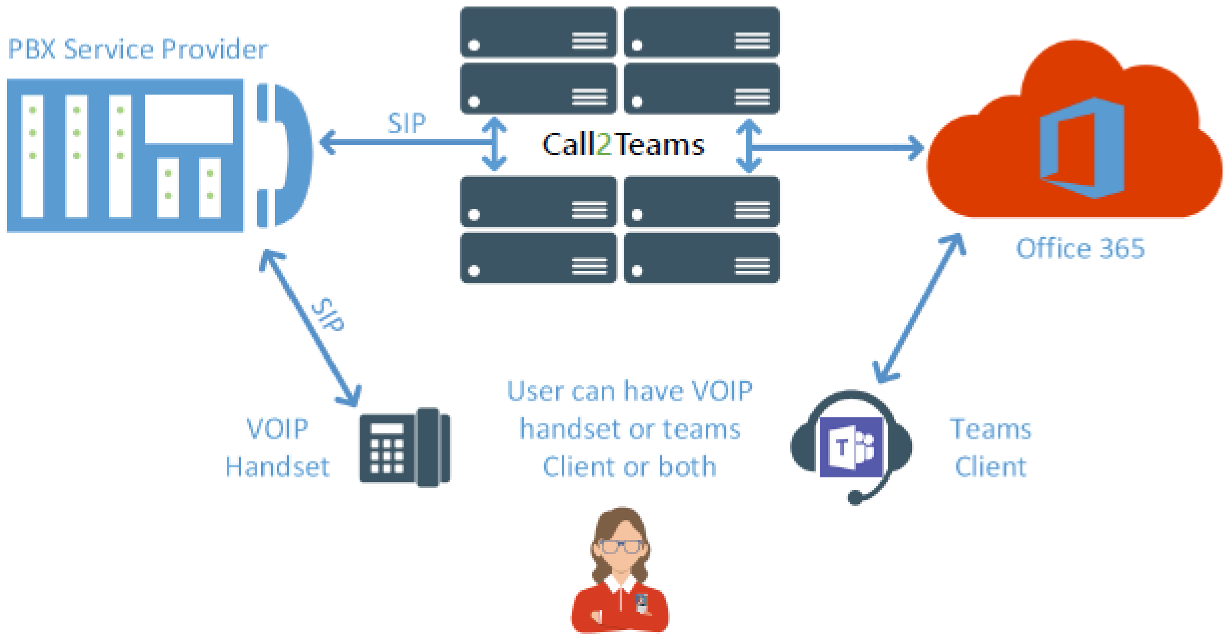
Direct Routing - A complete Mitel telephony infrastructure to Microsoft Teams integration with a certified MSA certified Session Border Controller allowing you to make calls from within the Microsoft Teams environment.



MiCollab Hotkey Dialing - Included with MiCollab, easily set up hotkeys to dial. Select a phone number in Microsoft Teams and click the MiCollab hotkey combination to initiate a MiCollab call to the phone number.


MiCollab
Search or Dial
Hotkeys
Home
Contacts
Chat
MiTeam
Call History
Voicemail
Meetings
Settings
Accept Call
Decline/End Call
Mute Microphone
Make a call
Search All: Start search
Search All: Select
Search All: Close
Search All: Close and clear

Call2Teams – Call2Teams is a third-party solution that integrates a Mitel phone system with Office 365, enabling softphone functionality within the Microsoft Teams environment.



Mitel Assistant - The Mitel Assistant is a bot that allows users to launch a MiTeam Meeting directly in the Microsoft Teams app through 1:1 or group chat. Users can download the Mitel Assistant in the Teams store and follow the bots prompts to get started.

6.3.2.5 Customer Interaction Solutions

6.3.2.5.1 MiVoice Business Automatic Call Distribution (ACD)

Overview

ACD is used to help businesses optimize their resources. It enables businesses to handle large numbers of incoming customer calls and answer these calls with as few trained agents as possible. An ACD system routes incoming calls to the longest idle agent within a specific group. If no agents are available, calls are queued and forwarded to an agent when one becomes available.

Typically, ACD systems have more incoming calls than there are agents available to answer them. This results in callers having to wait for agents to become available. To prevent waiting callers from hanging up and calling competitors, ACD systems play recorded announcements interspersed with music on hold. Recorded announcements set the expectations of waiting callers and reassure them their calls are important to the business. When an agent becomes available, the first caller in the queue is routed to the agent.

Networked ACD extends ACD functions over multiple telephone systems with a networked MiVoice Business environment. Agent groups on the various MiVoice Business telephone systems can answer calls on the network regardless of where the call first entered the network. D-channel signaling is used to queue the calls remotely.

A virtual contact center can evenly distribute calls among agents in a specific agent group, regardless of the agent’s geographic proximity to other agents in their group, or the agent controller to which they are registered.

ACD Resiliency provides seamless reporting in the event of a network or controller outage.

For more information regarding ACD, see the ***MiContact Center Business Solutions Deployment Guide.***

How ACD Works: The ACD Routing Engine

The ACD routing engine involves several basic components:

- Incoming lines or trunks that point to an ACD path
- Paths (queues) that point to an Agents Skills Group
- Agent Skills Groups
- Recorded announcements to greet waiting callers
- Music on hold (to entertain waiting callers between recordings)

On the MiVoice Business platform, the ACD is embedded on the platform itself. Each ACD agent is assigned a unique agent identification (ID) number that is associated with an answering group. The agent ID can be assigned a name in the telephone directory.

Agents can log into any ACD-enabled phone set and receive calls, providing agent mobility. During the agent login, the set assumes the personal profile of the agent, which includes the assigned name, class of service (CoS), class of restriction (CoR), Skills Goup memberships, and Path memberships. Central to ACD functionality is the ACD Path. A number of possible Paths can be defined in the ACD system. The ACD Path is a flexible call-routing method that provides the information required for handling incoming calls. Each Path has a set of parameters that determine how the system handles queued callers, what system resources to use, when the call is to be answered, and which group will answer the call. Based on customer requirements, each call received is directed to a path. Calls are queued for an agent group based on the path priority and the order of arrival at that path. Each ACD path is assigned a priority number. A call to a path adopts the path’s priority, which allows incoming calls to be directed based on their importance and expense.

Calls are routed to the longest idle agent within a group, or optionally, to the longest idle agent with the highest skill level for that group (as programmed on the MiVoice Business platform).

MiContact Center Business

Mitel MiContact Center Business integrates with MiVoice Business platform to provide contact centers with the tools they need to efficiently and effectively measure and manage contact center operations. Interactions are routed intelligently across all media types (voice, email, chat, SMS, and open media), increasing customer satisfaction and streamlining agent calls. Contact center efficiency and agent performance can be monitored both historically and in real time and can be measured using a wide variety of reports. Flexible licensing packages provide access to specific features and applications and enable you to build a contact center package that best suits your business needs.

Mitel MiContact Center Business is designed for organizations that need to process incoming calls in a formal way and require advanced capabilities. MiContact Center Business is a highly available scalable, resilient, dedicated instance solution for sophisticated contact centers of all sizes across one or more locations that supports advanced integrations, extensive reporting and sophisticated routing. MiContact Center Business provides:

- An award-winning graphical agent desktop
- A comprehensive set of historical and real-time reports
- Consolidated agent and queue management
- Rich voice Skills-Based Routing automatic call distribution (ACD) functionality
- Contact center management tools
- Contact center scheduling for automatic agent scheduling based on business rules and required skills
- Optional schedule adherence to verify agents are adhering to their schedules
- Real-time agent and queue control
- Call accounting and non ACD General Business, Hunt Group and Ring Group reporting
- 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 MiVoice Border Gateway teleworker service
- 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

All MiContact Center Business solutions are IP based, enabling customers to manage their contact centers from anywhere, anytime with an Internet connection. MiContact Center Business enhances ACD functionality and stores information in the industry-standard SQL database format.

Contact center solutions are described in detail in the MiContact Center Business General Information Guide.

MiVoice Analytics

MiVoice Analytics is comprised of two license levels: Call Accounting – for historical call costing, subscriber services, and traffic analysis reporting, and Business Reporter – for general business extension reporting.

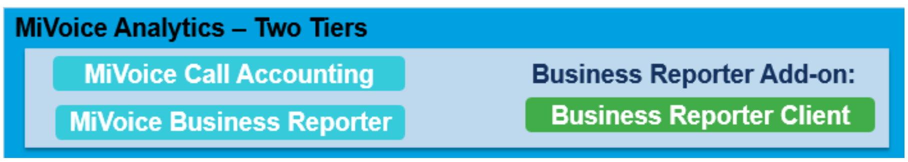


Figure 21: MiVoice Analytics Licensing Overview

Call Accounting is an entry level product that no longer includes real time monitors in the Contact Center Client (real-time call costing is unlocked with a full Business Reporter license). Call Accounting provides call costing and subscriber services for current day and historical reporting and continues to only be supported in North America, Latin America, United Kingdom and the Benelux region.

Business Reporter includes Call Accounting for the supported regions and adds to it real time and reporting for ring groups. Business Reporter Client replaces the existing Integrated Client license, PhoneSet Manager, and Screen Pop license options that were available in previous releases. These are primarily used for general business screen pop requirements. The Business Reporter Client license does not include a softphone license. Softphone licenses for Business Reporter are sold separately as an add-on license.

Both Call Accounting and Business Reporter extension licenses are named licenses and not concurrent.

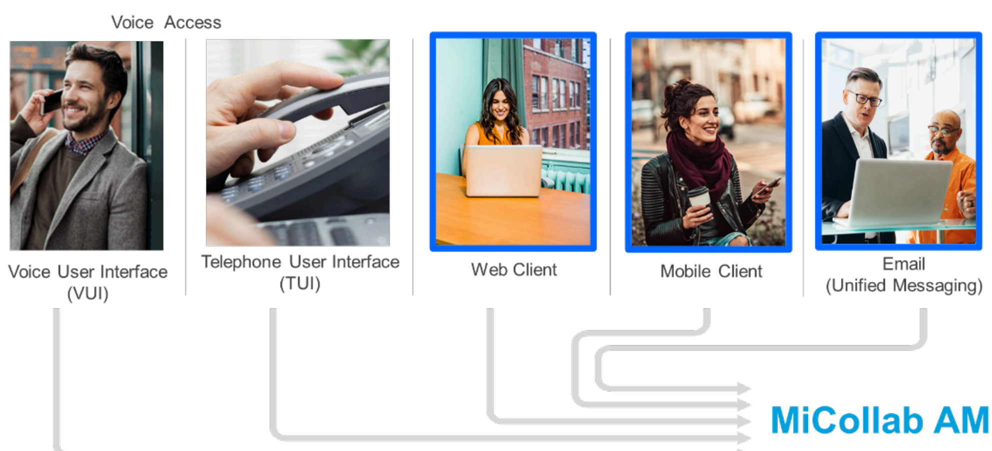
Mitel Interaction Recording

Mitel Interaction Recording (Powered by ASC) suite captures, saves and archives multiple communication channels including mobile voice, video, and chat for financial institutions, contact centers and public safety organizations. The recording suite provides you with communications recording and quality management as a service whereby capacities and features can be added as needed to react quickly and grow in the long-term. The solution offers the following capabilities.

- State-of-the-art recording and analysis for complex infrastructures
- Systematic capture and assessment of customer communications
- Solutions for financial institutions, contact centers and public safety organizations.
- Compliance with the highest security requirements and regulations such as MiFID II

6.3.2.6 MiCollab Advanced Messaging

MiCollab Advanced Messaging provides a powerful suite of Unified Messaging (UM) applications including advanced call processing, voice mail, e-mail integration, fax, speech and notification options. MiCollab AM can be ordered as a stand-alone add-on application for MiVoice Business and provides an alternative UM solution to the inbuilt MiCollab UM. Providing additional integration options, enhanced scaling and redundancy options.



Additionally, the MiCollab AM solution comes with multi-language TTS option for converting e-mail to speech as well as comprehensive ASR engine option to enable speech driven personal assistant capability and to allow inbound callers to navigate the auto-attendant capability using speech commands.

MiCollab AM offers strong scalability starting from 50 users and 4 ports up to 60000 and 800 ports spread out between multiple instances in a single system. The standard number of ports are automatically calculated based on the number of basic users and included with the user licenses.

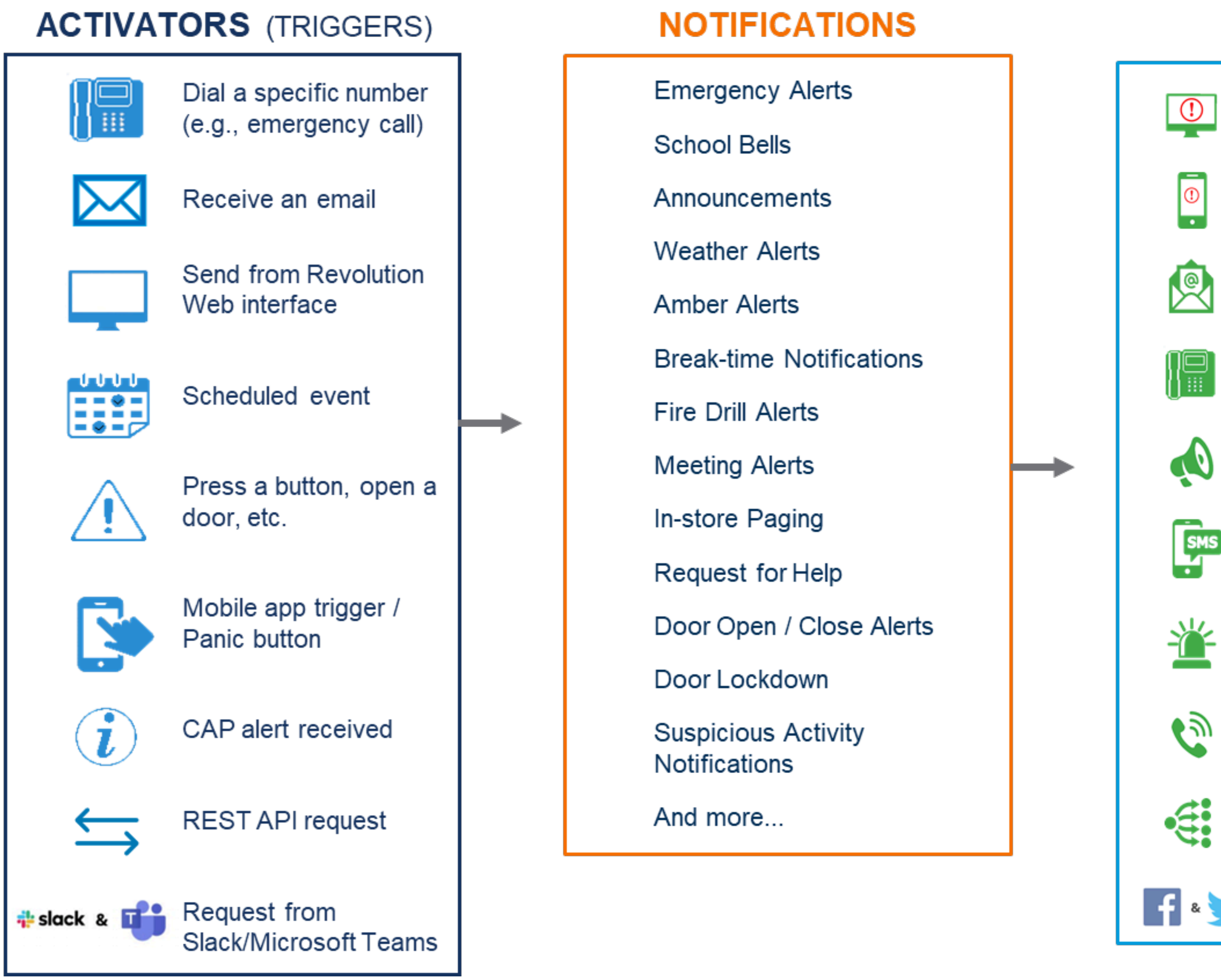
A single system can be distributed over 20 different locations with multiple “call servers” (handling VM/AA traffic) located in multiple sites combined together with the centrally located “system server” used as the “message store” and handling administration tasks.

Additionally, MiCollab AM offers exceptional redundancy options in a bare metal infrastructure using distributed call servers handling the traffic as well as never fail for site redundancy and disaster recovery of the back-office message store (System Server) to meet the requirements of even to most stringent customers.

6.3.2.7 Mitel Revolution

Mitel Revolution is built for quick and reliable mass notification in the modern mobile-centric world. With support for initiating alerts to on-premises, off-premise and mobile devices through emergency, pre-set, or location-based triggers, Mitel Revolution ensures that everyone will be notified, whether they are in the office or on the move. Built upon a fault-tolerant and adaptable architecture, Mitel Revolution can be tailored to address your critical or routine notification needs. Unicast to multicast technology removes the need for network multicast servers to make deployment simple and cost-effective. Analytics gives administrators quick access to insight into the health of your system, enabling corrective actions to be taken immediately, if required.

Mitel Revolution - How It Works



Key Features:

- Media-rich notifications to on and off-premise devices
- Support 22,000+ endpoints
- Geo-fencing for targeted location-based alerts
- Activate alerts directly from iOS or Android mobile application
- Desktop Override option prevents users from closing alerts until the event is terminated
- Unicast to multicast technology removes the need to deploy multicast servers throughout the entire network

- Integrations to external CAP-enabled feeds, such as National Weather Service and AMBER Alert
- Real-time notification status, analytics and system health reports
- Extend beyond out-of-the-box integrations with built-in REST & Device APIs, plus Mitel CloudLink

This functionality is added to the existing Emergency Services feature offered by MiVoice Business. The switch performs the actual routing of emergency calls to dispatch emergency personnel (for fires, medical emergencies, etc.)

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.

Mitel OIG

The Mitel Open Integration Gateway (OIG) is a web server that provides a single access point to web services available within a Mitel system. The Mitel OIG runs on the Mitel Standard Linux (MSL) operating system and can be deployed as an MSL software blade through the Mitel AMC licensing server, or as a virtual appliance. The Mitel OIG provides web services by integrating with a Mitel system (MiVoice call manager cluster or single MiVoice call manager node and Mitel applications). Mitel OIG supports the following web services:

- Session management service
- Call control services
- Data access services

Software developers are required to join the Mitel Solutions Alliance (MSA) developer partner program at one of the Developer Advanced membership levels in order to develop or modify Mitel OIG-based applications.

MiAUDIO

With MiAUDIO, developers can include the processing of phone audio streams in their applications for MiVoice Business. Examples of MiAUDIO applications include a voice mail system, or an automated recorded message delivery system.

Software developers are required to join the Mitel Solutions Alliance (MSA) developer partner program at one of the Developer Advanced membership levels in order to develop or modify MiAUDIO-based applications.

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 MiVoice Business, 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 OIG

Secure Recording Connector

Mitel Secure Recording Connector (SRC) is a call recording solution that enables call recording equipment to record Mitel encrypted voice streams. The 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 MiVoice Border Gateway software blade. Phones that are enabled for call recording register with MiVoice Business via the SRC. The 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.

The SRC supports Mitel Interaction Recording. Software developers are required to join the Mitel Solutions Alliance (MSA) developer partner program at one of the Developer Advanced membership levels in order to develop or modify SRC-based applications.

MiNET Phones XML API

Mitel MiNET phones support an XML API allowing external applications to control the display of the phone as well as its configuration. The XML API for Mitel 69xx MiNet Phones Development Guide describes the details of the XML objects and how to implement them and the XML Development Toolkit supports application developers implementing applications for Mitel's MiNET phones.

The XML browser in Mitel 69xx IP phones allows developers to create custom services that they can use via the phone's keypad and display. These services include things like weather and traffic reports, contact information, company info, stock quotes, or custom call scripts. The XML applications are hosted by one or multiple Web servers which will serve as a proxy to either other applications or to Internet Web servers.

Mitel Performance Analytics

Mitel Performance Analytics provides fault, inventory, and performance management for Mitel Networks Unified Communications systems, multiple enterprise VoIP systems and associated network infrastructure, both LAN and WAN. Mitel Performance Analytics supports monitoring and remote access both for private networks, such as enterprise LANs and MPLS VPNs, and for public network or Internet-reachable devices, such as access routers.

MiTeam Meetings

MiTeam Meetings application is Mitel's Cloud-based collaboration tool (based on CloudLink infrastructure and hosted in AWS) that provides MiCollab users with the ability to initiate Mitel Meetings from their MiCollab Client. With MiTeam Meetings you can:

- Manage collaboration meetings
- Hold chat sessions and receive chat notifications
- Store and share files
- Perform audio, video, and web sharing

MiTeam Meetings is supported with the following MiCollab Clients:

- MiCollab for PC Client
- MiCollab for Mac Client
- MiCollab Web Client
- MiCollab for Mobile Client (iOS or Android)

For information about MiTeam Meetings end-user features, see MiCollab Client End-User Online Help.

6.4 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.

6.4.1 Feature Support Matrix

The following tables summarize the features provided by 5300 and 6900 series of MiVoice IP Phones.

6.4.1.1 6900 Series

The MiVoice 6900 series is a family of powerful ‘Mobile First’ IP phones offering advanced integration with mobile phone calls and applications. Mitel’s MobileLink capability enables the user’s mobile phone to pair directly with the 6900’s Bluetooth interface to deliver access to mobile phone features from the desk phone allowing both cellphone and IP calls to be managed from a single device. MobileLink allows mobile phone users to leverage the exceptional HD audio and comfortable ergonomics of the 6900 series phones for both IP and cellphone calls. The 6900 phones deliver crystal clear audio through a unique corded or cordless voice optimized handset and high performance hands-free speakerphone. Unparalleled flexibility is achieved through a broad array of add-on user installable accessories that enable the phones to be tailored to specific user needs. The 6900 family provides the flexibility and capability needed to meet the demanding needs of today’s users. Added to the this are the new “T” editions that are first-of-their-kind IP phones built using antimicrobial treated plastics.

The 6905 IP Phone is a single line phone that provides wideband audio quality with full-duplex speaker phone and a 2.75 inch 128x48 pixel non-backlit display. Provides three paper labeled programmable hard keys. The 6905 is equipped with built-in two-port, 10/100 Fast Ethernet switch that enables direct or shared network connection. The phone supports connection for a wideband HD handset.

Figure 22: XX Mitel 6905

The 6910 IP Phone is a multi-line phone that provides enhanced wideband audio with full-duplex speak- erphone and a 3.4 inch 128x48 pixel backlit LCD display. Provides up to 8 programmable multi-function keys with LEDs. The 6910 is equipped with built-in two-port, 10/100/1000 Gigabit Ethernet switch that enables direct or shared network connection. The phone supports connection for DHSG/EHS or analog headsets and a wideband handset.

Figure 23: XX Mitel 6910

The **6920** IP phone is designed from the ground up for the enterprise user who requires an exceptional HD audio experience via its unique voice optimized handset. It offers a large color LCD display, dual GigE, 18 programmable personal keys, 4 context-sensitive soft keys, support for both USB & Analog headsets and support for MobileLink via the optional USB Bluetooth Dongle.

Figure 24: XX Mitel 6920

The MiVoice **6930**IP phone commands the desktop with its large 4.3” color display, powerfulcrystal clear HD audio through the voice optimized handset and 72 programmable personal keys, Bluetooth 4.1 interface with MobileLink mobile integration, mobile phone USB charging point and choice of expansion modules makes the 6930 the choice of power users.

Figure 25: XX Mitel 6930

The **6940** IP Phone is designed for executive users who demand a lot from their phone. The 6940 offers a large 7” touch display, powerful crystal clear HD audio through a unique cordless voice optimized handset and 96 programmable personal keys. Mobile Link mobile integration, Dual Gigabit Ethernet ports and the full-duplex speaker phone ensure the 6940 delivers a robust, productivity-enhancing executive desktop communication tool.

The standard cordless voice optimized handset allows users to enjoy clearly discernible conversations in a variety of environments without being physically tied to their desk phone.

Figure 26: XX Mitel 6940

The Mitel **6970** IP Conference Phone designed to make meetings easier and more efficient. A large 7-inch color touch screen grants excellent visibility to an intuitive user interface for quick navigation to essential meeting information and functions. A tight integration with Mitel platforms, applications, call managers delivers a unique conference experience that will not be found with 3 rd party devices. Built-in Bluetooth 4.1 and MobileLink grant you the ability to seamlessly pair with Bluetooth enabled audio devices and expand the capabilities of your mobile phone. Enjoy crystal clear audio with high definition speakers and 360° beam-forming microphones. With the Mitel 6970 IP Conference Phone, your entire meeting experience will be effortless.

Figure 27: XX Mitel 6970

MobileLink support on the 6930, 6940 and 6970 provides seamless mobile integration using Bluetooth wireless technology. With MobileLink users can:

- Sync the mobile phone's contact and call log list with the 6930 or 6940 IP Phone.
- answer a mobile phone call using the 6930 or 6940 IP Phone.
- switch audio from the IP Phone to the mobile phone and back again.

PCLink enables seamless handling of both phone calls and PC audio through a single easy-to-use device.on the 6930, and 6940 and 6970 and turns your phone into a high-quality audio device for PC based video collaboration.

6900 Series

The following table summarizes the 6900 series phone features:

Physical	6905	6910	6920	6930	6940
Desk/Wall Mountable	Yes	Yes	Yes	Yes	Yes
Desk/Wall Mount Stand	Only desk stand included	Only desk stand included	Only desk stand included	Only desk stand included	Only desk stand included

Physical	6905	6910	6920	6930	6940
Length of handset cord	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
Ethernet Cable (2 meters/7 feet)	Optional	Included	Included	Included	Included

Physical	6905	6910	6920	6930	6940
Compression Support	G.711, G.722, G.722.1,G.729 A	G.711, G.722, G.722.1,G.729 A	G.711, G.722, G.722.1,G.729 A	G.711, G.722, G.722.1,G.729 A	G.711, G.722, G.722.1,G.729 A
Voice QoS (802.1 p/q)	Yes	Yes	Yes	Yes	Yes
Encryption	128 bit AES**	128 bit AES**	128 bit AES**	128 bit AES**	128 bit AES**
802.1x Support	Yes - EAP PEAP	Yes - EAP PEAP	Yes - EAP PEAP	Yes - EAP PEAP	Yes - EAP PEAP
CLASS BSupport	Yes	Yes	Yes	Yes	Yes
Headset Jack	No	Yes	Yes	Yes	Yes

Physical	6905	6910	6920	6930	6940
Peripherals Support	No	No	Yes	Yes	Yes
USB Ports	No	No	1 (powered)	1 (powered)	1 (powered)
**Advanced Encryption Standard					

Powering Options	6905	6910	6920	6930	6940
Ethernet / AC Power Adapter Support (48 VDC LAN Power)	Yes	Yes	Yes	Yes	Yes
802.3af Power over Ethernet Compliant	Yes	Yes	Yes	Yes	Yes

Powering Options	6905	6910	6920	6930	6940
Power Consumption (Worst Case Maximum)	2.2W	2.4W	3.8W	8W	9.9W
with one M695 PKM	N/A	N/A	6.1W	10.3W	12.2W

Powering Options	6905	6910	6920	6930	6940
with two M695 PKMs	N/A	N/A	8.4W	12.6W	14.5W
with three M695 PKMs	N/A	N/A	17 W10.8W	15W	17W
* 10/100 MB Mode values. GB Mode values: Idle - 4.8W; Typical - 8.6W; Maximum - 9.2W					

Display	6905	6910	6920	6930	6940
Color	No	No	Yes	Yes	Yes
Size (inches)	2.75	3.4	3.5	4.3	7.0
Number of pixels (w x h)	128x48	128x48	320x240	480x272	800x480
Pixel Size	0.43x0.43mm	0.43x0.43mm	0.43x0.43mm	0.43x0.43mm	0.43x0.43mm
Illumination	Non-backlit Display	Backlit	Reflective Backlit White	Reflective Backlit White	Reflective Backlit White
Touch Interface	No	No	No	No	No
Contrast Adjust	Yes	Yes	Yes	Yes	Yes

Display	6905	6910	6920	6930	6940
Display (soft) Keys	Yes	Yes	Yes	Yes	Yes
Auto Dimming	Yes	Yes	Yes	Yes	Yes
Backlight Off Capability	Yes	Yes	Yes	Yes	Yes
Chinese Character Support	No	No	No	No	No

Function Keys	6905	6910	6920	6930	6940
Number of Programmable Feature/Line Appearance Keys	4	8	18	72	96
Fixed Feature Keys	8	13	10	10	10
Softkeys	3	3	4	5	6
Multiline	No	Yes	Yes	Yes	Yes
Hold	Yes	Yes	Yes	Yes	Yes
Redial	Yes	Yes	Yes	Yes	Yes

Function Keys	6905	6910	6920	6930	6940
Cancel	Yes	Yes	Yes	Yes	Yes
Volume Up/ Down Keys	Yes	Yes	Yes	Yes	Yes
Ringer Up/ Down Keys	Yes	Yes	Yes	Yes	Yes
Message Key	No	Yes	Yes	Yes	Yes
Transfer/ Conference Key	Yes	Yes (softkey)	Yes (softkey)	Yes (softkey)	Yes (softkey)
Call Forward (On/Off) Key**	Yes (definable)	Yes (definable)	Yes (definable)	Yes (definable)	Yes (definable)
Call Me Back Key	Yes (softkey)	Yes (softkey)	Yes (softkey)	Yes (softkey)	Yes (softkey)
Phonebook/ Contacts Directory Key	Yes (Phonebook softkey)	Yes	Yes	Yes	Yes
Microphone Key	No	No	No	No	No
Mute Key	Yes	Yes	Yes	Yes	Yes

Function Keys	6905	6910	6920	6930	6940
Speaker Phone	Yes	Yes	Yes	Yes	Yes
Program/ Superkey†	Yes (definable)	Yes (definable)	Yes (definable)	Yes (definable)	Yes (definable)
Desktop User Tool	Yes	Yes	Yes	Yes	Yes
*Accomplished using End Call softkey or equivalent fixed-function key. ** Call Forward Always only. † Must be programmed from ESM Tools					

Indicators	6905	6910	6920	6930	6940
Feature/Line Appearance LEDs*	N/A	Red	Red	Red	Red
Message Waiting LED	Red	Red	Red	Red	Red
Hold*	N/A	N/A	N/A	N/A	N/A
Hold Button*	N/A	N/A	N/A	N/A	N/A
Line LED Color*	Red	Red	Red	Red	N/A
Ringer LED	Red	Red	No	No	No

Indicators	6905	6910	6920	6930	6940
Microphone/ M ute LED	Red	Red	Red	Red	Red
*Not applicable. Colored icons in combination with flash rates are used to convey call and feature states.					

Acoustic Functions	6905	6910	6920	6930	6940
Ringing Volume Adjust	Yes	Yes	Yes	Yes	Yes
Handset Volume Adjust	Yes	Yes	Yes	Yes	Yes
Handsfree Speakerphone	Yes	Yes	Yes	Yes	Yes
Handsfree: Half Duplex Full Duplex	Full Duplex	Full Duplex	Full Duplex	Full Duplex	Full Duplex

Acoustic Functions	6905	6910	6920	6930	6940
Wideband Audio Hardware	Yes	Yes	Yes	Yes	Yes
On-Hook Dialing	Yes	Yes	Yes	Yes	Yes

Acoustic Functions	6905	6910	6920	6930	6940
On-Hook Call Announce (Paging Receive Capability)	Yes	Yes	Yes	Yes	Yes
Off-Hook Call Announce	No	No	Yes	Yes	Yes
Multicasting Capable	Yes	Yes	Yes	Yes	Yes
Amplified Receive > 12 dB	Yes	Yes	Yes	Yes	Yes
Hearing Aid Compatible (HAC)Handset	Yes	Yes	Yes	Yes	Yes
Ring Warble/Pitch Adjust/Ring Tones	Ring Tones (10)	Ring Tones (10)	Ring Tones (10)	Ring Tones (10)	Ring Tones (10)

System Software Requiremer	6905	6910	6920	6930	6940
MiVoice Business	MiVoice Business 9.0 SP3 or later	MiVoice Business 9.0 SP3 or later	MiVoice Business 8.0 or later	MiVoice Business 8.0 or later	MiVoice Business 8.0 or later

System Software Requiremen	6905	6910	6920	6930	6940
MiVoice Border Gateway telewroker service	Release 11.0 or later	Release 11.0 or later	Release 9.4 or later	Release 9.4 or later	Release 9.4 or later

6900 Series Sets Summary

Mitel 6905 IP Phone
- 2.75-inch Display
- 3 Programmable Keys
- 3 State-Sensitive Softkeys

Mitel 6910 IP Phone
- 3.4-inch Display
- 8 Programmable Keys
- 3 State-Sensitive Softkeys

Mitel 6920 IP Phone
- 3.5-inch Color Display
- 18 Programmable Keys
- 4 State-Sensitive Softkeys

Mitel 6930 IP Phone
- 4.3-inch Color Display
- 72 Programmable Keys
- 5 State-Sensitive Softkeys
- Optional Bluetooth Handset

Mitel 6940 IP Phone
- 7-inch Color Display
- 96 Programmable Keys
- 6 State-Sensitive Softkeys
- Bluetooth Handset

Mitel 6970 IP Conference Phone
- 7-inch Color Display
- 96 Programmable Keys
- 6 State-Sensitive Softkeys
- Embedded Bluetooth

Mitel 69xx IP Phone Variants
- Mitel 6920t IP Phone – protected with antimicrobial technology and smooth Handset
- Mitel 6930t IP Phone – protected with antimicrobial technology and smooth Handset
- Mitel 6930L IP Phone – Bluetooth removed
- Mitel 6930Lt IP Phone – protected with antimicrobial technology and smooth Handset. Bluetooth removed.

Figure 28: 6900 Series Desktop Application Phones

6.4.2 IP Phone Accessories

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

Accessory	6905	6910	6920	6930	6940
Mitel M695Programmable Key Module (PKM)	No	No	Yes	Yes	Yes

Accessory	6905	6910	6920	6930	6940
Integrated DECT Headset	No	No	No	Yes	Yes
Line Interface Module	See Note 3		Yes	Yes	Yes

* Supports embedded Gigabit

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

6.4.2.1 Mitel M695 Programmable Key Module (PKM)

The M695 PKM is designed to increase the power and flexibility of the Mitel MiVoice 6900 Series IP phones. Featuring a 4.3" 480x272 pixel color backlit LCD display and 28 programmable softkeys with LEDs, the M695 can be used with the 6920, 6930, and 6940 IP phones to create a powerful, feature rich console option.

The M695 PKMs can be daisy-chained with up to three additional M695 PKMs, all sharing power and signaling with the Mitel MiVoice 6900 Series IP phones. Designed for receptionists, administrative assis- tants, call center agents, power users, and executives who need to monitor and manage a large volume of calls on a regular basis, the M695 PKM provides an intelligent choice for all enterprise IP environments.

Figure 29: M695 Programmable Key Module with 6930

6.4.2.2 Mitel Headset Options

Mitel DECT Headset

Mitel Headsets

The Mitel Integrated DECT Headset physically attaches to 6930 & 6940 IP Phones to reduce desktop clutter and eliminate need for additional headset power adapter. The Mitel Integrated DECT Headset gives users the ability to untether themselves from their desk and take advantage of the added productivity that wireless communication delivers. Adding the benefit of completely hands-free communication, the Mitel Integrated DECT Headset enables users to work on their computers, handle documents, or take notes during calls. The unique phone-attached headset base reduces desktop clutter and solution footprint when compared to standalone cordless headset solutions. In addition, the fact that the base is powered by the phone eliminates having to find an available power outlet.

The Integrated DECT Headset delivers an extended range of up to 300 feet (100 meters) of personal area mobility, helping users avoid missed calls while stepping away to the printer, copier or a colleague's office. Home based users will enjoy the freedom to roam throughout the house without having to worry about missing calls. Mitel's DECT Accessories are an ideal fit for all organizations and verticals including call centers, education, healthcare, hospitality and retail environments.

Mitel’s headset family delivers a range of high-quality headsets from industry leading headset manufacturer, Jabra to address a range of softphone and desk phone-based user needs. Whether working from a home office or as an agent in a busy contact center, Mitel has a headset that will satisfy what’s needed to effectively tackle the task at hand.

H10

- The H10 is a corded stereo headset with an advanced digital chipset and three strategically placed noise cancelling microphones, for less background conversation noise on your calls
- The H20 has an extremely lightweight and unobtrusive design resulting in lowered agent fatigue and more productive calls.

H30

- The H30 provides a unique, 3-microphone system with intelligent noise-cancellation that filters out background noise and breathing sounds, giving callers a superior experience.
- The H40 provides superior wireless connectivity to a range of up to 150 meters / 490 feet with no loss in connection quality. It supports advanced noise canceling microphone and enhanced stereo speakers deliver crystal clear calls.

Mitel S720 BT

The MiVoice S720 Bluetooth Speakerphone is the perfect solution for office side table and small meeting room conference calls. The speakerphone's functionality is tightly integrated with the MiVoice 6930 & 6940 IP Phone's built-in Bluetooth 4.1 interface to expand its capabilities beyond that of any other third- party Bluetooth device, making it the ideal Bluetooth speakerphone for the 6900 IP phone series.

Figure 30: Figure XX S720 BT

- Coverage for up to 6 people: 360° microphone pickup makes for clear conversations during office side-table or small meeting room conference calls.
- Enjoy crystal clear sound: Always understand conference call conversations thanks to the crystal-clear wideband HD Voice audio.
- Unified audio controls: Synchronized call audio controls allow you to modify the call audio across all Mitel devices with one touch.
- Use all day without recharging: With up to 15 hours of talk-time users can benefit from a battery whose charge will last for your whole working day.

Mitel Wireless LAN Adapter

The Mitel Wireless LAN Adapter that adds wireless connectivity to Mitel MiVoice 6900 Series IP Phones. It allows the Ethernet-enabled phone to join a secure, high-speed network.

The Mitel Wireless LAN Adapter is designed to communicate in the 2.4 GHz and 5 GHz bands. Radio interference in the commonly used 2.4 GHz band can be avoided by utilizing 5 GHz.

Feature highlights include:

- Dual band IEEE 802.11a/b/g/n support:
    - Gigabit Ethernet support: The wired LAN port supports 10/100/1000BASE-T (auto-recognition).

The Mitel Wireless LAN Adapter is easy to set up using the enclosed network setup cable. No special drivers or software are required.

The Mitel Wireless LAN Adapter supports the following security functions:

- WEP (64 Bit/128 Bit)
- WPA-PSK (TKIP/AES)
- WPA2-PSK (AES)
- IEEE 802.1X EAP-PEAP, EAP-TLS, EAP-TTLS, EAP-FAST, EAP-LEAP

6.4.2.3 Mitel MiVoice Business Console

The MiVoice Business Console is a completely PC-based call-handling setup with an intuitive graphical user interface for the department or office attendant.

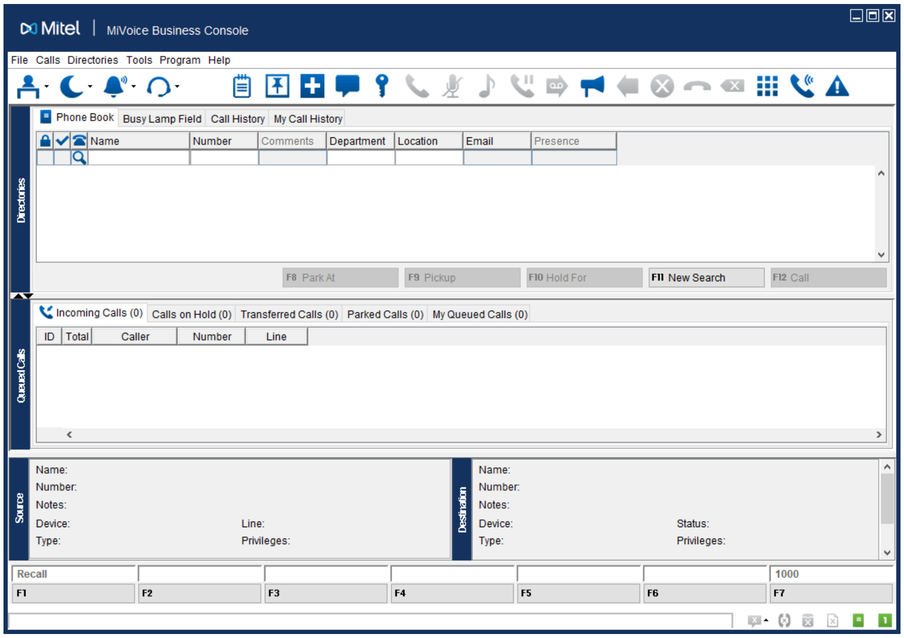


Figure 31: MiVoice Business Console

The Mitel MiVoice Business Console is the evolution of the 5550 IP Console to a software only solution without the adjunct telephony keyboard and handset. Instead, the MiVoice Business Console uses a USB headset and handset for audio and a standard PC keyboard for call handling and feature operation. Using an intuitive user interface, the MiVoice Business Console offers quick access to call processing and telephony features – all from your PC. With teleworker support, the corporate answering point is flexible and mobile. Attendants can now work from anywhere, at any time.

6.4.2.4 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 MiVoice 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 32: 5540 IP Console

6.5 Features

Features of MiVoice Business

The following table details MiVoice Business 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	Supported While Set on 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 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

Feature Name	Description	Supported While Set on Secondary Controller?
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	Yes

Feature Name	Description	Supported While Set on Secondary Controller?
	Call Completion field of the SMDR log.	
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
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 membAudit er 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

Feature Name	Description	Supported While Set on Secondary Controller?
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.	N/A
ACD Extended Agent Skill Groups	When this option is enabled, the maximum number of agent skill groups increases to 128. Each group can support up to 500 agents.	N/A
ACD 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

Feature Name	Description	Supported While Set on Secondary Controller?
ACD Real Time Event	Real time event records are used to monitor and record the activity of the ACD operation. Events are divided into two groups: call events and group statistics events. Call events report on individual ACD agent activity. Group statistics report on ACD group activity such as number of calls queued, longest waiting call, and number of active agents.	N/A
ACD Silent Monitor	Allows a supervisor to listen to an agent's 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	Yes

Feature Name	Description	Supported While Set on Secondary Controller?
	feature can also be used to monitor non-ACD sets, including ONS, SIP, and external Hot Desk user sets.	
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 traveling class marks across T1/D4 trunks	Yes
Advanced ARS	Allows you to program day and time zones,	N/A

Feature Name	Description	Supported While Set on Secondary Controller?
	route plans, and ARS assignment	
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	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).	Yes

Feature Name	Description	Supported While Set on Secondary Controller?
	All 5550 IP Console and MiVoice Business Consoles on the system, that have a network connection, share bulletin board	
Attendant Busy-Out (Console)	Places your attendant console in a busy-out condition (absent status) under certain circumstances. In the busy-out condition, incoming calls are automatically rerouted.	Yes
Attendant Busy-Out (Station)	Allows you to busy-out a specific station by using the attendant console	N/A
Attendant CAS Interface	Centralized Attendant Service interface allows an MiVoice Business 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

Feature Name	Description	Supported While Set on Secondary Controller?
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

Feature Name	Description	Supported While Set on Secondary Controller?
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 and MiVoice Business Console, you can view the system software version.	N/A
Attendant Language Selection	Enables attendant to choose the language of operation for the attendant console. The 5550 IP Console and MiVoice Business Console supports the following languages:	Yes

Feature Name	Description	Supported While Set on Secondary Controller?
	• 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 MiVoice Business system undergoes an update or restore.	
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	Yes

Feature Name	Description	Supported While Set on Secondary Controller?
	the cost of outgoing trunk calls	
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

Feature Name	Description	Supported While Set on Secondary Controller?
Attendant Setup and Cancellation of Station Features	Allows the attendant to set up and cancel certain station features such as Call Forward, Do Not Disturb, Callback, and Reminder	No
Attendant System Login	Requires the attendant to log on to the system to access certain programming functions from the attendant console	N/A
Attendant Tone Signaling	Allows the attendant to send tones over the circuit once a call has been established	Yes
Attendant Trunk Group Busy Status	Allows the attendant to display and/or print the busy status of the system trunk groups from the attendant console	Yes
Audio Files Update	Uploads audio files to the MiVoice Business 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 MiVoice Business system by using the System Audio Files Update form, or to multiple MiVoice Business systems by	No

Feature Name	Description	Supported While Set on Secondary Controller?
	using MiVoice Enterprise Manager.	
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.	

Feature Name	Description	Supported While Set on Secondary Controller?
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	No

Feature Name	Description	Supported While Set on Secondary Controller?
	• Provide Call Admission Control, that is, the rejection of new calls through a specific bottleneck point when consumed bandwidth exceeds maximum configured levels.	
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.	es
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	N/A

Feature Name	Description	Supported While Set on Secondary Controller?
	and exporting form data in .CSV format	
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.	Yes

Feature Name	Description	Supported While Set on Secondary Controller?
	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.	
Call By Name	See Phonebook.	Yes
Call Coverage	Provided through a combination of features, such as: Call Rerouting, Call Forward, Do Not Disturb, and Answer Plus-Mitel Call Distribution. The following Call Coverage Services can be configured and assigned to groups and users: Hot Desk PIN Security, Direct Transfer to Voice Mail, Post Call Destination, and Announcements.	Yes for all features except DND
Call Duration Display	Displays the call duration for incoming and outgoing	Yes

Feature Name	Description	Supported While Set on Secondary Controller?
	calls, in one minute increments (starting at 0:00)	
Call Forking	Enables a call to be split or forked so that several locations can ring simultaneously. MiVoice Business 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

Feature Name	Description	Supported While Set on Secondary Controller?
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	Yes

Feature Name	Description	Supported While Set on Secondary Controller?
	feature is supported on the 5330/5340 IP Phone and the MiCollab Client Softphone.	
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
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

Feature Name	Description	Supported While Set on Secondary Controller?
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 Screening	Allows devices configured as Secretaries (with Multicall and DSS/BLF appearances of other prime DNs - Boss's) to receive (ring) and route back Boss's incoming calls, while the Boss device is in Do Not Disturb (DND) state. The routed calls override the DND and ring the Boss's prime line. Screening can be activated or deactivated using the	Yes

Feature Name	Description	Supported While Set on Secondary Controller?
	appropriate programmable keys: DND on Boss, Superkey+DSS/BLF on Secretary.	
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	Yes

Feature Name	Description	Supported While Set on Secondary Controller?
	hearing the Call Waiting tone, the busy party can either respond or finish the current call.	
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
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	Yes

Feature Name	Description	Supported While Set on Secondary Controller?
	also be changed by using a Verified Account Code.	
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	Yes

Feature Name	Description	Supported While Set on Secondary Controller?
	Phones now support the G.722.1 wideband codec.	
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.	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
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	Yes

Feature Name	Description	Supported While Set on Secondary Controller?
	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.	
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	Yes

Feature Name	Description	Supported While Set on Secondary Controller?
	dumps, display phones, and attendant consoles.	
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	Yes

Feature Name	Description	Supported While Set on Secondary Controller?
	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.	
DID Single Ring Cadence	Gives single ring back to outside callers	N/A
Direct-In Lines (DIL)	Allows incoming trunks to be assigned to a specific station or hunt group so that calls from the trunk ring the station or hunt group directly	Yes
Direct Inward Dialing (DID)	Permits incoming calls on designated trunks to directly access predefined stations (or other answering points) on the system	Yes
DID Service	Offers a Direct Inward Dialing solution, alternative to DID Ranges for CPN Substitution and user-based System Speed Calls. This feature provides the ability to reallocate DID numbers to their answering points from ESM. Incoming calls can be routed to specified destinations based on the mapping of DID numbers to their destinations, without using the interim	Yes

Feature Name	Description	Supported While Set on Secondary Controller?
	Speed Call System. The feature provides a single consolidated provisioning interface: the DID Service form, but the configured data is stored in the Call Recognition Service form.	
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	Yes

Feature Name	Description	Supported While Set on Secondary Controller?
	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.	
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	Yes

Feature Name	Description	Supported While Set on Secondary Controller?
	the user is talking (priority based on key position)	
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	Supported While Set on 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 Name	Description	Supported While Set on Secondary Controller?
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

Feature Name	Description	Supported While Set on Secondary Controller?
Flexible Dimensioning	Allocates database memory to each feature resource. The amount of memory determines the maximum size of the feature resource; the system borrows memory from other resources that are not in use. This feature allows individual systems to be tailored to individual business needs, resulting in optimal performance for a particular system.	N/A
Forced Non-Verified Account Codes	Customers such as law firms require ways of tracking calls for billing purposes and need the ability to enter a number (account code) as a record for a call. These numbers do not have to be "verified", as the number might only be valid for the duration of a case. But they must be "forced" in order to ensure that an Account Code can be used as a billing tracking mechanism (tracked in SMDR record). The solution is to have the ability to use a Forced Non-Verified Account Code.	Yes
Ground Button	Allows you to place a call on Consultation Hold and return to dial tone to invoke station features. The	N/A

Feature Name	Description	Supported While Set on Secondary Controller?
	Ground Button provides an alternate method of producing a Switchhook Flash.	
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	Yes

Feature Name	Description	Supported While Set on Secondary Controller?
	“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 OIG.	
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

Feature Name	Description	Supported While Set on Secondary Controller?
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
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.	Yes

Feature Name	Description	Supported While Set on Secondary Controller?
	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.	
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,	Yes

Feature Name	Description	Supported While Set on Secondary Controller?
	green light. The Green BLF Lamp for Logged in Hotdesk User Class of Service option controls this capability.	
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
Integrated Directory Service	The Integrated Directory Service (IDS) feature uses the Lightweight Directory	N/A

Feature Name	Description	Supported While Set on Secondary Controller?
	Access Protocol (LDAP) to synchronize user and service data from your corporate directory server to the MiVoice Business platform. System Data Synchronization is then used to share the data among the administrative group. Note that in MiVoice Business Release 5.0, data is synchronized in one direction only, from the corporate directory server to MiVoice Business, 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 Services Configuration form.	
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.	N/A

Feature Name	Description	Supported While Set on Secondary Controller?
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	N/A

Feature Name	Description	Supported While Set on Secondary Controller?
	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
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 MiVoice IP Phones can use LLDP-MED to obtain the VoIP-specific configuration	N/A

Feature Name	Description	Supported While Set on Secondary Controller?
	information that they require to operate in a converged network—information such as VLAN ID, COS Priority, and DSCP values.	
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 MiVoice Business system	No

Feature Name	Description	Supported While Set on Secondary Controller?
	undergoes an update or restore.	
Last Group Member Routing (LGMR)	The Last Group Member Routing (LGMR) feature is an enhancement to the Ring Groups feature. When enabled, connects a caller to the same ring group member that attended the caller within a specified period of time.	Yes
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 four types of lines: Prime, Non-Prime, No Where Prime (also referred to as a Phantom Line), and Mobile Line (which applies to 69xx sets only).	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.	Yes

Feature Name	Description	Supported While Set on Secondary Controller?
	The feature involves two components: • zone identification based on IP address • device location information in the 3300 ICP's SMDR records	
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
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	N/A
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.	Yes

Feature Name	Description	Supported While Set on Secondary Controller?
	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).	
Meet-Me Answer	Allows a paged party to respond to a Group Page without knowing the identity or location of the paging party	Yes
Meet-Me Conference	Allows you to set up a conference where up to eight people can dial in and join from anywhere. Each participant dials an MMC access number, bridge number, and an optional PIN to join the conference.	No
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	Yes

Feature Name	Description	Supported While Set on Secondary Controller?
	can review the message on the display (if applicable) and/or call the sender back.	
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

Feature Name	Description	Supported While Set on Secondary Controller?
Multiple Consoles	See Attendant Consoles (Multiple)	Yes
Multi-Level Auto Attendant	Allows a hierarchical menu to be programmed on the auto attendant. This provides callers with better self-service access to the person or department they are calling	N/A
Multi-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 license	Simplifies and cost reduces the hospitality solution where hotels require multiple devices in a single suite. Hoteliers can license hotel rooms as single suites: up to 6 devices can be configured in a suite while consuming only a single System license.	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 (MoH) provides callers with music	Yes

Feature Name	Description	Supported While Set on Secondary Controller?
	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 and four types of Music on Hold sources. Music on Hold types: • Call Coverage Based or User Based - Uses the MoH source assigned to the user who is holding the call or the destination the call is queued on. • Zone Based - Uses the source provisioned for the user or destination zone in the Network Zones form. • System Based - Uses the default source when the above types are not provisioned, invalid, or otherwise inappropriate.	
	Music on Hold sources: • Analog • Digital • Embedded (allows systems to use	

Feature Name	Description	Supported While Set on Secondary Controller?
	embedded .wav files as music sources) • Live Music on Hold Over IP (uses a PC with an internet connection or third-party MoH server to stream audio to an IP endpoint in the system)	
Music On Hold over IP	Music on Hold over IP plays live music from a source on the Internet.	No
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

Feature Name	Description	Supported While Set on Secondary Controller?
Networking	The system supports both analog and digital networking. See Node ID Recognition and Uniform Numbering Plan.	N/A
Networking using MSDN/ MSAN	MSDN/DPNSS provides fast call setup capabilities and feature transparency across the network. No significant difference between making a local call and a network call is apparent to the user All of the MSDN networking packages require that each PBX has MSDN Voice I or MSAN installed.	N/A
Networked ACD	Supports ACD functions over a Mitel Switched Digital Network (MSDN). Agent skill groups at different locations (on different systems) may service calls on the network independently of where the call entered the network.	No
Networked Group Page	Group Paging can be completed across a network or network cluster, allowing, for example, a set on system A to page a specific group on system B.	Yes

Feature Name	Description	Supported While Set on Secondary Controller?
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 MiVoice Business 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 MiVoice Business system. Allows transitions between all of the supported service	Yes

Feature Name	Description	Supported While Set on Secondary Controller?
	modes (Day, Night 1, or Night 2) at independent times	
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

Feature Name	Description	Supported While Set on Secondary Controller?
Off-Hook Detection to Display sets	Used in hospitals and nursing applications. If someone fails to complete dialing, the alert is sent to a set	Yes
Off-Hook Voice Announce	Allows you to receive a direct page during a handset or headset call. See Direct Page	Yes
ONS Ports as Music Sources	Allows a music source, either the system's Music on Hold source, or an ACD alternate music source, to be an ONS port instead of a DMP module. The Alternative Recording Device (ARD) is an off-hook ONS port that connects to callers in a listen-only conference. The user decides what is supplied on the ONS port - silence, music, or endless loop recordings.	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

Feature Name	Description	Supported While Set on Secondary Controller?
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	Yes

Feature Name	Description	Supported While Set on Secondary Controller?
	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.	
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)	N/A

Feature Name	Description	Supported While Set on Secondary Controller?
	and as dedicated data communications printers.	
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	Yes

Feature Name	Description	Supported While Set on Secondary Controller?
	the PBX when a user checks in or checks out.	
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	N/A

Feature Name	Description	Supported While Set on Secondary Controller?
	the first programmed trunk including Class of Service, Day, Night1, Night2 and Circuit Descriptor Number. Trunk Name and Comments are left blank.	
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

Feature Name	Description	Supported While Set on Secondary Controller?
Redial	Automatically dials the last manually dialed number	Yes
Redial -Saved Number	Allows you to save a number for future dialing. The number remains saved until a replacement number is saved	Yes
Release	Allows you to release from an attempted connection to an external party without going on-hook. Release is useful when you encounter a busy or unavailable external party that you are attempting to add to a conference.	Yes
Reminder	Allows you to program your set to ring and provide a message at a specified time within a 24-hour period	No
Remote Wake-up Calls	Wake-up calls can be set or cancelled remotely from a 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	No

Feature Name	Description	Supported While Set on Secondary Controller?
	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.	
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	Yes

Feature Name	Description	Supported While Set on Secondary Controller?
	configure IP Phone and IP Console resiliency from the System Administration Tool of the local element , or through OPS Manager.	
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; can contain local, resilient and remote DNs as members.	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

Feature Name	Description	Supported While Set on Secondary Controller?
Ringing -Discriminating	Allows you to distinguish between incoming internal calls, incoming trunk calls, tie line calls, and Callbacks by using different ringing patterns (cadences)	Yes
Ringing -Discriminating (Optional)	Allows you to change the Discriminating Ringing patterns on ONS/OPS lines so that you hear internal ringing (1 second on and 3 seconds off) for both internal and external calls	N/A
Ringing Line Select	Allows you to answer any ringing line by going off-hook.	Yes
Registering and Auto-provisioning Multiline IP Telephones without a DN	Reduces and simplifies the provisioning and installation effort for IP Phones to be registered as a basic Userless Device —a device with service level "IP Device Only" that allows the user to log in, but does not require a license. With this Automatic DN Selection registration and auto-provisioning method, a device can be registered and brought to service without being pre-configured first and the installer does not need to specify the device's Directory Number (DN— only the Set Registration	No

Feature Name	Description	Supported While Set on Secondary Controller?
	Access Code—to initiate the registration.	
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	N/A

Feature Name	Description	Supported While Set on Secondary Controller?
	(SNMP) governs the management and monitoring of network devices and their functions.	
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

Feature Name	Description	Supported While Set on Secondary Controller?
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
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.	No

Feature Name	Description	Supported While Set on Secondary Controller?
	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.	
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	N/A

Feature Name	Description	Supported While Set on Secondary Controller?
	to four POTS phones are connected directly to the Central Office via LS Trunks.	
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
CTI Support	Supports OIG and MiTAI 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	Yes

Feature Name	Description	Supported While Set on Secondary Controller?
	functionality). When the curfew time is reached, users receive a warning tone indicating that calls in progress will be cleared down.	
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 Service Templates: Enables you to program a subset of the information normally added on the User and Services 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

Feature Name	Description	Supported While Set on Secondary Controller?
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	N/A

Feature Name	Description	Supported While Set on Secondary Controller?
	better system resource management	
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
Traveling Class Marks	Traveling Class Marks (TCM) extend users access to features and services available to them on their host MiVoice Business system to other MiVoice Business 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	Yes

Feature Name	Description	Supported While Set on Secondary Controller?
	a maintenance phone is required.	
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	Yes

Feature Name	Description	Supported While Set on Secondary Controller?
	label and trunk number is displayed.	
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	No

Feature Name	Description	Supported While Set on Secondary Controller?
	the trunks at the Central Office (CO)	
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 Services 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
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	N/A

Feature Name	Description	Supported While Set on Secondary Controller?
	• Voice Mail - Softkey support with Mitel's MiCollab Embedded Voice Mail • Voice Mail - ONS Interface.	
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 signaling delivers voice over PRI channels, with MSDN call setup, feature invocation, and tear-down signaling over the IP network. Full XNET DPNSS feature transparency is maintained.	N/A

6.5.1 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)

Feature	Description
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

Feature	Description
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 canceled prior to sending.
Message Addressing	Subscribers can address messages to multiple recipients and hear the recipient's name played back to confirm valid entry of mailbox numbers.
Forward Voice Mail to E-Mail	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.

<table>
<tr><th>Feature</th><th>Description</th></tr>
<tr><td>Message Notification</td><td>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.</td></tr>
</table>

Feature	Description
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.
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

Feature	Description
	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 MiVoice Business 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 Embedded Mitel Express Manager (networked and clustered), NuPoint Messenger, and other VPIM2-compliant mail servers (G.711 compliant), and is compatible with Hot Desking.

Feature	Description
Personal Contacts	Personal Contacts allow users to store alternate numbers where callers can contact them instead of leaving a message. Callers are prompted in the greeting to press a key to have their call transferred to the alternate number—they are never told the number. Users can program up to ten (10) Personal Contacts.
Distribution Lists	A Distribution List allows mailbox subscribers to send messages to several people at one time. There are two types of distribution lists: personal lists and global lists. Personal lists are set up by individual subscribers for their own use. Global lists are for use by all subscribers and are set up using the VM Distribution Lists Form. Only the system administrator can set up or change the global lists. Up to 49 global lists (001-049) can be created. A fiftieth list (000) is already set up to broadcast messages to every local mailbox. Users can create up to 10 personal lists (050-059). Each distribution list can have up to 750 contacts.
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

Feature	Description
	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	MiVoice Business supports a single, large, voice mail hunt group with up to 240 members.

6.5.2 Embedded Voice Mail 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.

Feature	Description
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".
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

Feature	Description
	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.
VM Fax Detection	Detects an incoming fax tone and directs it to the fax mailbox/extension

6.5.3 Features supported by protocols

The following tables summarize the features supported by Q.SIG and PRI protocols:

6.5.3.1 QSIG (3300 ICP Only)

Q.SIG is supported on the 3300 ICP Platforms only.

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.
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.

Feature	Description
	• 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

Standard	Feature	Mitel 3300
ISO 11571	Numbering Plan	X
ISO 11582	Generic SS Platform (GF)	X
ISO 14136	Calling Line Identification Presentation (CLIP)	X
ISO 14136	Connected Line Identification Presentation (COLP)	X
ISO 14136	CLIP/COLP Restriction (CLIR)	X
ISO 13864, 13868	Calling Name Identification Presentation (CNIP)	X
ISO 13864, 13868	Connected Name Identification Presentation (CONP)	X
ISO 13864, 13868	CNIP/CONP Restriction (CNIR)	X
ISO 13872, 13873	Call Forwarding Unconditional (CFU)	X (note 1)
ISO 13872, 13873	Call Forwarding Busy (CFB)	X (note 1)
ISO 13872, 13873	Call Forwarding No Reply (CFNR)	X (note 1)
ISO 13865, 13869	Call Transfer (CT)	X (By join)

Standard	Feature	Mitel 3300
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)

Does not support Interrogation. It is a way to determine the call forwarding status of a remote phone. 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 th e other side so that the Mitel switch initiates the optimization request. Does not support MWI interrogate function. It is a way to determine the message waiting lamp status of a remote phone. Does not support connection retention. Connection retention holds up a virtual call between the two end-points. Mitel supports path reservation which ensur es that resources are available when User B can accept User A's call and service retention in that the call is compe lled to complete.

6.5.3.2 PRI

The following table lists features supported by PRI.

Feature	Description
ANI/DNIS/ISDN Number Delivery	Automatic Number Identification and Dialed Number Identification Service

Feature	Description
	identify numbers that are transmitted on an incoming trunk.
Call-by-call Service	With Call-by-Call Service, access channels do not have to be dedicated to specific services such as OUTWATS or 800 services. This enables the customer to reduce facilities and integrate dedicated and switched, inbound and outbound, voice and data traffic on a single facility. It also allows a business with calling peaks to dynamically allocate coverage across channels so that access lines are optimized. This implementation ensures that incoming calls are not turned away because all incoming channels are busy while adjacent outgoing channels are idle.
Calling Line Identification	The 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.

6.5.3.3 MSDN/DPNSS (3300 ICP Only)

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.

Feature	Description
SMDR -External	Collects data for outgoing and incoming trunk calls
SMDR -Internal	Collects data for calls made between stations within the system
Recall	Allows an incoming caller, who has been transferred to an idle station and not answered within a specified time-out period, call back the last party who handled the call. Similar time-out recalls occur for parties who are transferred to busy stations or placed on hold.
Tandem Trunking	The system can transparently interconnect trunk circuits originating from one CO or PBX and terminating on another (tandem trunking), without attendant intervention.
Trunk Select - Direct	Allows you to access an outside trunk for the purposes of originating and receiving external calls. Because the trunk is assigned to a line appearance, you can access the trunk to make or answer calls without trunk access codes.
Override	Allows you to enter a conversation at a busy station or ring a station with Do Not Disturb activated. Before you enter the conversation, all parties receive a warning tone.
Serial Call	Allows a centralized attendant to set up serial calls for users on remote PBXs.

Feature	Description
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.

6.5.4 Security Features

Encrypted media and signaling path is supported for all Mitel's IP phones.

6.5.4.1 Encrypted Media Path

Media path security between IP phones or between an IP phone and a controller is accomplished with either the Secure Real Time Protocol (SRTP), which is a standards based protocol described by RFC 3711, or a Mitel variation of SRTP termed Mitel SRTP, both using the 128-bit Advanced Encryption Standard (AES). Mitel-SRTP uses the same encryption algorithm as SRTP.

The MiVoice Business controller specifies streaming connections using SRTP or Mitel-SRTP based on whether SRTP is enabled on the MiVoice Business and the capabilities of the connection endpoints, including Mitel and third party phones. If SRTP is enabled and supported by both end points, SRTP is chosen; if not, Mitel SRTP is chosen. Connections to third party equipment must use SRTP; some older models of Mitel phones only support Mitel SRTP. For details, refer to the Mitel IP Sets Engineering Guidelines.

6.5.4.2 Encrypted Signaling Path

Two main protocols are supported and either may be used to secure a signaling channel. These are:

- Secure MiNET, which is a Mitel standard: When using Secure MiNET, the phone will use a local port in the range of TCP 6900-6999 and the MiVB will use TCP Port 6802.
- TLS (Transport Layer Security), which is an open standard: When using TLS, the phone will use a local port in the range of TCP 6900-6999 and the MiVB will use TCP Port 6901

Mitel's Secure MiNET protocol uses the Advanced Encryption Standard (AES) to encrypt call control packets. Using secure MiNET ensures that call control signaling packets between the IP phones and the 3300 ICP (or MiVoice Business) are protected from eavesdropping. Using secure MiNET also protects the call control engine from unauthorized control packets.

The TLS security protocol provides data encryption, server authentication message integrity, and optional client authentication for a TCP/IP connection. TLS will prevent unauthorized access to administrative functions. TLS encrypts all traffic on the link to prevent sniffing of user names and passwords.

For Release 9.1 the system can be configured to only support TLS 1.2.

6.5.4.3 Phone and User Authentication

Mitel IP phones are available using two protocols.

1. MiNET – an encrypted proprietary stimulus based protocol. For a Mitel set to register with the MiVoice Business call control the set type, PIN number and MAC address must be accepted by the system. This information is entered by the administrator (it may bulk entered too). After registration, the MiVoice Business call control has knowledge of the relationship between MAC Address, IP address, extension number and PIN Registration Number. This relationship of MAC/IP/Ext/PIN must be valid in order for the MiVoice Business to allow communications to proceed.
2. Session Initiation Protocol (SIP) – MiVoice Business supports SIP stations that have been tested successfully through an interoperability process run by the SIP Center of Excellence team at Mitel. The SIP set must use a username and password combination to successfully register. This may be encrypted for additional security – assuming the SIP end point supports encryption.

6.5.4.4 Worm and Virus Protection

Applications such as MiVoice Business must process data in real-time. Real-time applications require unfettered access to processor resources, memory systems, disk drive accesses and network communications. When MiVoice Business is deployed on proprietary hardware, industry standard servers or virtual machines - as per Mitel's MiVoice Business Engineering Guidelines - the machine's resources will have been sized to ensure that the applications will have unrestricted and timely access to the resources that they require.

Since these real-time data processing applications are executing on carefully sized computing platforms, the installation of antivirus software is not recommended.

6.5.4.5 Prevention of Toll Abuse

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

The MiVoice Business system provides comprehensive toll control as an integral part of the call control engine.

The MiVoice Business call control gives the administrator the ability to restrict a user's access to trunk routes and/or specific external directory numbers.

The MiVoice Business call control offers several Class of Restriction (CoR) and Class of Service (CoS) capabilities, that when used correctly can substantially reduce the risk of toll abuse by disallowing the dialing of certain external telephone numbers or ranges of numbers (Call Barring).

This is achieved by associating in software each extension and trunk with a CoR and providing specific barring plans with each CoR.

Mitel’s implementation of CoR affords great flexibility. Up to 64 different Classes of Restriction can be specified. An extension user attempting to dial barred numbers will result in them receiving a number unobtainable tone.

As a deterrent to toll abuse by internal callers, Station Message Detail Recording (SMDR) logs 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.

6.5.4.6 Secure Management Interfaces

MiVoice Business includes a fully integrated set of management tools designed to install, manage, and administer MiVoice Business 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 Transport Layer Security (TLS) for data encryption.

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

Mitel also offers Mitel Performance Analytics, which supports two factor authentication to provide secure remote administration.

6.5.4.7 Secure Applications

Mitel addresses application security via:

- MiCollab Softphone - 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 TLS and provides strong certificate-based authentication for API use.

6.5.4.8 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.

Hospitality 7

This chapter contains the following sections:

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

The 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
- 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

MiVoice Business 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.

7.1 Property Management System

MiVoice Businesscan 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 a third-party Serial-to-IP port converter.

When guest information is changed in the PMS system, the PMS sends messages to the PBX.

7.2 Clustered Hospitality

A Clustered Hospitality solution provides feature functionality across multiple MiVoice Business controllers. 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 MiVoice Business Console and can also host guest room extensions.

Clustered Hospitality supports

- Hotel logs and reports via a networked printer
- Shared Telephone Service (STS), available if all members reside on the same MiVoice Business as the linked suite.
- Configuration of room extensions and suites from any MiVoice Business 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 MiVoice Businesscontrollers to ensure the data on both systems matches on a cluster-wide basis)

7.3 Centralized Hospitality Deployment

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

The architecture provides scalability and centralization. MiVoice Business 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.

7.3.1 5540 IP Console

MiVoice Businesscontinues 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, MiVoice Businessnow uses a smaller footprint as phone appliances are clustered.

7.3.2 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.

7.3.3 Capacity

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